

# **Oracle® Communications Session Border Controller**

ACLI Configuration Guide  
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# About this Guide

The ACLI Configuration Guide provides information about:

- Basic concepts that apply to the key features and abilities of your Oracle Communications Session Border Controller
- Information about how to load the Oracle Communications Session Border Controller system software image you want to use and establish basic operating parameters
- System-level functionality for the Oracle Communications Session Border Controller
- Configuration of all components of the Oracle Communications Session Border Controller

Release Version S-CZ7.2.0 is supported on the Acme Packet 3820, 4500, 4600, 6100, and 6300 series platforms.

## Related Documentation

The following table lists the members that comprise the documentation set for this release:

Document Name	Document Description
Acme Packet 4500 Hardware Installation Guide	Contains information about the components and installation of the AP4500.
Acme Packet 3820 Hardware Installation Guide	Contains information about the components and installation of the AP 3800.
Acme Packet 6300 Hardware Installation Guide	Contains information about the components and installation of the AP 6300.
Acme Packet 6100 Hardware Installation Guide	Contains information about the components and installation of the AP 6100.
Release Notes	Contains information about the current documentation set release, including new features and management changes.
ACLI Configuration Guide	Contains information about the administration and software configuration of the Oracle Communications Session Border Controller.
ACLI Reference Guide	Contains explanations of how to use the ACLI, as an alphabetical listings and descriptions of all ACLI commands and configuration parameters.
Maintenance and Troubleshooting Guide	Contains information about Oracle Communications Session Border Controller logs, performance announcements, system management, inventory management, upgrades, working with configurations, and managing backups and archives.
MIB Reference Guide	Contains information about Management Information Base (MIBs), Acme Packet's enterprise MIBs, general trap information, including specific details about standard traps and enterprise traps, Simple Network Management Protocol (SNMP) GET query information (including standard and enterprise SNMP GET query names, object identifier names and numbers, and descriptions), examples of scalar and table objects.
Accounting Guide	Contains information about the Oracle Communications Session Border Controller's accounting support, including details about RADIUS accounting.
HDR Resource Guide	Contains information about the Oracle Communications Session Border Controller's Historical Data Recording (HDR) feature. This guide includes HDR configuration and system-wide statistical information.

## About this Guide

Document Name	Document Description
Administrative Security Essentials	Contains information about the Oracle Communications Session Border Controller's support for its Administrative Security license.
Security Guide	Contains information about security considerations and best practices from a network and application security perspective for the Oracle Communications Session Border Controller family of products.

## Revision History

Date	Description
June 2014	<ul style="list-style-type: none"> <li>Initial Release</li> </ul>
November 2014	<ul style="list-style-type: none"> <li>Removes cavium processor on ETC card as hardware supporting RTCP generation.</li> <li>Removes incorrect platform limitations for SIPREC.</li> <li>Clarifies physical interface configuration.</li> <li>Removes Phy Link Redundancy section</li> <li>Corrects cut-and-paste error that used charging-function-address instead of charging-vector-mode.</li> <li>Updates <b>restriction-mask</b> parameter with maximum value of 128 to accommodate for IPv6 address masking</li> <li>Self-signed certificates are allowed only for MSRP connections.</li> <li>Removes ip-can-change from specific-action-subscription</li> </ul>
December 2014	<ul style="list-style-type: none"> <li>Updated trust-me references.</li> <li>Updates Session Agent Ping Message Formatting Configuration task and Session Agent Ping Message Formatting Configuration topic to reflect valid descriptions for the <b>session-agent &gt; ping-to-user-part</b> and <b>session-agent &gt; ping-from-user-part</b> parameters.</li> <li>Updates Denial of Service feature group.</li> </ul>
March 2015	<ul style="list-style-type: none"> <li>Adds note regarding gateway address in <b>host-route</b> feature description.</li> </ul>
May 2015	<ul style="list-style-type: none"> <li>Adds descriptive information on HMR reserved variables.</li> <li>Removes sections in Transcoding chapter under the SNMP heading. Adds note explaining their new location in MIB Reference document.</li> </ul>
October 2015	<ul style="list-style-type: none"> <li>Corrects command arguments for p-early-media-header parameter.</li> <li>Removes reference to unsupported SRR feature.</li> <li>Updates configuration in QoS Accounting Configuration section.</li> <li>Corrects re-connection timing criteria for establishing connections to external policy servers.</li> <li>Removes ENUM from the "Compound Key LRT Lookup" topic because ENUM does not support compound lookup keys.</li> <li>Re-writes description of packet trace command.</li> <li>Updates Session Agent Minimum Reserved Bandwidth topic with limitation to Acme Packet 3820 and 4500</li> <li>Edits the sections on Admin Security and Admin Security ACP to clarify the differences between the features, and to explain the ramifications of uninstalling the licenses</li> <li>Remove MGCP support in documentation</li> </ul>
November 2015	<ul style="list-style-type: none"> <li>Adds text on conditional codec's re-order support.</li> </ul>



Date	Description
December 2015	<ul style="list-style-type: none"> <li>• Renames HIP section in Network Interfaces to Administrative Applications Over Media Interfaces. Corrects description regarding firewalls and ports. Adds gateway requirement and overlapping subnet advisory.</li> <li>• Minor edits to Network Interfaces conceptual and Configuration sections.</li> <li>• Updates HIP Address Configuration for clarity. Adds SSH parameter. Adds gateway requirement and overlapping subnet advisory.</li> <li>• Removes Source-based Routing section from System Configuration chapter.</li> </ul>
January 2015	<ul style="list-style-type: none"> <li>• Corrects the explanation of the tos-value setting.</li> <li>• Updates list of transcodable codecs for the Pooled Transcoding feature.</li> </ul>
January 2016	<ul style="list-style-type: none"> <li>• Removes FIPS documentation.</li> </ul>
March 2016	<ul style="list-style-type: none"> <li>• Adds requirement that any media interface configured with comm-monitor must also belong to a realm.</li> <li>• Adds refreshRegForward option to the ims-aka-profile.</li> <li>• Corrects the description of the ext-policy-server parameter disconnect-on-timeout .</li> <li>• Corrects description of parameter for source realm selection in multi-stage routing.</li> <li>• Adds the Request URI Construction as Forwarded to SAG-member Session Agent feature description to the Session Routing and Load Balancing Chapter</li> </ul>
April 2016	<ul style="list-style-type: none"> <li>• Corrects call flow diagrams for refer-initiated transfers</li> </ul>
June 2016	<ul style="list-style-type: none"> <li>• Added a note in the section Tunnel mode vs Transport mode under IPSec support</li> <li>• Edited IPSec support</li> <li>• Clarifies Static Flows as unidirectional</li> <li>• Corrects explanation of diameter manipulation</li> </ul>
September 2016	<ul style="list-style-type: none"> <li>• Adds new scenario under Fax Transcoding topic</li> <li>• Removes Single SRS capacity section</li> </ul>
December 2016	<ul style="list-style-type: none"> <li>• Corrects UDP checksum parameter's platform applicability</li> </ul>
January 2017	<ul style="list-style-type: none"> <li>• Updates the Ingress Policy in Transcoding</li> </ul>
February 2017	<ul style="list-style-type: none"> <li>• Corrects show sipd agents &lt;agent name&gt; output example</li> <li>• Clarifies that stop-sag-recurse ends message sequence, including attempts to higher cost agents</li> </ul>



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# Oracle Communications Session Border Controller Basics

This chapter introduces some basic concepts that apply to the key features and abilities of your Oracle Communications Session Border Controller. It is necessary that you understand the information included in this chapter to comprehend the ways to configure your Oracle Communications Session Border Controller. This chapter only provides a high level overview of some important Oracle Communications Session Border Controller concepts.

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## What Is a Realm

A realm is a logical way of identifying a domain, a network, a collection of networks, or a set of addresses. Realms are used when a Oracle Communications Session Border Controller communicates with multiple network elements over a shared intermediate connection. Defining realms allows flows to pass through a connection point between two networks.

From an external perspective, a realm is a collection of systems that generates real-time interactive communication sessions comprised of signaling messages and media flows, or a group of multiple networks containing these systems. These systems may be session agents such as call agents, softswitches, SIP proxies, H.323 gatekeepers, IP PBXs, etc., that can be defined by IPv4 addresses. These systems can also be IP endpoints such as SIP phones, IADs, MTAs, media gateways, etc.

From an internal perspective, a realm is associated with Oracle Communications Session Border Controller configurations to define interfaces and resources in a logical way. Realms are used to support policies that control the collection of systems or networks that generate media sessions. Realms are referenced by other configuration elements in order to support this functionality across the protocol the Oracle Communications Session Border Controller supports and to make routing decisions.

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## Nested Realms

Nested Realms is a Oracle Communications Session Border Controller feature that supports hierarchical realm groups. One or more realms may be nested within higher order realms. Realms and sub-realms may be created for media and bandwidth management purposes. This feature supports:

- Separation of signaling & media on unique network interfaces
- Signaling channel aggregation for Hosted IP Services applications
- Configuration scalability
- Per-realm media scalability beyond single physical interface capacity

- Nested bandwidth admission control policies

## What Is a Session Agent

---

A session agent defines an internal signaling endpoint. It is an internal next hop signaling entity that applies traffic shaping attributes to flows. For each session agent, concurrent session capacity and rate attributes can be defined. Service elements such as gateways, softswitches, and gatekeepers are defined automatically within the Oracle Communications Session Border Controller as session agents. The Oracle Communications Session Border Controller can also provide load balancing across the defined session agents.

### SIP session agents

SIP session agents can include the following:

- Softswitches
- SIP proxies
- Application servers
- SIP gateways

### H.323 session agents

H.323 session agents can include the following:

- gatekeepers
- gateways
- MCUs

## Why You Need Session Agents

---

You can use session agents to describe next or previous hops. You can also define and identify preferred carriers to use for traffic coming from session agents. This set of carriers is matched against the local policy for requests coming from the session agent. Constraints can also be set for specific hops.

In addition to functioning as a logical next hop for a signaling message, session agents can provide information regarding next hops or previous hops for SIP packets, including providing a list of equivalent next hops.

## How to Use Session Agents

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You can use session agents and session agent groups (along with local policies) to define session routing for SIP and H.323 traffic. You can associate a realm with a session agent to identify the realm for sessions coming from or going to the session agent.

## What is a Session Agent Group

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A session agent group contains individual session agents bundled together. A SAG indicates that its members are logically equivalent and can be used interchangeably. This allows for the creation of constructs like hunt groups for application servers or gateways. Session agent groups also assist in load balancing among session agents.

Session agent groups can be logically equivalent to the following:

- Application server cluster
- Media gateway cluster
- Softswitch redundant pair
- SIP proxy redundant pair

- Gatekeeper redundant pair

## High Availability

---

High Availability (HA) is a network configuration used to ensure that planned and unplanned outages do not disrupt service. In an HA configuration, Oracle Communications Session Border Controllers (SBC) are deployed in a pair to deliver continuous high availability for interactive communication services. Two SBCs operating in this way are called an HA node. The HA node design ensures that no stable call is dropped in the event of an outage.

In an HA node, one SBC operates in the active mode and the other SBC operates in the standby mode.

- **Active.** The active member of the HA node is the system actively processing signal and media traffic. The active member continuously monitors itself for internal process and IP connectivity health. If the active member detects a condition that can interrupt or degrade service, it hands over its role as the active member of the HA node to the standby member.
- **Standby.** The standby member of the HA node is the backup system. The standby member is fully synchronized with active member's session status, but it does not actively process signal and media traffic. The standby member monitors the status of the active member and it can assume the active role without the active system having to instruct it to do so. When the standby system assumes the active role, it notifies network management using an SNMP trap.

The SBC establishes active and standby roles in the following ways.

- If an SBC boots up and is alone in the network, it is automatically the active system. If you pair a second SBC with the first one to form an HA node, the second system automatically establishes itself as the standby.
- If both SBCs in the HA node boot up at the same time, they negotiate with each other for the active role. If both systems have perfect health, then the SBC with the lowest HA rear interface IPv4 address becomes the active SBC. The SBC with the higher HA rear interface IPv4 address becomes the standby SBC.

If the rear physical link between the two SBCs is unresponsive during boot up or operation, both will attempt to become the active SBC. In this circumstance, processing does not work properly.

The standby SBC assumes the active role when:

- it does not receive a checkpoint message from the active SBC for a certain period of time.
- it determines that the active SBC health score declined to an unacceptable level.
- the active SBC relinquishes the active role.

To produce a seamless switch over from one SBC to the other, the HA node members share their virtual MAC and virtual IP addresses for the media interfaces in a way that is similar to Virtual Router Redundancy Protocol (VRRP). Sharing these addresses eliminates the possibility that the MAC address and the IPv4 address set on one SBC in an HA node will be a single point of failure. Within the HA node, the SBCs advertise their current state and health to one another in checkpointing messages to apprise each one of the other one's status. Using the Oracle HA protocol, the SBCs communicate with UDP messages sent out and received on the rear interfaces. During a switch over, the standby SBC sends out an ARP request using the virtual MAC address to establish that MAC address on another physical port within the Ethernet switch. To the upstream router, the MAC address and IP address are still alive. Existing sessions continue uninterrupted.



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## Getting Started

Prior to configuring your Oracle Communications Session Border Controller for service, we recommend that you review the information and procedures in this chapter.

This chapter offers information that will help you:

- Review hardware installation procedures
- Connect to your Oracle Communications Session Border Controller using a console connection, Telnet, or SSH (secure shell)
- Become familiar with the Oracle Communications Session Border Controller's boot parameters and how to change them if needed
- Obtain, add, and delete Oracle Communications Session Border Controller software licenses
- Load and activate a Oracle Communications Session Border Controller software image
- Choose a configuration mechanism: ACLI, Oracle Communications Session Element Manager or ACP/XML
- Enable RADIUS authentication
- Customize your login banner

---

## Installation and Start-Up

After you have completed the hardware installation procedures outlined in the the relevant *Hardware Installation Guide*, you are ready to establish a connection to your Oracle Communications Session Border Controller. Then you can load the software image you want to use and establish basic operating parameters.

### Hardware Installation Process

Installing the Oracle Communications Session Border Controller hardware in a rack requires the following process.

1. Unpack the Oracle Communications Session Border Controller hardware.
2. Install the Oracle Communications Session Border Controller hardware into the rack.
3. Install the power supplies.
4. Install the fan modules.
5. Install the physical interface cards.
6. Cable the Oracle Communications Session Border Controller hardware.



**Note:** Complete installation procedures fully and note the safety warnings to prevent physical harm to yourself and damage to the Oracle Communications Session Border Controller hardware.

For more information, see the hardware documentation.

# Connecting to Your Oracle Communications Session Border Controller

You can connect to your Oracle Communications Session Border Controller either through a direct console connection, or by creating a remote Telnet or SSH session. Both of these access methods provide you with the full range of configuration, monitoring, and management options.



**Note:** By default, Telnet and FTP connections to your Oracle Communications Session Border Controller are enabled.

## Create a Console Connection

Using a serial connection, you can connect your laptop or PC directly to the Acme Packet hardware. If you use a laptop, you must take appropriate steps to ensure grounding.

One end of the cable plugs into your terminal, and the other end plugs into the RJ-45 Console port on the NIU (or management ports area on the Acme Packet 6300).

To make a console connection to your hardware:

1. Set the connection parameters for your terminal to the default boot settings:
  - Baud rate: 115,200 bits/second
  - Data bits: 8
  - Parity: No
  - Stop bit: 1
  - Flow control: None
2. Connect a serial cable to between your PC and the hardware's console port.
3. Apply power to the hardware.
4. Enter the appropriate password information when prompted to log into User mode of the ACLI.

You can set the amount of time it takes for your console connection to time out by setting the **console-timeout** parameter in the system configuration. If your connection times out, the login sequence appears again and prompts you for your passwords. The default for this field is 0, which means that no time-out is being enforced.

## Incoming Telnet Connections and Time-outs

You can Telnet to your Oracle Communications Session Border Controller. Using remote Telnet access, you can provision the Oracle Communications Session Border Controller remotely through the management IP interface.

The Oracle Communications Session Border Controller, when running on Acme Packet platforms can support up to 5 concurrent Telnet sessions. When running on other platform types, only 4 concurrent Telnet sessions are available. In both cases, only one Telnet session may be in configuration mode at a time.



**Note:** Telnet does not offer a secure method of sending passwords. Using Telnet, passwords are sent in clear text across the network.

To Telnet to your Oracle Communications Session Border Controller, you need to know the IP address of its administrative interface (wancom0/eth0). The wancom0/eth0 IP address of your Oracle Communications Session Border Controller is found by checking the **inet on ethernet** value in the boot parameters or visible from the front panel display.

You can manage incoming Telnet connections from the ACLI:

- To set a time-out due to inactivity, use the **telnet-timeout** parameter in the system configuration. You can set the number of seconds that elapse before the Telnet connection or SSH connection is terminated. The default for this field is 0, which means that no time-out is being enforced.
- To view the users who are currently logged into the system, use the ACLI **show users** command. You can see the ID, timestamp, connection source, and privilege level for active connections.
- From Superuser mode in the ACLI, you can terminate the connections of other users in order to free up connections. Use the **kill user** command with the corresponding connection ID.
- From Superuser mode in the ACLI, you can globally enable and disable Telnet connections:



- Telnet service is enabled by default unless explicitly disabled as shipped.
- To disable Telnet, type the **management disable telnet** command at the Superuser prompt and reboot your system. The Oracle Communications Session Border Controller then refuses any attempts at Telnet connections. If you want to restart Telnet service, type **management enable telnet**.
- If you reboot your Oracle Communications Session Border Controller from a Telnet session, you lose IP access and therefore your connection.

## SSH Remote Connections

For increased security, you can connect to your Oracle Communications Session Border Controller using SSH. An SSH client is required for this type of connection.

The Oracle Communications Session Border Controller supports five concurrent SSH and/or SFTP sessions.

There are two ways to use SSH to connect to your Oracle Communications Session Border Controller. The first works the way a Telnet connection works, except that authentication takes place before the connection to the Oracle Communications Session Border Controller is made. The second requires that you set an additional password.

1. To initiate an SSH connection to the Oracle Communications Session Border Controller without specifying users and SSH user passwords:

- a) Open your SSH client (with an open source client, etc.).
- b) At the prompt in the SSH client, type the **ssh** command, a Space, the IPv4 address of your Oracle Communications Session Border Controller, and then press Enter.

The SSH client prompts you for a password before connecting to the Oracle Communications Session Border Controller. Enter the Oracle Communications Session Border Controller's User mode password. After it is authenticated, an SSH session is initiated and you can continue with tasks in User mode or enable Superuser mode.



**Note:** You can also create connections to the Oracle Communications Session Border Controller using additional username and password options.

2. To initiate an SSH connection to the Oracle Communications Session Border Controller with an SSH username and password:

- a) In the ACLI at the Superuser prompt, type the **ssh-password** and press Enter. Enter the name of the user you want to establish. Then enter a password for that user when prompted. Passwords do not appear on your screen.

```
ACMEPACKET# ssh-password
SSH username [saved]: MJones
Enter new password: 95X-SD
Enter new password again: 95X-SD
```

After you configure **ssh-password**, the SSH login accepts the username and password you set, as well as the default SSH/SFTP usernames: User and admin.

- b) Configure your SSH client to connect to your Oracle Communications Session Border Controller's management IPv4 address using the username you just created. The standard version of this command would be:

```
ssh -l MJones 10.0.1.57
```

- c) Enter the SSH password you set in the ACLI.

```
MJones@10.0.2.54 password: 95X-SD
```

- d) Enter your User password to work in User mode on the Oracle Communications Session Border Controller. Enable Superuser mode and enter your password to work in Superuser mode.
- e) A Telnet session window opens and you can enter your password to use the ACLI.

## System Boot

When your Oracle Communications Session Border Controller boots, the following information about the tasks and settings for the system appear in your terminal window.

## Getting Started

---

- System boot parameters
- From what location the software image is being loaded: an external device or internal flash memory
- Requisite tasks that the system is starting
- Log information: established levels and where logs are being sent
- Any errors that might occur during the loading process

After the loading process is complete, the ACLI login prompt appears.

## Oracle Communications Session Border Controller Boot Parameters

---

Boot parameters specify the information that your Oracle Communications Session Border Controller uses at boot time when it prepares to run applications. The Oracle Communications Session Border Controller's boot parameters:

- Allow you to set the IP address for the management interface (wancom0).
- Allow you to set a system prompt. The target name parameter also specifies the title name displayed in your web browser and SNMP device name parameters.
- Determine the software image to boot and from where the system boots that image.
- Sets up the username and password for network booting from an external FTP server.

In addition to providing details about the Oracle Communications Session Border Controller's boot parameters, this section explains how to view, edit, and implement them.

When displaying the boot parameters, your screen shows a help menu and the first boot parameter (boot device). Press Enter to continue down the list of boot parameters.

### Boot Parameters

- boot device—The boot device for the Oracle Communications Session Border Controller should be **eth0**.
- file name—The file name for the Oracle Communications Session Border Controller should start with **/boot/** (if local).

### Sample Oracle Communications Session Border Controller Boot Parameters

The full set of Oracle Communications Session Border Controller boot parameters appears like the ones in this sample:

```
ACMEPACKET(configure)# bootparam
'.' = clear field; '-' = go to previous field; ^D = quit
boot device           : eth0
processor number      : 0
host name             : acmepacket8
file name             : /boot/nnSC600.gz
inet on ethernet (e)  : 10.0.1.57:ffff0000
inet on backplane (b) : 0.0.0.0
host inet (h)         : 10.0.1.5
gateway inet (g)      : 10.0.0.1
user (u)              : user
ftp password (pw)     : password
flags (f)             : 0x08
target name (tn)      : acmesystem
startup script (s)    : 0
other (o)             :
NOTE: These changed parameters will not go into effect until reboot. Also, be aware that some boot parameters may also be changed through the PHY and Network Interface Configurations.
```

### Boot Parameter Definitions

The following table defines each of the Oracle Communications Session Border Controller's boot parameters.


Boot Parameter	Description
boot device	Management interface name and port number of the device from which an image is downloaded (e.g., wancom0 or eth0) from an external device.
processor number	Processor number on the backplane.
host name	Name of the boot host used when booting from an external device.
file name	Name of the image file to be booted; can be entered with the filename path.  If you are booting from the flash memory, this filename must always match the filename that you designate when you FTP the image from the source to the Oracle Communications Session Border Controller.  When booting from internal flash memory, this filename must start with /boot); for example, /boot/nnECZ710.bz.
inet on ethernet (e)	Internet address of the Oracle Communications Session Border Controller.  This field can have an optional subnet mask in the form inet_adrs:subnet_mask. If DHCP is used to obtain the parameters, lease timing information may also be present. This information takes the form of lease_duration:lease_origin and is appended to the end of the field.  In this parameter, the subnet mask ffff0000 = 255.255.0.0.  When you use the ACLI acquire-config command, this is the IPv4 address of the Oracle Communications Session Border Controller from which you will copy a configuration.
inet on backplane (b)	Internet address of the backplane interface, eth0.  This parameter can have an optional subnet mask and/or lease timing information, such as e (inet on ethernet) does.
host inet (h)	Internet address of the boot host used when booting from an external device.
gateway inet (g)	Internet address of the gateway to the boot host.  Leave this parameter blank if the host is on the same network.
user (u)	FTP username on the boot host.
ftp password (pw)	FTP password for the FTP user on the boot host.
flags (f)	Codes that signal the Oracle Communications Session Border Controller from where to boot. Also signals the Oracle Communications Session Border Controller about which file to use in the booting process. This sequence always starts with 0x (these flags are hexadecimal). The most common codes are:  0x08: Means that the system looks at the filename defined in the boot configuration parameters to determine where to boot from and what file to use. If the file name parameter contains /tffsX/filename, then the system boots off the flash memory (see options below). If the file name parameter just contains a filename, then the Oracle Communications Session Border Controller boots off the external host defined and looks for the filename in the /tftpboot directory on that host.  0x80008: Used for source routing.  If your requirements differ from what these flags allow, contact your Oracle customer support representative for further codes.

## Getting Started

Boot Parameter	Description
target name (tn)	Name of the Oracle Communications Session Border Controller as it appears in the system prompt. For example, ACMEPACKET> or ACMEPACKET#. You need to know the target name if you are setting up an HA node.  This name is required to be unique among Oracle Communications Session Border Controllers in your network. This name can be 64 characters or less.
startup script (s)	For Oracle use only.
other (o)	For Oracle use only.

## Boot Parameter Changes

You can access and edit boot parameters by using either the ACLI or by interrupting the system boot process.

 **Note:** Changes to boot parameters do not go into effect until you reboot the Oracle Communications Session Border Controller.

Oracle recommends that you use management port 0 (wancom0) as the boot interface, and that your management network is either:

- directly a part of your LAN for management port 0
- accessible through management port 0

Otherwise, your management messages may use an incorrect source address.

### Change Boot Parameters from the ACLI

To access and change boot parameters from the ACLI:

1. In Superuser mode, type `configure terminal`, and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `bootparam`, and press Enter. The boot device parameters display.

```
ACMEPACKET(configure)# bootparam  
'.' = clear field; '-' = go to previous field; ^D = quit  
boot device      : eth0
```

To navigate through the boot parameters, press Enter and the next parameter appears on the following line.

You can navigate through the entire list this way. To go back to a previous line, type a hyphen (-) and press Enter. Any value that you enter entirely overwrites the existing value and does not append to it.

3. To change a boot parameter, type the new value that you want to use next to the old value. For example, if you want to change the image you are using, type the new filename next to the old one. You can clear the contents of a parameter by typing a period and then pressing Enter.

```
ACMEPACKET(configure)# bootparam  
'.' = clear field; '-' = go to previous field; ^D = quit  
boot device      : eth0  
processor number : 0  
host name       : goose  
file name       : /boot/nnPCz100.gz /boot/nnPCz200.gz
```

When you have scrolled through all of the boot parameters, the system prompt for the configure terminal branch displays.

```
ACMEPACKET(configure)#
```


4. Exit the configure terminal branch.
5. Reboot the Oracle Communications Session Border Controller for the changes to take effect.

The ACLI **reboot** and **reboot force** commands initiate a reboot. With the **reboot** command, you must confirm that you want to reboot. With the **reboot force** command, you do not have to make this confirmation.

```
ACMEPACKET# reboot force
```

The Oracle Communications Session Border Controller completes the full booting sequence. If necessary, you can stop the auto-boot at countdown to fix any boot parameters.

If you configured boot parameters correctly, the system prompt displays and you can go ahead with configuration, management, or monitoring tasks.

 **Note:** If you configured the boot parameters incorrectly, the Oracle Communications Session Border Controller goes into a booting loop and displays an error message.

```
Error loading file: errno = 0x226.
Can't load boot file!!
```

Press the space bar to stop the loop. Correct the error in the boot parameter, and reboot the system.

## Change Boot Parameters by Interrupting a Boot in Progress

To access and change boot parameters by interrupting a boot in progress:

1. When the Oracle Communications Session Border Controller is in the process of booting, you can press the space bar on your keyboard to interrupt when you see the following message appear:

```
Press the space bar to stop auto-boot...
```

2. After you stop the booting process, you can enter the letter **p** to display the current parameters, the letter **c** to change the boot parameters or the **@** (at-sign) to continue booting.

```
[Acme Packet Boot]: c
'.' = clear field; '-' = go to previous field; ^D = quit
boot device      : wancom0
```

To navigate through the boot parameters, press Enter and the next parameter appears on the following line.

You can navigate through the entire list this way. To go back to a previous line, type a hyphen (-) and press Enter. Any value that you enter entirely overwrites the existing value and does not append to it.

3. To change a boot parameter, type the new value that you want to use next to the old value. For example, if you want to change the image you are using, type the new filename next to the old one.


```
ACMEPACKET(configure)# bootparam
'.' = clear field; '-' = go to previous field; ^D = quit
boot device      : wancom0
processor number  : 0
host name        : goose
file name        : /code/nnPCz100.bz /code/nnPCz200.bz
```

4. After you have scrolled through the complete list of boot parameters, you return to the boot prompt. To reboot with your changes taking effect, type **@** (the at-sign), and press Enter.

```
[Acme Packet Boot]: @
```

The Oracle Communications Session Border Controller completes the full booting sequence, unless there is an error in the boot parameters.

If you have configured boot parameters correctly, the system prompt displays and you can go ahead with configuration, management, or monitoring tasks.

 **Note:** If you have configured the boot parameters incorrectly, the Oracle Communications Session Border Controller goes into a booting loop and displays an error message.

```
Error loading file: errno = 0x226.
Can't load boot file!!
```

Press the space bar to stop the loop. Correct the error, and reboot your system.

### Acme Packet 6000 Boot Parameters

The Acme Packet 6000's boot parameters are slightly different than the Acme Packet 4500's boot parameters. Boot parameters specify the information your Acme Packet 6000 uses at boot time when it prepares to run the software image. The boot parameters:

- Show the Oracle Communications Session Border Controller's IPv4 address for the management interface (wancom0)
- Allow you to set a system prompt
- Determine the software image and its location
- Configures the external (T)FTP server used for a net boot

Configuring boot parameters has repercussions for the Oracle Communications Session Border Controller's physical and network interface configurations. When you configure these interfaces, you can set values that might override the boot parameters.

The following is a screen capture of the bootparam configuration list:

```
[Acme Boot]: p
Boot File      : /boot/bzImage-bones64
IP Address     : 172.44.12.89
VLAN          :
Netmask       : 255.255.0.0
Gateway       : 172.44.0.1
Host IP       :
FTP username   :
FTP password  :
Flags         : 0x00000030
Target Name   : ACMEPACKET
Console Device : COM1
Console Baudrate : 115200
Other         :

[Acme Boot]: ?
?             - print this list
@             - boot (load and go)
p             - print boot params
c             - change boot params
v             - print boot logo with version
r             - reboot
s             - show license information

Boot flags:
0x02 - enable kernel debug
0x04 - disable crashdumps
0x08 - disable bootloader countdown
0x10 - enable debug login
0x40 - use DHCP for wancom0
0x80 - use TFTP instead of FTP
```

**Boot File** — The name and path of the software image you are booting. Include the absolute path for a local boot from the local /boot volume and for a net boot when a path on the FTP server is needed.

**IP Address** — IP address of wancom0.

**VLAN** — VLAN of management network over which this address is accessed.



**Note:** VLANs over management interfaces are not supported on the Acme Packet 6000

**Netmask** — Netmask portion of the wancom0 IP Address.

**Gateway** — Network Gateway that this wancom0 interface uses.

Host IP — IP Address of FTP server to download and execute a software image from.

FTP Username — FTP Server Username.

FTP Password — FTP Server Password.

Flags — Available flags are printed at the end of the bootparam configuration. Only change this under the director of Oracle personnel.

Target Name — Sets the text to display at every system prompt.

Console Device — Set this to COM1

Console Baud Rate — The speed in bits per second which the console port operates at. It operates at 115200 BPS, 8 data bits, no stop bit, parity NONE.

Other — Unused.

---

## Setting Up System Basics

Before configuring and deploying the Oracle Communications Session Border Controller, you might want to establish some basic attributes such as a system prompt, new User and Superuser passwords, and NTP synchronization.

### New System Prompt

The ACLI system prompt is set in the boot parameters. To change it, access the boot parameters and change the **target name** value to make it meaningful within your network. The target name may be up to 38 characters. A value that identifies the system in some way is often helpful.

---

## Using the Oracle Communications Session Border Controller Image

The Oracle Communications Session Border Controller arrives with the most recent, manufacturing-approved run-time image installed on the flash memory. If you want to use this image, you can install the Oracle Communications Session Border Controller as specified in the *Acme Packet Hardware Installation Guide*, establish a connection to the Oracle Communications Session Border Controller, and begin to configure it. On boot up, the system displays information about certain configurations not being present. You can dismiss these displays and begin configuring the Oracle Communications Session Border Controller.

If you want to use an image other than the one installed on the Oracle Communications Session Border Controller when it arrives, you can use the information in this section to obtain and install the image.

### Software Images

Software image filenames for S-CZ7.0.2F1 and higher end with the .bz extension. They should be placed in the /boot volume.

### Obtaining a New Image

You can download a software image onto the Oracle Communications Session Border Controller platform from the following sources.

- Obtain an image from the FTP site and directory where you or your Oracle customer support representative placed the image. For example, this may be a special server that you use expressly for images and backups.
- Obtain an image from your Oracle customer support representative, who will transfer it to your system.

Regardless of the source, you can use FTP or SFTP to copy the image from the source to the Oracle Communications Session Border Controller.

### Copy an Image to the Oracle Communications Session Border Controller using FTP

The /boot directory on the Oracle Communications Session Border Controller has 32mb available, and operating system files of approximately 9mb each. Oracle recommends storing no more than two images at a time in this location. One of these should be the latest version. The /boot directory is used for the on-board system flash memory. If you do not put the image in this directory, the Oracle Communications Session Border Controller will not find it.



**Note:** You can also use SFTP.

To copy an image on your Oracle Communications Session Border Controller using FTP:

1. Go to the directory where the image is located.
2. Check the IP address of the Oracle Communications Session Border Controller's management port (wancom0). (You might think of this as a management address since it is used in the management of your Oracle Communications Session Border Controller.)
3. Create the connection to the Oracle Communications Session Border Controller. In your terminal window, type **ftp** and the IPv4 address of the Oracle Communications Session Border Controller management port (wancom0), and press Enter. Once a connection has been made, the system displays a confirmation note followed by the FTP prompt.
4. Enter your FTP username and FTP password information. The username is always user, and the password is the same as the one you use for the User mode login.
5. Go to the directory where you want to put the image.
6. From the FTP prompt, do the following:

- Change the directory to /boot.

```
ftp> cd /boot
```

- Invoke binary mode.

```
ftp> binary
```



**Note:** Be sure to use binary transfer mode. If you do not, all transfers will be corrupted.

- At the FTP prompt, enter the put command, a Space, the name of the image file, and press Enter.

```
ftp> put [file name]
```

The system displays confirmation that the connection is opening and that transfer is taking place.

- After the file transfer is complete, you can quit.

```
ftp> quit
```

7. Boot the Oracle Communications Session Border Controller using the image you just transferred.

In the ACLI, change any boot configuration parameters that need to be changed. It is especially important to change the filename boot parameter to the filename you used during the FTP process. Otherwise, your system will not boot properly.

Alternatively, from the console you can reboot to access the boot prompt and then configure boot parameters from there.

8. In the ACLI, execute the **save-config** command in order to save your changes.
9. Reboot the Oracle Communications Session Border Controller.
10. The Oracle Communications Session Border Controller runs through its loading processes and returns you to the ACLI logon prompt.

#### System Image Filename

The system image filename is a name you set for the image. This is also the filename the boot parameters uses when booting your system. This filename must match the filename specified in the boot parameters. When you use it in the



boot parameters, it should always start with `/boot` to signify that the Oracle Communications Session Border Controller is booting from the `/boot` directory.

If the filename set in the boot parameters does not point to the image you want sent to the Oracle Communications Session Border Controller via FTP, then you could not only fail to load the appropriate image, but you could also load an image from a different directory or one that is obsolete for your purposes. This results in a boot loop condition that you can fix by stopping the countdown, entering the appropriate filename, and rebooting the Oracle Communications Session Border Controller.

## Booting an Image on Your Oracle Communications Session Border Controller

You can either boot your Oracle Communications Session Border Controller from the system's local storage or from an external device. Both locations can store images from which the system can boot. This section describes both booting methods.

For boot parameters to go into effect, you must reboot your Oracle Communications Session Border Controller. Since a reboot stops all call processing, Oracle recommends performing tasks that call for a reboot during off-peak hours. If your Oracle Communications Session Border Controllers are set up in an HA node, you can perform these tasks on the standby system first.

### Booting from Flash Memory

Once you have installed an image, you can boot your Oracle Communications Session Border Controller from its flash memory. With the exception of testing an image before you install it on the flash memory, this is generally the method you use for booting.

To boot from your Oracle Communications Session Border Controller flash memory:

1. Confirm that the boot parameters are set up correctly, and make any necessary changes.

You can check the boot configuration parameters by accessing the `bootparam` command from the configure terminal menu.

```
ACMEPACKET# configure terminal
ACMEPACKET# bootparam
```

2. Change any boot configuration parameters that you need to change. It is especially important to change the `file name` boot configuration parameter. The file name parameter needs to use the `/boot` value so that the Oracle Communications Session Border Controller boots from the flash.
3. Reboot your Oracle Communications Session Border Controller.
4. You are returned to the CLI login prompt. To continue with system operations, enter the required password information.

### Booting from an External Device

Booting from an external device means that your Oracle Communications Session Border Controller connects to a server to retrieve the boot image at boot time. Rather than using an image stored on your system's flash memory, it downloads the image from the external device each time it reboots.

When you are testing a new image before putting it on your Oracle Communications Session Border Controller, you might want to boot from an external device. Ordinarily, you would not want to boot an image on your Oracle Communications Session Border Controller this way.

To boot an image from an external device:

1. Confirm that the Oracle Communications Session Border Controller is cabled to the network from which you are booting. This is port 0 on the rear panel of the Oracle Communications Session Border Controller chassis (wancom0). The image is loaded from the source using FTP.
2. Log into the system you want to mount.

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3. On the Oracle Communications Session Border Controller, configure the information for the boot parameters and confirm the following:
  - boot device—device to which you will FTP

This parameter value must contain the name of the applicable management interface, and then the number of the appropriate 10/100 port. Usually, this value is wancom0.
  - file name—name on the host of the file containing the image

The image file must exist in the home directory of the user on the image source.
  - host inet—IPv4 address of the device off of which you are booting
  - gateway inet—IPv4 address of the gateway to use if the device from which you are booting is not on the same network as your Oracle Communications Session Border Controller
  - user—username for the FTP account on the boot host
  - password—password for the FTP account on the boot host
4. Reboot your Oracle Communications Session Border Controller.
5. You are returned to the CLI login prompt. To continue with system operations, enter the required password information.

## Software Licensing

---

New users should migrate to the *Provisioning Entitlements* method of licensing their system(s). Please contact an Oracle representative for more information. Task descriptions for the legacy keyed method of licensing your system remain in this section. The following guidelines apply to Oracle software licenses:

- Each license is bound to a specific Oracle Communications Session Border Controller by serial number.
- Oracle does not allow transferring a license from one Oracle Communications Session Border Controller to another.
- Multiple licenses can be active on the same Oracle Communications Session Border Controller simultaneously.
- If a feature is covered by more than one license, the latest expiration date applies.
- You can activate and deactivate a license in real time.
- A license is fully extensible and upgradable.
- The system does not display the interface and configuration parameters for an unlicensed component.

Oracle software licenses are aggregate. When a new license is added to the original license set, the related capacity, protocol, or interface becomes part of the functionality that you can configure and deploy.

The only software component that requires a license is Administration Security. This software component enables the use of Administration Security features; installation and use of this feature set should be executed with care.

## Unlicensed Oracle Communications Session Border Controller

When your Oracle Communications Session Border Controller arrives, you must obtain a key to activate the licenses for the functionality you want to use. Obtain the license key from Oracle customer support at <http://www.oracle.com/us/support/licensecodes/index.html>.

If you log onto an unlicensed Oracle Communications Session Border Controller, the system displays a warning message that a valid license is required. Until you enter a valid license, you can configure general system parameters, but not parameters for protocols and features.

## Legacy - Obtaining a License

If you choose to add functionality to your Oracle Communications Session Border Controller, each new feature will require its own key. To obtain additional licenses for functions on your Oracle Communications Session Border Controller, contact your customer support or sales representative directly. You can request and purchase a license for the software you want, obtain a key for it, and then activate it on your Oracle Communications Session Border Controller.

When you obtain licenses, you need to provide Oracle with the serial number of your Oracle Communications Session Border Controller. You can see the system's serial number by using the ACLI **show version boot** command.

## Trial License

Oracle offers a trial license for software components to allow testing a feature before deployment.

A trial license is active for specified period of time. When the trial license expires, the feature stops responding and the system removes the configuration selections. To continue using the feature you must obtain a license. For more information about licensing, contact your Oracle sales representative or technical support representative.

## Standalone System Licensing

This section shows you how to add licenses and delete them from standalone Oracle Communications Session Border Controllers. The process for two systems making up an HA node is different, so follow the procedure relevant to your configuration.

### Adding a License to a Standalone System

Once you have obtained a license key, you can add it to your Oracle Communications Session Border Controller and activate it.

To add and activate a license on your Oracle Communications Session Border Controller:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **system** and press Enter.

```
ACMEPACKET (configure) # system
ACMEPACKET (system) #
```

3. Type **license** and press Enter.

```
ACMEPACKET (system) # license
ACMEPACKET (license) #
```

4. Using the **add** command and the key generated by Oracle, add the license to your Oracle Communications Session Border Controller.

```
ACMEPACKET (license) # add s125o39pvtqhas4v2r2jcl0aen9e01o21b1dmh3
```

5. You can check that the license has been added by using the ACLI **show** command within the license configuration.

```
ACMEPACKET (license) # show
1: High Availability
2: Accounting
3: SIP
4: H323
5: 250 sessions, ACP
6: QOS
ACMEPACKET (license) #
```

6. To activate your license, type the **activate-config** command and press Enter. The Oracle Communications Session Border Controller then enables any of the processes that support associated features.

```
ACMEPACKET# activate-config
```

### Deleting a License from a Standalone System

You can delete a license from your Oracle Communications Session Border Controller, including licenses that have not expired. If you want to delete a license that has not expired, you need to confirm the deletion.

To delete a license from the Oracle Communications Session Border Controller:

## Getting Started

---

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type `system` and press Enter.

```
ACMEPACKET(configure)# system
ACMEPACKET(system)#
```

3. Type `license` and press Enter.

```
ACMEPACKET(system)# license
ACMEPACKET(license)#
```

4. Type the `no` command and press Enter. A list of possible licenses to delete appears.

```
ACMEPACKET(license)# no
feature:
1: High Availability
2: Accounting
3: SIP
4: H323
5: 250 sessions, ACP
6: QOS
selection:
ACMEPACKET(license)#
```

5. Type the number corresponding to the license you want to delete and press Enter.

```
selection:6
```

6. If the license has not expired, you are asked to confirm the deletion.

```
Delete unexpired license [y/n]?: y
ACMEPACKET(license)#
```

When you show the licenses, the one you deleted should no longer appear on the list.

7. To clear the license from the system, type the `activate-config` command and press Enter. The Oracle Communications Session Border Controller then disables any of the processes that support associated features.

```
ACMEPACKET# activate-config
```

## High Availability (HA) Pair Licensing

When adding and deleting licenses, you must perform the procedure across both members of an HA pair during the same service window. The licenses on each peer must be identical. Peers with mismatched licenses may exhibit unexpected behavior.

### Adding a License to an HA Node

To add a license to both systems in an HA node, you add your licenses to both systems and reboot both systems when they are in standby mode. The process requires that you carefully confirm system synchronization in between various steps.

This procedure uses the designations Oracle Communications Session Border Controller1 as the original active and Oracle Communications Session Border Controller2 as the original standby.

To add a license on systems in an HA node:

1. Confirm that Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 are synchronized.

You must make sure that all of the running and current configurations on Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 have the same number. In the examples below, all of the configuration versions are 5.

- On Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2, use the CLI `show health` command to make sure that all processes are synchronized.

- On Oracle Communications Session Border Controller1, show the current configuration version by using the ACLI `display-current-cfg-version` command. Then use the same command on Oracle Communications Session Border Controller2 and be sure that its current configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-current-cfg-version
Current configuration version is 5
ACMEPACKET1#
ACMEPACKET2# display-current-cfg-version
Current configuration version is 5
ACMEPACKET2#
```

- On Oracle Communications Session Border Controller1, show the running configuration version by using the ACLI `display-running-cfg-version` command. Then use the same command on Oracle Communications Session Border Controller2 and be sure that its running configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-running-cfg-version
Running configuration version is 5
ACMEPACKET1#
ACMEPACKET2# display-running-cfg-version
Running configuration version is 5
ACMEPACKET2#
```

2. Now you can add a license to SBC1. To begin, type `configure terminal` and press Enter.

```
ACMEPACKET1# configure terminal
ACMEPACKET1(configure)#
```

3. Type `system` and press Enter.

```
ACMEPACKET1(configure)# system
ACMEPACKET1(system)#
```

4. Type `license` and press Enter.

```
ACMEPACKET1(system)# license
ACMEPACKET1(license)#
```

5. Using the `add` command and the key generated by Oracle, add the license to your Oracle Communications Session Border Controller.

```
ACMEPACKET1(license)# add sjk14i45987p43hh0938hnhjlaie10983
```

6. You can check that the license has been added by using the ACLI `show` command within the license configuration.

```
ACMEPACKET(license)# show
1: High Availability
2: Accounting
3: SIP
4: H323
5: 250 sessions, ACP
6: QOS
ACMEPACKET(license)#
```

7. Repeat typing `exit`, pressing Enter after each entry, until you reach the main Superuser prompt.

```
ACMEPACKET1(license)# exit
ACMEPACKET1(system)# exit
ACMEPACKET1(configure)# exit
ACMEPACKET1#
```

8. Repeat steps 2 through 7 on SBC2.

```
ACMEPACKET2(license)# add [License for SBC2]
```

At the end of this step, licenses are installed and verified on both SBC1 and SBC2.

9. Return to SBC1 and type the `save-config` command and press Enter.

```
ACMEPACKET1# save-config
```

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10. Type the **activate-config** command and press Enter. The Oracle Communications Session Border Controller then enables any of the processes that support associated features.

```
ACMEPACKET1# activate-config
```

11. Confirm that Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 are synchronized.

You must make sure that all of the running and current configurations on Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 have the same number. In the examples below, all of the configuration versions are 6.

- On Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2, use the ACLI **show health** command to make sure that all processes are synchronized.
- On Oracle Communications Session Border Controller1, show the current configuration version by using the ACLI **display-current-cfg-version** command. Then use the same command on Oracle Communications Session Border Controller2 and be sure that its current configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-current-cfg-version
Current configuration version is 6
ACMEPACKET1#
ACMEPACKET2# display-current-cfg-version
Current configuration version is 6
ACMEPACKET2#
```

- On Oracle Communications Session Border Controller1, show the running configuration version by using the ACLI **display-running-cfg-version** command. Then use the same command on Oracle Communications Session Border Controller2 and be sure that its running configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-running-cfg-version
Running configuration version is 6
ACMEPACKET1#
ACMEPACKET2# display-running-cfg-version
Running configuration version is 6
ACMEPACKET2#
```

12. Execute the ACLI **show health** command to make sure that all processes are synchronized.

13. Return to SBC2 and execute the reboot command.

```
ACMEPACKET2# reboot
```

14. Reconfirm that Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 are synchronized.

You must make sure that all of the running and current configurations on Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 have the same number. In the examples below, all of the configuration versions are 6.

- On Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2, use the ACLI **show health** command to make sure that all processes are synchronized.
- On Oracle Communications Session Border Controller1, show the current configuration version by using the ACLI **display-current-cfg-version** command. Then use the same command on Oracle Communications Session Border Controller2 and be sure that its current configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-current-cfg-version
Current configuration version is 6
ACMEPACKET1#
ACMEPACKET2# display-current-cfg-version
Current configuration version is 6
ACMEPACKET2#
```

- On Oracle Communications Session Border Controller1, show the running configuration version by using the ACLI **display-running-cfg-version** command. Then use the same command on Oracle Communications

Session Border Controller2 and be sure that its running configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-running-cfg-version
Running configuration version is 6
ACMEPACKET1#
ACMEPACKET2# display-running-cfg-version
Running configuration version is 6
ACMEPACKET2#
```

15. Trigger a switchover between the two systems in the HA node so the originally standby system assumes the active role. This means that the standby system will transition to active.

```
ACMEPACKET1# notify berpd force
```

16. Wait for Oracle Communications Session Border Controller2 to transition to the active state. Confirm that it is in the active state by using the ACLI **show health** command.

17. Reconfirm that Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 are synchronized.

You must also make sure that all of the running and current configurations on Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 have the same number. In the examples below, all of the configuration versions are 6.

- On Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2, use the ACLI **show health** command to make sure that all processes are synchronized.
- On Oracle Communications Session Border Controller2, show the current configuration version by using the ACLI **display-current-cfg-version** command. Then use the same command on Oracle Communications Session Border Controller1 and be sure that its current configuration version is the same as the one on Oracle Communications Session Border Controller2.

```
ACMEPACKET2# display-current-cfg-version
Current configuration version is 6
ACMEPACKET2#
ACMEPACKET1# display-current-cfg-version
Current configuration version is 6
ACMEPACKET1#
```

- On Oracle Communications Session Border Controller2, show the running configuration version by using the ACLI **display-running-cfg-version** command. Then use the same command on Oracle Communications Session Border Controller1 and be sure that its running configuration version is the same as the one on Oracle Communications Session Border Controller2.

```
ACMEPACKET2# display-running-cfg-version
Running configuration version is 6
ACMEPACKET2#
ACMEPACKET1# display-running-cfg-version
Running configuration version is 6
ACMEPACKET1#
```

18. Return to SBC1 and execute the reboot command.

```
ACMEPACKET1# reboot
```

19. Wait for SBC1 to complete rebooting. When finished, execute the ACLI **show health** command to make sure that all processes are synchronized.

At this point both SBCs should be synchronized and contain the same license configuration.


20. If desired, trigger a switchover between the two systems in the HA node so the originally active system (SBC1) assumes the active role.

```
ACMEPACKET2# notify berpd force
```

At this point both SBCs should be synchronized and contain the same license configuration.

### Deleting a License from an HA Node

To delete a license from both systems in an HA node, you remove your licenses from both systems and reboot both systems when they are in standby mode. The process requires that you carefully confirm system synchronization in between various steps.

 **Note:** Licenses should be deleted on both nodes during the same service window.

This procedure uses the designations Oracle Communications Session Border Controller1 as the original active and Oracle Communications Session Border Controller2 as the original standby.

To delete a license from systems in an HA node:

1. Confirm that Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 are synchronized.

You must make sure that all of the running and current configurations on Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 have the same number. In the examples below, all of the configuration versions are 7.

- On Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2, use the ACLI show health command to make sure that all processes are synchronized.
- On Oracle Communications Session Border Controller1, show the current configuration version by using the ACLI display-current-cfg-version command. Then use the same command on Oracle Communications Session Border Controller2 and be sure that its current configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-current-cfg-version
Current configuration version is 7
ACMEPACKET1#
ACMEPACKET2# display-current-cfg-version
Current configuration version is 7
ACMEPACKET2#
```

- On Oracle Communications Session Border Controller1, show the running configuration version by using the ACLI display-running-cfg-version command. Then use the same command on Oracle Communications Session Border Controller2 and be sure that its running configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-running-cfg-version
Running configuration version is 7
ACMEPACKET1#
ACMEPACKET2# display-running-cfg-version
Running configuration version is 7
ACMEPACKET2#
```

2. Now you can delete a license from SBC1. To begin, type **configure terminal** and press Enter.

```
ACMEPACKET1# configure terminal
ACMEPACKET1(configure)#
```

3. Type **system** and press Enter.

```
ACMEPACKET1(configure)# system
```

4. Type **license** and press Enter.

```
ACMEPACKET1(system)# license
ACMEPACKET1(license)#
```

5. Type the **no** command and press Enter. A list of possible license to delete appears.

```
ACMEPACKET(license)# no
1: High Availability
2: Accounting
3: SIP
4: H323
5: 250 sessions, ACP
```



```
6: QOS
selection#
```

- Type the number corresponding to the license you want to delete and press Enter.

```
selection:6
```

- If the license has not expired, you are be asked to confirm the deletion.

```
Delete unexpired license [y/n]?: y
ACMEPACKET1(license)#
```

When you show the licenses, the one you deleted should no longer appear on the list.

- Repeat typing **exit**, pressing Enter after each entry, until you reach the main Superuser prompt.

```
ACMEPACKET1(license)# exit
ACMEPACKET1(system)# exit
ACMEPACKET1(configure)# exit
ACMEPACKET1#
```

- Repeat steps 2 through 8 on SBC2.

```
ACMEPACKET(license)# no
feature:
1: High Availability
2: Accounting
3: SIP
4: H323
5: 250 sessions, ACP
6: QOS
selection#
```

At the end of this step, licenses are removed and verified on both SBC1 and SBC2.

- Return to SBC1 and type the **save-config** command and press Enter.

```
ACMEPACKET1# save-config
```

- Type the **activate-config** command and press Enter.

```
ACMEPACKET1# activate-config
```

- Confirm that Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 are synchronized.

You must make sure that all of the running and current configurations on Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 have the same number. In the examples below, all of the configuration versions are 5.

- On Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2, use the ACLI show health command to make sure that all processes are synchronized.
- On Oracle Communications Session Border Controller1, show the current configuration version by using the ACLI display-current-cfg-version command. Then use the same command on Oracle Communications Session Border Controller2 and be sure that its current configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-current-cfg-version
Current configuration version is 7
ACMEPACKET1#
ACMEPACKET2# display-current-cfg-version
Current configuration version is 7
ACMEPACKET2#
```

- On Oracle Communications Session Border Controller1, show the running configuration version by using the ACLI display-running-cfg-version command. Then use the same command on Oracle Communications Session Border Controller2 and be sure that its running configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-running-cfg-version
Running configuration version is 7
```

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```
ACMEPACKET1#  
ACMEPACKET2# display-running-cfg-version  
Running configuration version is 7  
ACMEPACKET2#
```

13. Execute the ACLI **show health** command to make sure that all processes are synchronized.

14. Return to SBC2 and execute the reboot command.

```
ACMEPACKET2# reboot
```

15. Reconfirm that Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 are synchronized.

You must make sure that all of the running and current configurations on Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 have the same number. In the examples below, all of the configuration versions are 7.

- On Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2, use the ACLI **show health** command to make sure that all processes are synchronized.
- On Oracle Communications Session Border Controller1, show the current configuration version by using the ACLI **display-current-cfg-version** command. Then use the same command on Oracle Communications Session Border Controller2 and be sure that its current configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-current-cfg-version  
Current configuration version is 7  
ACMEPACKET1#  
ACMEPACKET2# display-current-cfg-version  
Current configuration version is 7  
ACMEPACKET2#
```

- On Oracle Communications Session Border Controller1, show the running configuration version by using the ACLI **display-running-cfg-version** command. Then use the same command on Oracle Communications Session Border Controller2 and be sure that its running configuration version is the same as the one on Oracle Communications Session Border Controller1.

```
ACMEPACKET1# display-running-cfg-version  
Running configuration version is 7  
ACMEPACKET1#  
ACMEPACKET2# display-running-cfg-version  
Running configuration version is 7  
ACMEPACKET2#
```

16. Trigger a switchover between the two systems in the HA node so the originally standby system assumes the active role. This means that the originally standby system will transition to active.

```
ACMEPACKET1# notify berpd force
```

17. Wait for Oracle Communications Session Border Controller2 to transition to the active state. Confirm that it is in the active state by using the ACLI **show health** command.

18. Reconfirm that Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 are synchronized.

You must also make sure that all of the running and current configurations on Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2 have the same number. In the examples below, all of the configuration versions are 8.

- On Oracle Communications Session Border Controller1 and Oracle Communications Session Border Controller2, use the ACLI **show health** command to make sure that all processes are synchronized.
- On Oracle Communications Session Border Controller2, show the current configuration version by using the ACLI **display-current-cfg-version** command. Then use the same command on Oracle Communications Session Border Controller1 and be sure that its current configuration version is the same as the one on Oracle Communications Session Border Controller2.

```
ACMEPACKET2# display-current-cfg-version  
Current configuration version is 7
```

```
ACMEPACKET2#
ACMEPACKET1# display-current-cfg-version
Current configuration version is 7
ACMEPACKET1#
```

- On Oracle Communications Session Border Controller2, show the running configuration version by using the ACLI `display-running-cfg-version` command. Then use the same command on Oracle Communications Session Border Controller1 and be sure that its running configuration version is the same as the one on Oracle Communications Session Border Controller2.

```
ACMEPACKET2# display-current-cfg-version
Current configuration version is 7
ACMEPACKET2#
ACMEPACKET1# display-current-cfg-version
Current configuration version is 7
ACMEPACKET1#
```

19. Return to SBC1 and execute the reboot command.

```
ACMEPACKET1# reboot
```

20. Wait for SBC1 to complete rebooting. When finished, execute the ACLI `show health` command to make sure that all processes are synchronized.
21. If desired, trigger a switchover between the two systems in the HA node so the originally active system (SBC1) assumes the active role.

```
ACMEPACKET2# notify berpd force
```

At this point both SBCs should be synchronized and contain the same license configuration.

## License Expiration

When a license expires, you are no longer able to use the features associated with it. The Oracle Communications Session Border Controller automatically disables all associated processes.

To avoid a license unexpectedly expiring and therefore potentially disrupting service, we recommend that you track expiration dates and renew licenses well in advance of expiration.

Expired licenses appear in your Oracle Communications Session Border Controller ACLI displays until you delete them, though you cannot use the features associated with them. Deleting an expired license requires that you take the same steps as you do for deleting a valid one.

## Viewing Licenses

There are two ways to view licenses in the ACLI.

- You can use the `show features` command at the main ACLI user prompt.

```
ACMEPACKET# show features
Total session capacity: 2250
Enabled protocols: SIP, H.323, IWF
Enabled features: ACP
ACMEPACKET#
```

- Within the license menu, use the `show` command to see all licenses with detailed information.

```
ACMEPACKET(license)# show
License #1: 2000 sessions, SIP, ACP
           no expiration
           installed at 12:34:42 APR 01 2005
License #2: H323
           expired at 23:59:59 APR 08 2005
           installed at 12:35:43 APR 01 2005
License #3: 250 sessions, IWF
           expires at 23:59:59 APR 28 2005
           installed at 12:36:44 APR 01 2005
License #4: QOS
```

```
starts at 00:00:00 APR 08 2004
expires at 23:59:59 OCT 27 2005
installed at 12:37:45 APR 01 2005
Total session capacity: 2250
ACMEPACKET(license)#
```

### Granularity and Oversubscription Limits

Only on the Acme Packet 3810, the Oracle Communications Session Border Controller uses a 10-to-1 oversubscription limit, meaning that the system allows ten registrations for a single licensed session. The system enforces the limits across all signalling protocols.

An SNMP OID, `apSysRegistrationCapacity`, supports querying the percentage of used registration capacity. When the percentage approaches the registration capacity limit, an alarm triggers and the Acme Packet 3810 sends an SNMP trap.

- SIP—For SIP, the 10-to-1 ratio limits has possible implications for the SIP registrations cache limiting feature. When you enable that feature, the Oracle Communications Session Border Controller rejects new registrations when they exceed the configurable registration cache limit. Likewise, the system can reject registrations when they exceed the global oversubscription limit. It uses whichever is the lower of the two.

The Acme Packet 3810 first checks the configurable registrations cache limits. If you have configured this value to be higher than the global oversubscription limits, the Acme Packet 3810 leaves the registration cache limit value intact. However, if registrations go over the global oversubscription limit, the Acme Packet 3810 will reject them, regardless of the cache limit, and the corresponding traps and alarms might not be triggered.

- H.323—The Acme Packet 3800 tracks the number of `CallSignalingAddress` records as a means of counting registrations. This methods relies on each endpoint having a unique `CallSignalingAddress`.

### SNMP Support for Global Registration Capacity

For the Acme Packet 3800 only, you can use the `apSysRegistrationCapacity` object to query the percentage of used global registration capacity on your system. This object and corresponding group are now part of the `apSystemManagement` MIB, `ap-smgmt.mib`. The OID and its value are also sent as parameters in the `apSysMgmtGroupTrap` when an alarm condition occurs. The alarm for this condition is `SYS_REG_OVER_THRESHOLD` with these values: `0x0002003A` (hexadecimal) and `131130` (decimal).

The alarm condition depends on whether or not you have set any alarm thresholds for the session type in the system configuration.

- If you have configured them, the thresholds apply to registration capacity. The registration capacity alarm uses the same percentage values and severities for the alarm as those set for the session alarm thresholds.
- If you have not configured them, then the registration capacity alarm triggers at 90%.

The alarm clears when two successive checks, performed once every five seconds, report a value under the threshold.

### Denial of Service Feature Group

For the Acme Packet 3800 only, a denial of service (DoS) license now exists. When the DoS license is not present, certain whole configurations and specific parameters within unrestricted configurations related to DoS functionality are not available. You can neither configure them, nor can you see them when you use the ACLI `show configuration` command.

The table below details the restrictions.

Restricted Configuration Element	Restricted Parameters
access-control	realm-id source-address destination-address application-protocol

Restricted Configuration Element	Restricted Parameters
	transport-protocol access average-rate-limit trust-level invalid-signal-threshold maximum-signal-threshold untrusted-signal-threshold deny-period
media-manager	max-signaling-bandwidth max-untrusted-signaling min-untrusted-signaling fragment-msg-bandwidth tolerance-window arp-msg-bandwidth rtcp-rate-limit
media-profile	average-rate-limit
realm-config	average-rate-limit access-control-trust-level invalid-signal-threshold maximum-signal-threshold untrusted-signal-threshold nat-trust-threshold deny-period
static-flow	average-rate-limit

## System Provisioning

The majority of the Oracle Communications Session Border Controller's capacities and features are self provisioned at the ACLI. Initial system configuration consists of setting the system type and available features. These choices determine the feature groups that are available for use and the maximum session capacities available for certain features. The system provisions a set of default entitlements based on the software version, the hardware on which the software runs, and the product type that you choose.

### Provisioning a New System


An uninitialized system does not have the product type set, nor does it have any features installed. A system that has legacy keyed licenses installed is considered an uninitialized system.

You may not enable features until the product type has been set. Use the **setup product** command to set the system product type and use the **setup entitlements** command to enable features.

### Controlled Features

Certain features are enabled only by entering a license key in the **system** > **license** configuration element, and not through the **setup entitlements** command. Contact your sales representative for more information on purchasing these features.

The following controlled features are available:

 **Note:** Not all of the above controlled features are available on all platforms.

### Provisioning a System with Existing Keyed Licenses

When changing the Oracle Communications Session Border Controller licensing technique from the legacy keyed licenses method to the self-provisioned method, be aware of the following:

- After running **setup product** and **setup entitlements**, the system will be seeded with the existing licenses' functionality.
- When the system is seeded with its previous functionality to the provisioned entitlements system, functionality may be changed with the **setup entitlements** command.
- You may notice that there are fewer entitlements than there were keyed licenses. This is normal.
- After setting up provisioned features, the **show features** command will still function to display the previously installed keyed entitlements.

### Product Configuration

1. Type **setup product** at the ACLI. If this is the first time running the command on this hardware, the product will show as Uninitialized.
2. Type **1** <Enter> to modify the uninitialized product.
3. Type the number followed by <Enter> for the product type you wish to initialize.
4. Type **s** <Enter> to commit your choice as the product type of this platform.

### Entitlements Configuration

1. Type **setup entitlements** at the ACLI. Currently provisioned entitlements are printed on the screen.
2. Type the number of the entitlement you wish to configure followed by pressing the <Enter> Key. Some entitlements are set as enabled/disabled (provisionable feature entitlement), and some entitlements are provisioned with a maximum capacity value (provisionable capacity entitlement). The command will let you provision these values as appropriate.
3. Type **enabled** or **disabled** to set a provisionable feature entitlement, or type an integer value for a provisionable capacity entitlement. Both input types are followed by pressing the <Enter> key.
4. Repeat steps 2 and 3 to configure additional entitlements.
5. Type **d** followed by the <Enter> key to review the full range of your choices. Note that disabled entitlements display their state as blank.
6. Type **s** followed by the <Enter> key to commit your choice as an entitlement for your system. After saving the value succeeds you will be returned to the ACLI.
7. Reboot your Oracle Communications Session Border Controller.

```
ACMEPACKET# setup entitlements
-----
Entitlements for Session Border Controller
Last Modified: 2014-06-01 18:57:55
-----
 1 : Session Capacity           : 32000
 2 :      Accounting             : enabled
```

```

3 : IPv4 - IPv6 Interworking :
4 : IWF (SIP-H323) : enabled
5 : Load Balancing : enabled
6 : Policy Server : enabled
7 : Quality of Service : enabled
8 : Routing : enabled
9 : SIPREC Session Recording : enabled
10: Admin Security with ACP/NNC :
11: IMS-AKA Endpoints : 0
12: IPSec Trunking Sessions : 0
13: MSRP B2BUA Sessions : 0
14: SRTP Sessions : 0

Enter 1 - 14 to modify, d' to display, 's' to save, 'q' to exit.
[s]: s
SAVE SUCCEEDED

```

## Keyed Licenses and Provisioned Entitlements Compatibility

The Oracle Communications Session Border Controller supports keyed licenses and the provisioned entitlements process for provisioning features on the system. Know the following about how licensing mechanisms work with each other.

- You are not required to begin using the provisioned entitlement system as of this release.
- You may continue to obtain keyed licenses and install them as necessary.
- Upon migrating to the provisioned entitlements system, the current range of your installed, keyed licenses will be seeded to the provisioned entitlements system. The system's functionality will remain identical.
- If you upgrade to the provisioned entitlements system, then downgrade the software to a version without the provisioned entitlements system, the pre-existing keyed licenses will still function.
- If you upgrade to the provisioned entitlements system, then change the functionality (such as, adding more SIP sessions or removing a feature set), upon downgrade to a pre-provisioned entitlements image, the new functionality will not be present.

## RADIUS Authentication

A security feature that extends beyond the designation of ACLI User and Superuser privileges, the User Authentication and Access control feature supports authentication using your RADIUS server(s). In addition, you can set two levels of privilege, one for all privileges and more limited set that is read-only.

User authentication configuration also allows you to use local authentication, localizing security to the Oracle Communications Session Border Controller ACLI log-in modes. These modes are User and Superuser, each requiring a separate password.

The components involved in the RADIUS-based user authentication architecture are the Oracle Communications Session Border Controller and your RADIUS server(s). In these roles:

- The Oracle Communications Session Border Controller restricts access and requires authentication via the RADIUS server; the Oracle Communications Session Border Controller communicates with the RADIUS server using either port 1812 or 1645, but does not know if the RADIUS server listens on these ports
- Your RADIUS server provides an alternative method for defining Oracle Communications Session Border Controller users and authenticating them via RADIUS; the RADIUS server supports the VSA called ACME\_USER\_CLASS, which specifies what kind of user is requesting authentication and what privileges should be granted.

The Oracle Communications Session Border Controller also supports the use of the Cisco Systems Inc.™ Cisco-AVPair vendor specific attribute (VSA). This attribute allows for successful administrator login to servers that do not support the Oracle authorization VSA. While using RADIUS-based authentication, the Oracle Communications Session Border Controller authorizes you to enter Superuser mode locally even when your

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RADIUS server does not return the ACME\_USER\_CLASS VSA or the Cisco-AVPair VSA. For this VSA, the Vendor-ID is 1 and the Vendor-Type is 9. The list below shows the values this attribute can return, and the result of each:

- shell:priv-lvl=15—User automatically logged in as an administrator
- shell:priv-lvl=1—User logged in at the user level, and not allowed to become an administrator
- Any other value—User rejected

When RADIUS user authentication is enabled, the Oracle Communications Session Border Controller communicates with one or more configured RADIUS servers that validates the user and specifies privileges. On the Oracle Communications Session Border Controller, you configure:

- What type of authentication you want to use on the Oracle Communications Session Border Controller
- If you are using RADIUS authentication, you set the port from which you want the Oracle Communications Session Border Controller to send messages
- If you are using RADIUS authentication, you also set the protocol type you want the Oracle Communications Session Border Controller and RADIUS server to use for secure communication


Although most common set-ups use two RADIUS servers to support this feature, you are allowed to configure up to six. Among other settings for the server, there is a class parameter that specifies whether the Oracle Communications Session Border Controller should consider a specific server as primary or secondary. As implied by these designation, the primary servers are used first for authentication, and the secondary servers are used as backups. If you configure more than one primary and one secondary server, the Oracle Communications Session Border Controller will choose servers to which it sends traffic in a round-robin strategy. For example, if you specify three servers are primary, the Oracle Communications Session Border Controller will round-robin to select a server until it finds an appropriate one; it will do the same for secondary servers.

The VSA attribute assists with enforcement of access levels by containing one of the three following classes:

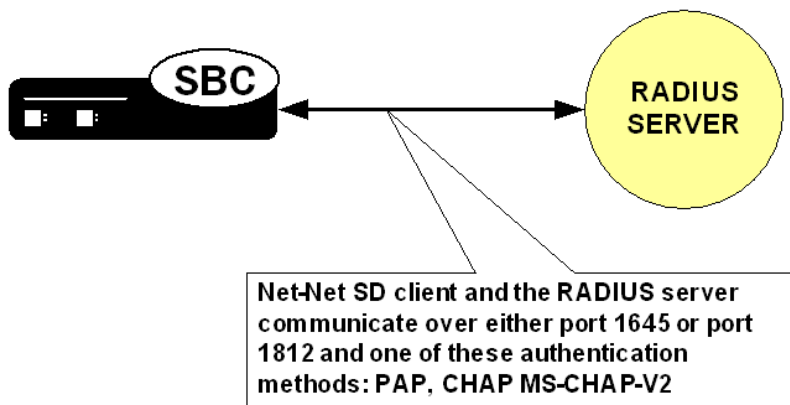
- None—All access denied
- User—Monitoring privileges are granted; your user prompt will resemble ACMEPACKET>
- Admin—All privileges are granted (monitoring, configuration, etc.); your user prompt will resemble ACMEPACKET#

Once it has selected a RADIUS server, the Oracle Communications Session Border Controller initiates communication and proceeds with the authentication process. The authentication process between the Oracle Communications Session Border Controller and the RADIUS server takes place uses one of three methods, all of which are defined by RFCs:

Protocol	RFC
PAP (Password Authentication Protocol)	B. Lloyd and W. Simpson, PPP Authentication Protocols, RFC 1334, October 1992
CHAP (Challenge Handshake Authentication Protocol)	B. Lloyd and W. Simpson, PPP Authentication Protocols, RFC 1334, October 1992 W. Simpson, PPP Challenge Handshake Authentication Protocol (CHAP), RFC 1994, August 1996
MS-CHAP-V2	G. Zorn, Microsoft PPP CHAP Extensions, Version 2, RFC 2759, January 2000

 **Note:** MS-CHAP-V2 support includes authentication only; password exchange is not supported or allowed on the Oracle Communications Session Border Controller.





## PAP Handshake

For PAP, user credentials are sent to the RADIUS server include the user name and password attribute. The value of the User-Password attribute is calculated as specified in RFC 2865.

### PAP Client Request Example

```
Radius Protocol
Code: Access Request (1)
Packet identifier: 0x4 (4)
Length: 61
Authenticator: 0x0000708D000002C5900002EB600003F37
Attribute value pairs
  t:User Name(1) l:11, value:"TESTUSER1"
    User-Name: TESTUSER1
  t:User Password (2) l:18, value:739B3A0F25094E4B3CDA18AB69EB9E4
  t:NAS IP Address(4) l:6, value:168.192.68.8
    Nas IP Address: 168.192.68.8(168.192.68.8)
  t:NAS Port(5) l:6, value:118751232
```

### PAP RADIUS Response

```
Radius Protocol
Code: Access Accept (2)
Packet identifier: 0x4 (4)
Length: 20
Authenticator: 0x36BD589C1577FD11E8C3B5BB223748
```

## CHAP Handshake

When the authentication mode is CHAP, the user credentials sent to the RADIUS server include “username,” “CHAP-Password,” and “CHAP-Challenge.” The “CHAP-Password” credential uses MD-5 one way. This is calculated over this series of the following values, in this order: challenge-id (which for the Oracle Communications Session Border Controller is always 0), followed by the user password, and then the challenge (as specified in RFC 1994, section 4.1).

### CHAP Client Request Example

```
Radius Protocol
Code: Access Request (1)
Packet identifier: 0x5 (5)
Length: 80
Authenticator: 0x0000396C000079860000312A00006558
Attribute value pairs
  t:User Name(1) l:11, value:"TESTUSER1"
    User-Name: TESTUSER1
  t:CHAP Password (3) l:19, value:003D4B1645554E881231ED7A137DD54FBF
  t:CHAP Challenge (60) l:18, value: 000396C000079860000312A00006558
```

## Getting Started

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```
t:NAS IP Address(4) l:6, value:168.192.68.8
  Nas IP Address: 168.192.68.8(168.192.68.8)
t:NAS Port(5) l:6, value:118751232
```

### CHAP RADIUS Response

```
Radius Protocol
Code: Access Accept (2)
Packet identifier: 0x4 (4)
Length: 20
Authenticator: 0x3BE89EED1B43D91D80EB2562E9D65392
```

## MS-CHAP-v2 Handshake

When the authentication method is MS-CHAP-v2, the user credentials sent to the RADIUS server in the Access-Request packet are:

- username
- MS-CHAP2-Response—Specified in RFC 2548, Microsoft vendor-specific RADIUS attributes
- MS-CHAP2-Challenge—Serves as a challenge to the RADIUS server

If the RADIUS authentication is successful, the Access-Accept packet from the RADIUS server must include an MS-CHAP2-Success attribute calculated using the MS-CHAP-Challenge attribute included in the Access-Request. The calculation of MS-CHAP2-Success must be carried out as specified in RFC 2759. The Oracle Communications Session Border Controller verifies that the MS-CHAP2-Success attribute matches with the calculated value. If the values do not match, the authentication is treated as a failure.

### MS-CHAP-v2 Client Request Example

Some values have been abbreviated.

```
Radius Protocol
Code: Access Request (1)
Packet identifier: 0x5 (5)
Length: 80
Authenticator: 0x0000024C000046B30000339F00000B78
Attribute value pairs
  t:User Name(1) l:11, value:"TESTUSER1"
    User-Name: TESTUSER1
  t:Vendor Specific(26) l:24, vendor:Microsoft(311)
  t:MS CHAP Challenge(11) l:18, value:0000024C000046B30000339F00000B78
  t:Vendor Specific(26) l:58, vendor:Microsoft(311)
  t:MS CHAP2 Response(25) l:52, value:
0000000024C000046B30000339F00000B78...
  t:NAS IP Address(4) l:6, value:168.192.68.8
    Nas IP Address: 168.192.68.8(168.192.68.8)
  t:NAS Port(5) l:6, value:118751232
```

### MS-CHAP-v2 RADIUS Response

```
Radius Protocol
Code: Access Accept (2)
Packet identifier: 0x6 (6)
Length: 179
Authenticator: 0xECB4E59515AD64A2D21FC6D5F14D0CC0
Attribute value pairs
  t:Vendor Specific(26) l:51, vendor:Microsoft(311)
    t:MS CHAP Success(11) l:45, value:003533s33d3845443532443135453846313...
  t:Vendor Specific(26) l:42, vendor:Microsoft(311)
    t:MS MPPE Recv Key(17) l:36, value:96C6325D22513CED178F770093F149CBBA...
  t:Vendor Specific(26) l:42, vendor:Microsoft(311)
    t:MS MPPE Send Key(16) l:36, value:9EC9316DBFA701FF0499D36A1032678143...
  t:Vendor Specific(26) l:12, vendor:Microsoft(311)
    t:MS MPPE Encryption Policy(7) l:6, value:00000001
```

```
t:Vendor Specific(26) l:12, vendor:Microsoft(311)
t:MS MPPE Encryption Type(8) l:6, value:00000006
```

## Management Protocol Behavior

When you use local authentication, management protocols behave the same way that they do when you are not using RADIUS servers. When you are using RADIUS servers for authentication, management protocols behave as described in this section.

- **Telnet**—The “user” or admin accounts are authenticated locally, not via the RADIUS server. For all other accounts, the configured RADIUS servers are used for authentication. If authentication is successful, the user is granted privileges depending on the ACME\_USER\_CLASS VSA attribute.
- **FTP**—The “user” or admin accounts are authenticated locally, not via the RADIUS server. For all other accounts, the configured RADIUS servers are used for authentication.
- **SSH in pass-through mode**—When SSH is in pass through mode, the Oracle Communications Session Border Controller behave the same way that it does for Telnet.
- **SSH in non-pass-through mode**—When you create an SSH account on the Oracle Communications Session Border Controller, you are asked to supply a user name and password. Once local authentication succeeds, you are prompted for the ACLI user name and password. If your user ACLI name is user, then you are authenticated locally. Otherwise, you are authenticated using the RADIUS server. If RADIUS authentication is successful, the privileges you are granted depend on the ACME\_USER\_CLASS VSA attribute.
- **SFTP in pass-through mode**—If you do not configure an SSH account on the Oracle Communications Session Border Controller, the RADIUS server is contacted for authentication for any user that does not have the user name user. The Oracle Communications Session Border Controller uses local authentication if the user name is user.
- **SFTP in non-pass-through mode**—The “user” or admin accounts are authenticated locally, not via the RADIUS server. For all other accounts, the configured RADIUS servers are used for authentication.

## RADIUS Authentication Configuration

To enable RADIUS authentication and user access on your Oracle Communications Session Border Controller, you need to configure global parameters for the feature and then configure the RADIUS servers that you want to use.

### Global Authentication Settings

To configure the global authentication settings:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **security** and press Enter.

```
ACMEPACKET(configure)# security
```

3. Type **authentication** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(security)# authentication
ACMEPACKET(authentication)#
```

From here, you can view the entire menu for the authentication configuration by typing a **?**. You can set global parameters for authentication. You can also configure individual RADIUS servers; instructions for configuring RADIUS server appear in the next section.

4. **type**—Set the type of user authentication you want to use on this Oracle Communications Session Border Controller. The default value is **local**. The valid values are:
  - local | radius
5. **protocol**—If you are using RADIUS user authentication, set the protocol type to use with your RADIUS server(s). The default is **pap**. The valid values are:
  - pap | chap | mschap2

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6. **source-port**—Set the number of the port you want to use from message sent from the Oracle Communications Session Border Controller to the RADIUS server. The default value is 1812. The valid values are:
  - 1645 | 1812
7. **allow-local-authorization**—Set this parameter to **enabled** if you want the Oracle Communications Session Border Controller to authorize users to enter Superuser (administrative) mode locally even when your RADIUS server does not return the ACME\_USER\_CLASS VSA or the Cisco-AVPair VSA. The default for this parameter is **disabled**.

### RADIUS Server Settings

The parameters you set for individual RADIUS servers identify the RADIUS server, establish a password common to the Oracle Communications Session Border Controller and the server, and establish trying times.

Setting the class and the authentication methods for the RADIUS servers can determine how and when they are used in the authentication process.

To configure a RADIUS server to use for authentication:

1. Access the RADIUS server submenu from the main authentication configuration:

```
ACMEPACKET (authentication) # radius-servers
ACMEPACKET (radius-servers) #
```

2. **address**—Set the remote IP address for the RADIUS server. There is no default value, and you are required to configure this address.
3. **port**—Set the port at the remote IP address for the RADIUS server. The default port is set to **1812**. The valid values are:
  - 1645 | 1812
4. **state**—Set the state of the RADIUS server. Enable this parameter to use this RADIUS server to authenticate users. The default value is **enabled**. The valid values are:
  - enabled | disabled
5. **secret**—Set the password that the RADIUS server and the Oracle Communications Session Border Controller share. This password is transmitted between the two when the request for authentication is initiated; this ensures that the RADIUS server is communicating with the correct client.
6. **nas-id**—Set the NAS ID for the RADIUS server. There is no default for this parameter.
7. **retry-limit**—Set the number of times that you want the Oracle Communications Session Border Controller to retry for authentication information from this RADIUS server. The default value is **3**. The valid range is:
  - Minimum—1
  - Maximum—5

If the RADIUS server does not respond within this number of tries, the Oracle Communications Session Border Controller marks is as dead.
8. **retry-time**—Set the amount of time (in seconds) that you want the Oracle Communications Session Border Controller to wait before retrying for authentication from this RADIUS server. The default value is **5**. The valid range is:
  - Minimum—5
  - Maximum—10
9. **dead-time**—Set the amount of time in seconds before the Oracle Communications Session Border Controller retries a RADIUS server that it has designated as dead because that server did not respond within the maximum number of retries. The default is **10**. The valid range is:
  - Minimum—10
  - Maximum—10000

**10. maximum-sessions**—Set the maximum number of outstanding sessions for this RADIUS server. The default value is **255**. The valid range is:

- Minimum—1
- Maximum—255

**11. class**—Set the class of this RADIUS server as either primary or secondary. A connection to the primary server is tried before a connection to the secondary server is tried. The default value is **primary**. Valid values are:

- primary | secondary

The Oracle Communications Session Border Controller tries to initiate contact with primary RADIUS servers first, and then tries the secondary servers if it cannot reach any of the primary ones.

If you configure more than one RADIUS server as primary, the Oracle Communications Session Border Controller chooses the one with which it communicates using a round-robin strategy. The same strategy applies to the selection of secondary servers if there is more than one.

**12. authentication-methods**—Set the authentication method you want the Oracle Communications Session Border Controller to use with this RADIUS server. The default value is **pap**. Valid values are:

- all | pap | chap | mschapv2

This parameter has a specific relationship to the global protocol parameter for the authentication configuration, and you should exercise care when setting it. If the authentication method that you set for the RADIUS server does not match the global authentication protocol, then the RADIUS server is not used. The Oracle Communications Session Border Controller simply overlooks it and does not send authentication requests to it. You can enable use of the server by changing the global authentication protocol so that it matches.

**13.** Save your work and activate your configuration.

## TACACS+

TACACS+ (Terminal Access Controller Access Control System Plus) is a protocol originally developed by Cisco Systems, and made available to the user community by a draft RFC, *TACACS+ Protocol, Version 1.78* (draft-grant-tacacs-02.txt). TACACS+ provides AAA (Authentication, Authorization, and Accounting) services over a secure TCP connection using Port 49.

### TACACS+ Introduction

Like DIAMETER and RADIUS, TACACS+ uses a client/server model in which a Network Access Server (NAS) acts in the client role and a TACACS+ equipped device (a daemon in TACACS+ nomenclature) assumes the server role. For purposes of the current implementation, the Oracle Oracle Communications Session Border Controller functions as the TACACS+ client. Unlike RADIUS, which combines authentication and authorization, TACACS+ provides three distinct applications to provide finer grade access control.

Authentication is the process that confirms a user's purported identity. Authentication is most often based on a simple username/password association, but other, and more secure methods, are becoming more common. The following authentication methods are support by the current implementation: simple password, PAP (Protocol Authentication Protocol), and CHAP (Challenge Handshake Authentication Protocol).

Authorization is the process that confirms user privileges. TACACS+ can provide extremely precise control over access to system resources. In the current implementation, TACACS+ controls access to system administrative functions.

TACACS+ provides secure communication between the client and daemon by encrypting all packets. Encryption is based on a shared-secret, a string value known only to the client and daemon. Packets are encrypted in their entirety, save for a common TACACS+ header.

The cleartext header contains, among other fields, a version number, a sequence number, and a session ID. Using a methodology described in Section 5 of the TACACS+ draft RFC, the sender encrypts outbound cleartext messages by repetitively running the MD5 hash algorithm over the concatenation of the session ID, shared-secret, version number,

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and sequence number values, eventually deriving a virtual one-time-pad of the same length as the message body. The sender encrypts the cleartext message with an XOR (Exclusive OR) operation, using the cleartext message and virtual one-time-pad as inputs.

The message recipient, who possesses the shared-secret, can readily obtain the version number, sequence number, session ID, and message length from the cleartext header. Consequently, the recipient employs the same methodology to derive a virtual one-time-pad identical to that derived by the sender. The recipient decrypts the encrypted message with an XOR operation, using the encrypted message and virtual one-time-pad as inputs.

Details on TACACS+ are available in the ACLI Configuration Guide.

The TACACS+ implementation is based upon the following internet draft.

draft-grant-tacacs-02.txt, *The TACACS+ Protocol Version 1.78*

Other relevant documents include

RFC 1321, *The MD-5 Message Digest Algorithm*

RFC 1334, *PPP Authentication Protocols* .

RFC 1994, *PPP Challenge Handshake Authentication Protocol (CHAP)*

## TACACS+ Authentication

The Oracle Communications Session Border Controller uses TACACS+ authentication services solely for the authentication of user accounts. Administrative users must be authenticated locally by the Oracle Communications Session Border Controller.

The current TACACS+ implementation supports three types of user authentication: simple password (referred to as `ascii` by TACACS+), PAP, and CHAP.

### ascii Login

`ascii` login is analogous to logging into a standard PC. The initiating peer is prompted for a username, and, after responding, is then prompted for a password.

### PAP Login

PAP is defined in RFC 1334, *PPP Authentication Protocols*. This protocol offers minimal security in that passwords are transmitted as unprotected cleartext. PAP login differs from `ascii` login in that the username and password are transmitted to the authenticating peer in a single authentication packet, as opposed to the two-step prompting process used in `ascii` login.

### CHAP Login

CHAP is defined in RFC 1994, *PPP Challenge Handshake Authentication Protocol*. CHAP is a more secure than PAP in that it is based on a shared-secret (known only to the communicating peers), and therefore avoids the transmission of cleartext authentication credentials. CHAP operations can be summarized as follows.

After a login attempt, the initiator is tested by the authenticator who responds with a packet containing a challenge value — an octet stream with a recommended length of 16 octets or more. Receiving the challenge, the initiator concatenates an 8-bit identifier (carried within the challenge packet header), the shared-secret, and the challenge value, and uses the shared-secret to compute an MD-5 hash over the concatenated string. The initiator returns the hash value to the authenticator, who performs the same hash calculation, and compares results. If the hash values match, authentication succeeds; if hash values differ, authentication fails.

### Authentication Message Exchange

All TACACS+ authentication packets consist of a common header and a message body. Authentication packets are of three types: START, CONTINUE, and REPLY.

START and CONTINUE packets are always sent by the Oracle Communications Session Border Controller, the TACACS+ client. START packets initiate an authentication session, while CONTINUE packets provide authentication data requested by the TACACS+ daemon. In response to every client-originated START or

CONTINUE, the daemon must respond with a REPLY packet. The REPLY packet contains either a decision (pass or fail), which terminates the authentication session, or a request for additional information needed by the authenticator.

### TACACS+ Header

The TACACS+ header format is as follows.

maj	min	type	seq_no	flags
ver	ver			
session_id				
length				

maj ver

This 4-bit field identifies the TACACS+ major protocol version, and must contain a value of 0xC .

min ver

This 4-bit field identifies the TACACS+ minor protocol version, and must contain either a value of 0x0 (identifying TACACS+ minor version 0) or a value of 0x1 . (identifying TACACS+ minor version 1). Minor versions 0 and 1 differ only in the processing of PAP and CHAP logins.

type

This 8-bit field identifies the TACACS+ AAA service as follows:

0x1 — TACACS+ Authentication

0x2 — TACACS+ Authorization

0x3 — TACACS+ Accounting

sequence-no

This 8-bit field contains the packet sequence for the current session.

The first packet of a TACACS+ session must contain the value 1; each following packet increments the sequence count by 1. As TACACS+ sessions are always initiated by the client, all client-originated packets carry an odd sequence number, and all daemon-originated packets carry an even sequence number. TACACS+ protocol strictures do not allow the sequence\_no field to wrap. If the sequence count reaches 255, the session must be stopped and restarted with a new sequence number of 1.

flags

This 8-bit field contains flags as described in Section 3 of the draft RFC; flags are not under user control.

session\_id

This 32-bit field contains a random number that identifies the current TACACS+ session — it is used by clients and daemons to correlate TACACS+ requests and responses.

length

This 32-bit field contains the total length of the TACACS+ message, excluding the 12-octet header — in other words, the length of the message body.

### Authentication START Packet

The Oracle Communications Session Border Controller, acting as a TACACS+ client, sends an authentication START packet to the TACACS+ daemon to initiate an authentication session. The daemon must respond with a REPLY packet.

The authentication START packet format is as follows.

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Common Header			
type contains 0x1			
action	priv_lvl	authen_ type	service
user_len	port_len	rem_addr_ _len	data_len
user ...			
port ...			
rem-addr ...			
data ...			

### action

This 8-bit field contains an enumerated value that identifies the requested authentication action. For the current TACACS+ implementation, this field always contains a value of 0x01 , indicating user login authentication.

### priv\_lvl

This 8-bit field contains an enumerated value that identifies the privilege level requested by an authenticating user. For the current TACACS+ authentication implementation, this field always contains a value of 0x01 , indicating the user level.

### authen-type

This 8-bit field contains an enumerated value that identifies the authentication methodology. Supported values are as follows:

0x01 ASCII — simple login, Oracle Communications Session Border Controller prompts for username and password

0x02 PAP — as specified in RFC 1334

0x03 CHAP — as specified in RFC 1994

### service

This 8-bit field contains an enumerated value that identifies the service requesting the authentication. For the current TACACS+ implementation, this field always contains a value of 0x01 , indicating user login authentication.

### user\_len

This 8-bit field contains the length of the user field in octets.

### port\_len

This 8-bit field contains the length of the port field in octets. As the port field is not used in the current TACACS+ authentication implementation, the port\_len field always contains a value of 0 as specified in Section 4 of the TACACS+ draft RFC.

### rem\_addr\_len

This 8-bit field contains the length of the rem\_addr field in octets. As the rem\_addr field is not used in the current TACACS+ authentication implementation, the rem\_addr\_len field always contains a value of 0 as specified in Section 4 of the TACACS+ draft RFC.

### data\_len

This 8-bit field contains the length of the data field in octets.

### user



This variable length field contains the login name of the user to be authenticated.

port

This variable length field contains the name of the Oracle Communications Session Border Controller port on which authentication is taking place. Following Cisco Systems convention, this field contains the string tty10 .

rem\_addr

This variable length field contains the location of the user to be authenticated. This field contains the localhost address.

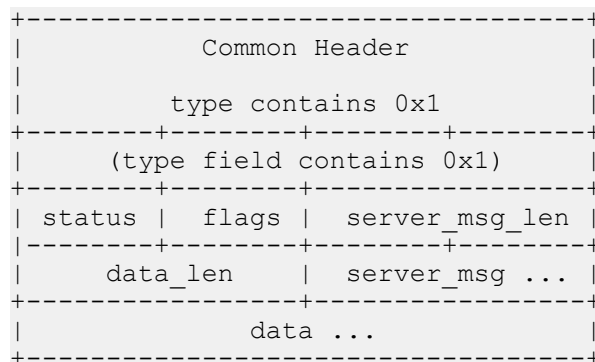
data

This optional variable length field contains miscellaneous data.

## Authentication REPLY Packet

The TACACS+ daemon sends an authentication REPLY packet to the Oracle Communications Session Border Controller in response to a authentication START or authentication CONTINUE packet. Depending on the contents of the status field, the authentication REPLY packet either ends the authentication transaction, or continues the transaction by requesting addition information needed by the authenticator.

The authentication REPLY packet format is as follows.



status

This 16-bit field contains an enumerated value that specifies the current state of the authentication process. Supported values are as follows:

0x01 PASS — the user is authenticated, thus ending the session

0x02 FAIL — the user is rejected, thus ending the session

0x04 GETUSER — daemon request for the user name

0x05 GETPASS — daemon request for the user password

0x06 RESTART — restarts the transaction, possibly because the sequence number has wrapped, or possibly because the requested authentication type is not supported by the daemon

0x07 ERROR — reports an unrecoverable error

flags

This 8-bit field contains various flags that are not under user control.

server\_msg\_len

This 16-bit field contains the length of the server\_msg field in octets. As the server\_msg field is not used in REPLY packets sent by the current TACACS+ authentication implementation, the server\_msg\_len field always contains a value of 0 as specified in Section 4 of the TACACS+ draft RFC.

data\_len

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This 16-bit field contains the length of the data field in octets. As the data field is not used in REPLY packets sent by the current TACACS+ authentication implementation, the data\_len field always contains a value of 0 as specified in Section 4 of the TACACS+ draft RFC.

server\_msg

This optional variable length field contains a server message intended for display to the user. The current TACACS+ authentication implementation does not use this field.

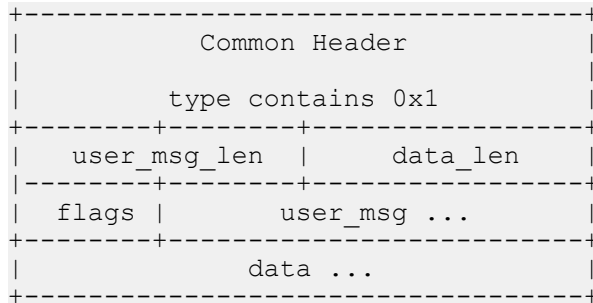
data

This optional variable length field contains data pertinent to the authentication process. The current TACACS+ authentication implementation does not use this field.

### Authentication CONTINUE Packet

The Oracle Communications Session Border Controller, acting as a TACACS+ client, sends an authentication CONTINUE packet to the TACACS+ daemon in response to a REPLY message which requested additional data required by the authenticator.

The authentication CONTINUE packet format is as follows.



user\_msg\_len

This 16-bit field contains the length of the user\_msg field in octets.

data\_len

This 16-bit field contains the length of the data field in octets. As the data field is not used in the current TACACS+ authentication implementation, the data field always contains a value of 0 as specified in Section 4 of the TACACS+ draft RFC.

flags

This 8-bit field contains various flags that are not under user control.

user\_msg

This variable length field contains a string that responds to an information request contained in a REPLY message.

data

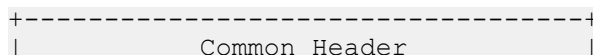
This optional variable length field contains miscellaneous data, often in response to a daemon request. The current TACACS+ authentication implementation does not use the data field in Authentication CONTINUE packets.

### Authentication Scenarios

Each of the supported user authentication scenarios is described in terms of packet flow in the following sections.

#### ASCII Authentication

The Oracle Communications Session Border Controller initiates the authentication with an authentication START packet.



minor_version contains 0x0			
type contains 0x1			
action	priv_lvl	authen_	service
0x01	0x01	type	0x01
0x01	0x01	0x01	0x01
user_len	port_len	rem_addr_	data_len
0	N	len	0
0	N	N	0
port			
tty10			
rem_addr			
localhost address			

- The action field specifies the requested authentication action — 0x01 for TAC\_PLUSAUTHEN\_LOGIN (authentication of a user login).
- The priv\_lvl field specifies the privilege level requested by the user — 0x01 for TAC\_PLUS\_PRIV\_LVL\_USER.
- The authen\_type field specifies the authentication methodology — 0x01 for TAC\_PLUS\_AUTHEN\_TYPE\_ASCII (simple login).
- The service field specifies the requesting service — 0x01 for TAC\_PLUS\_AUTHEN\_SVC\_LOGIN (login service).
- The user\_len and data\_len fields contain a value of 0, as required by the TACACS+ protocol.
- The port\_len and rem\_addr\_len fields contain the length, in octets, of the port and rem\_addr fields.
- The port field contains the name of the Oracle Communications Session Border Controller port on which authentication is taking place. Following Cisco Systems convention, this field contains the string tty10.
- The rem\_addr field specifies the location of the user to be authenticated. This field contains the localhost address.

The TACACS+ daemon returns an authentication REPLY requesting the username.

Common Header		
minor_version contains 0x0		
type contains 0x1		
status	flags	server_msg_len
0x04		0
data_len		
0		

- The status field specifies a daemon request — 0x04 for TAC\_PLUS\_AUTH\_STATUS\_GETUSER (get username).
- The server\_msg\_len data\_len fields both contain a value of 0, as required by the TACACS+ protocol.

The Oracle Communications Session Border Controller responds with an authentication CONTINUE packet.

Common Header	
minor_version contains 0x0	
type contains 0x1	
user_msg_len	data_len
	0
flags	user_msg ...

- The user\_msg\_len field contains the length, in octets, of the user\_msg field.

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- The data\_len field contains a value of 0 , as required by the TACACS+ protocol.
- The user\_msg field contains the username to be authenticated.

The TCACS+ daemon returns a second authentication REPLY requesting the user password.

```
+-----+
|          Common Header          |
|  minor_version contains 0x0     |
|          type contains 0x1     |
+-----+-----+
| status | flags | server_msg_len |
|  0x05  |     |           0    |
+-----+-----+
| data_len |
|         0 |
+-----+
```

- The status field specifies a daemon request — 0x05 for TAC\_PLUS\_AUTH\_STATUS\_GETPASS (get user password).
- The server\_msg\_len and data\_len fields both contain a value of 0 , as required by the TACACS+ protocol.

The Oracle Communications Session Border Controller responds with a second authentication CONTINUE packet.

```
+-----+-----+
|          Common Header          |
|  minor_version contains 0x0     |
|          type contains 0x1     |
+-----+-----+
| user_msg_len | data_len |
|             |         0 |
+-----+-----+
| flags | user_msg ... |
+-----+
```

- The user\_msg\_len field contains the length, in octets, of the user\_msg field.
- The data\_len field contains a value of 0 , as required by the TACACS+ protocol.
- The user\_msg field contains the user password to be authenticated.
- Other, optional fields are not used.

The TCACS+ daemon returns a third authentication REPLY reporting the authentication result, and terminating the authentication session.

```
+-----+-----+
|          Common Header          |
|  minor_version contains 0x0     |
|          type contains 0x1     |
+-----+-----+
| status | flags | server_msg_len |
|  0x01  |     |           0    |
+-----+-----+
| data_len |
|         0 |
+-----+
```

- The status field specifies the authentication result — 0x01 for TAC\_PLUS\_AUTH\_STATUS\_PASS (authorization succeeds), or 0x02 for TAC\_PLUS\_AUTH\_STATUS\_FAIL (authorization fails).
- The server\_msg\_len , and data\_len fields both contain a value of 0 , as required by the TACACS+ protocol.

### PAP Authentication

The Oracle Communications Session Border Controller initiates the authentication with an authentication START packet.

```
+-----+
|          Common Header          |
+-----+
```

minor_version contains 0x1			
type contains 0x1			
action	priv_lvl	authen_	service
0x01	0x01	type	0x01
0x02			
user_len	port_len	rem_addr_	data_len
N	N	len	N
user			
port			
tty10			
rem_addr			
localhost address			
data ...			

- The action field specifies the requested authentication action — 0x01 for TAC\_PLUSAUTHEN\_LOGIN (authentication of a user login).
- The priv\_lvl field specifies the privilege level requested by the user — 0x01 for TAC\_PLUS\_PRIV\_LVL\_USER.
- The authen\_type field specifies the authentication methodology — 0x02 for TAC\_PLUS\_AUTHEN\_TYPE\_PAP (PAP login).
- The service field specifies the requesting service — 0x01 for TAC\_PLUS\_AUTHEN\_SVC\_LOGIN (login service).
- The user\_len field contains the length, in octets, of the user field.
- The port\_len field contains the length, in octets, of the port field.
- The rem\_addr\_len field contains the length, in octets, of the rem\_addr field.
- The data\_len field contains the length, in octets, of the data field.
- The user field contains the username to be authenticated.
- The port field contains the name of the Oracle Communications Session Border Controller port on which authentication is taking place. Following Cisco Systems convention, this field contains the string tty10 .
- The rem\_addr field specifies the location of the user to be authenticated. This field contains the localhost address.
- The data field contains the password to be authenticated.

The TCACS+ daemon returns an authentication REPLY reporting the authentication result.

Common Header		
minor_version contains 0x1		
type contains 0x1		
status	flags	server_msg_len
0x01		0
data_len		
0		

- The status field specifies the authentication result — 0x01 for TAC\_PLUS\_AUTH\_STATUS\_PASS (authorization succeeds), or 0x02 for TAC\_PLUS\_AUTH\_STATUS\_FAIL (authorization fails).
- The server\_msg\_len and data\_len fields both contain a value of 0 , as required by the TACACS+ protocol.
- Other, optional fields are not used.

**CHAP Authentication**

The Oracle Communications Session Border Controller initiates the authentication with an authentication START packet.

```

+-----+
|          Common Header          |
|  minor_version contains 0x1    |
|          type contains 0x1     |
+-----+-----+-----+-----+
| action | priv_lvl | authen_ | service |
|  0x01  |   0x01  |   type  |         |
|-----+-----+-----+-----+
| user_len | port_len | rem_addr | data_len |
|         |         |   _len  |         |
|         |         |         |         |
|         |         |         |         |
+-----+-----+-----+-----+
|          user                   |
+-----+-----+-----+-----+
|          port                   |
|          tty10                  |
+-----+-----+-----+-----+
|          rem_addr               |
|          localhost address      |
+-----+-----+-----+-----+
|          data ...               |
+-----+-----+-----+-----+

```

- The action field specifies the requested authentication action — 0x01 for TAC\_PLUSAUTHEN\_LOGIN (authentication of a user login).
- The priv\_lvl field specifies the privilege level requested by the user — 0x01 for TAC\_PLUS\_PRIV\_LVL\_USER.
- The authen\_type field specifies the authentication methodology — 0x03 for TAC\_PLUS\_AUTHEN\_TYPE\_CHAP (CHAP login).
- The service field specifies the requesting service — 0x01 for TAC\_PLUS\_AUTHEN\_SVC\_LOGIN (login service).
- The user\_len field contains the length, in octets, of the user field.
- The port\_len field contains the length, in octets, of the port field.
- The rem\_addr\_len field contains the length, in octets, of the rem\_addr field.
- The data\_len field contains the length, in octets, of the date field.
- The user field contains the username to be authenticated.
- The port field contains the name of the Oracle Communications Session Border Controller port on which authentication is taking place. Following Cisco Systems convention, this field contains the string tty10 .
- The rem\_addr field specifies the location of the user to be authenticated. This field contains the localhost address.
- The data field contains the password to be authenticated.

The TCACS+ daemon returns an authentication REPLY reporting the authentication result.

```

+-----+-----+-----+-----+
|          Common Header          |
|  minor_version contains 0x1    |
|          type contains 0x1     |
+-----+-----+-----+-----+
| status | flags | server_msg_len |
|  0x01  |      |              0 |
+-----+-----+-----+-----+
| data_len |
|         |
|         |
+-----+-----+-----+-----+

```

- The status field specifies the authentication result — 0x01 for TAC\_PLUS\_AUTH\_STATUS\_PASS (authorization succeeds), or 0x02 for TAC\_PLUS\_AUTH\_STATUS\_FAIL (authorization fails).

- The `server_msg_len` and `data_len` fields both contain a value of 0, as required by the TACACS+ protocol.
- Other, optional fields are not used.

## TACACS+ Authorization

The Oracle Communications Session Border Controller uses TACACS+ services to provide administrative authorization. With TACACS+ authorization enabled, each individual CLI command issued by an admin user is authorized by the TACACS+ authorization service. The Oracle Communications Session Border Controller replicates each CLI command in its entirety, sends the command string to the authorization service, and suspends command execution until it receives an authorization response. If TACACS+ grants authorization, the pending command is executed; if authorization is not granted, the Oracle Communications Session Border Controller does not execute the CLI command, and displays an appropriate error message.

The daemon's authorization decisions are based on a database lookup. Data base records use regular expressions to associate specific command string with specific users. The construction of such records is beyond the scope of this document.

### Authorization Message Exchange

All TACACS+ authorization packets consist of a common header and a message body. Authorization packets are of two types: REQUEST and RESPONSE.

The REQUEST packet, which initiates an authorization session, is always sent by the Oracle Communications Session Border Controller. Upon receipt of every REQUEST, the daemon must answer with a RESPONSE packet. In the current TACACS+ implementation, the RESPONSE packet must contain an authorization decision (pass or fail). The exchange of a single REQUEST and the corresponding RESPONSE completes the authorization session.

### Authorization REQUEST Packet

The Oracle Communications Session Border Controller, acting as a TACACS+ client, sends an authorization REQUEST packet to the TACACS+ daemon to initiate an authorization session.

The authorization REQUEST packet format is as follows.

```

+-----+
|           Common Header           |
+-----+
|           type contains 0x2       |
+-----+
|authen_ |priv_lvl|authen_ |authen- |
|method  |        |type      |service |
+-----+
|user_len|port_len|rem_addr|arg_cnt |
|        |        |_len    |        |
+-----+
|arg1_len|arg2_len| ...   |argN_len|
|        |        |      |      |
+-----+
|           user ...               |
+-----+
|           port ...               |
+-----+
|           rem-addr ...           |
+-----+
|           arg1 ...               |
+-----+
|           arg2 ...               |
+-----+
|           argN ...               |
+-----+

```

`authen_method`

## Getting Started

---

This 8-bit field contains an enumerated value that identifies the method used to authenticate the authorization subject — that is, an admin user. Because the admin user was authenticated locally by the Oracle Communications Session Border Controller, this field always contains a value of 0x05 , indicating authentication by the requesting client.

priv\_lvl

This 8-bit field contains an enumerated value that identifies the privilege level associated with the authorization subject. For the current TACACS+ authorization implementation, this field always contains a value of 0x00 .

authen-type

This 8-bit field contains an enumerated value that identifies the methodology. used to authenticate the authorization subject. Because the admin user was authenticated with a simple username/password exchange, this field always contains a value of 0x01 , indicating ascii login.

authen\_service

This 8-bit field contains an enumerated value that identifies the service that requested authentication. Because an admin user is authenticated with a simple username/password exchange, this field always contains a value of 0x01 , the login service.

user\_len

This 8-bit field contains an integer that specifies the length, in octets, of the user field.

port\_len

This 8-bit field contains an integer that specifies the length, in octets, of the port field.

rem\_addr\_len

This 8-bit field contains an integer that specifies the length, in octets, of the rem\_addr field.

arg\_cnt

This 8-bit field contains an integer that specifies the number or arguments contained with the REQUEST. Given the design of the current TACACS+ implementation, this field always contains a value of 0x02 .

arg1\_len

This 8-bit field contains an integer that specifies the length, in octets, of the first argument.

Subsequent fields contain the length of each sequential argument.

user

This variable length field contains the login name of the user to be authorized.

port

This variable length field contains the name of the Oracle Communications Session Border Controller port on which authorization is taking place. Following Cisco Systems convention, this field contains the string tty10 .

rem\_addr

This variable length contains the location of the user to be authorized. This field contains the localhost address.

arg...

This variable length field contains a TACACS+ attribute value pair (AVP); each arg field holds a single AVP.

A TACACS+ AVP is an ASCII string with a maximum length of 255 octets. The string consists of the attribute name and its assigned value separated by either an equal sign (=) or by an asterisk (\*). The equal sign (=) identifies a mandatory argument, one that must be understood and processed by the TACACS+ daemon; the asterisk (\*) identifies an optional argument that may be disregarded by either the client or daemon.

Administrative authorization requires the use of only two TACACS+ AVPs: service and cmd .

The service AVP identifies the function to be authorized. In the case of the current implementation, the attribute value is always shell . Consequently the attribute takes the follow format:



```
service=shell
```

The cmd AVP identifies the specific ACLI command to be authorized. The command is passed in its entirety, from the administrative configuration root, **configure terminal**, through the final command argument. For example,

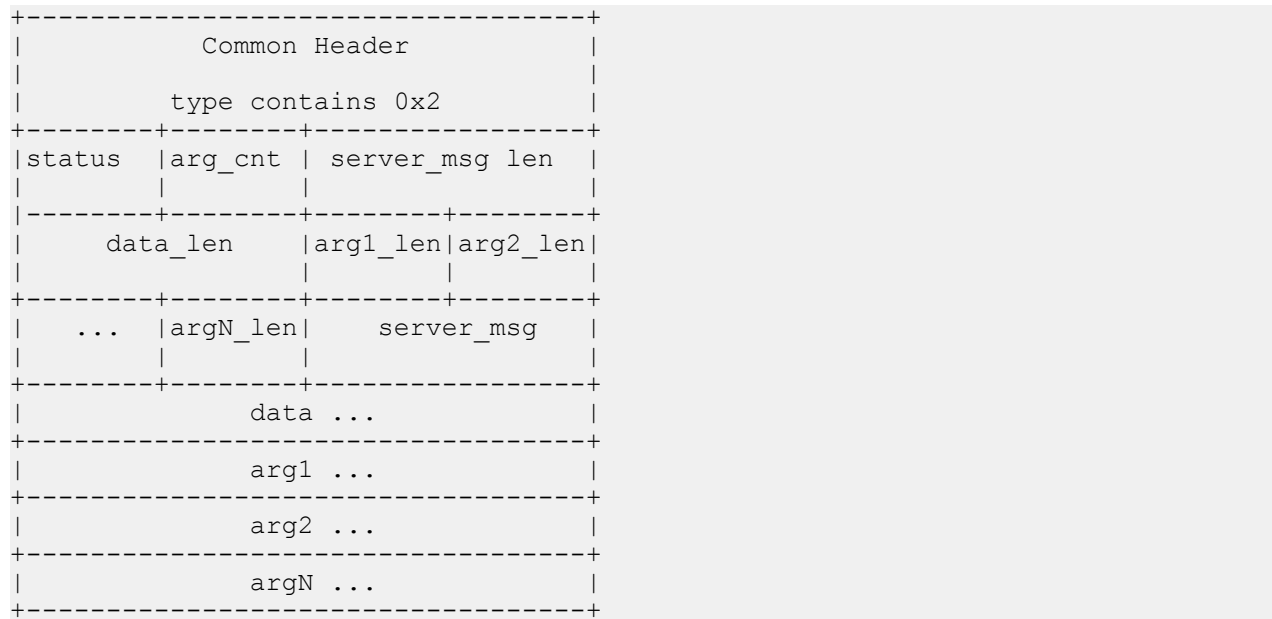
```
cmd=configure terminal security authentication type tacacsplus
```

Note the equal sign (=) used in the attribute examples, indicating that both are mandatory arguments.

### Authorization RESPONSE Packet

The TACACS+ daemon sends an authorization RESPONSE packet to the Oracle Communications Session Border Controller to report authorization results.

The authorization RESPONSE packet format is as follows.



status

This 8-bit field contains an enumerated value that specifies the results of the authorization process. Supported values are 0x01 (Pass), 0x10 (Fail), and 0x11 (Error). Fail indicates that the authorization service rejected the proposed operation, while Error indicates the authorization service failed.

If authorization succeeds (status=0x01), the ACLI command is executed; if authorization fails, for whatever the reason (status=0x10 or 0x11), the ACLI command is not executed, and an appropriate error message is generated.

arg\_cnt

This 8-bit field contains an integer that specifies the number of arguments contained with the RESPONSE. Given the design of the current TACACS+ implementation, this field always contains a value of 0x02.

server\_msg\_len

This 16-bit field contains an integer that specifies the length, in octets, of the server\_msg field.

data\_len

This 16-bit field contains an integer that specifies the length, in octets, of the data field.

arg1\_len

This 8-bit field contains an integer that specifies the length, in octets, of the first argument.

Subsequent fields contain the length of each sequential argument.

server-msg

## Getting Started

This optional variable length field contains a string that can be presented to the user.

data

This optional variable length field contains a string that can be presented to an administrative display, console, or log.  
arg...

This optional variable length field contains a TACACS+ attribute value pair (AVP); each arg field holds a single AVP.

No arguments are generated in RESPONSE packets within the current TACACS+ implementation.

### Authorization Scenarios

Successful and failed administrative authorization is described in terms of packet flow in the following sections.

### Authorization Pass

The Oracle Communications Session Border Controller initiates the authorization with an authorization REQUEST packet.

```
+-----+
|           Common Header           |
|           type contains 0x2       |
+-----+
|authen_ |priv_lvl|authen_ |authen_ |
|method  |       |type    |service |
| 0x05   | 0x00  | 0x01   | 0x01   |
+-----+
|user_len|port_len|rem_addr|arg_cnt |
|   N    |   N    |  _len  |   2    |
+-----+
|arg1_len|arg2_len|   user ... |
|   N    |   N    |   login name |
+-----+
|           port                    |
|           tty10                   |
+-----+
|           rem_addr                 |
|           localhost address       |
+-----+
|           arg1                     |
|           AVP                     |
|           service=shell           |
+-----+
|           arg2                     |
|           AVP                     |
|           cmd=configure terminal security |
+-----+
```

- The `authen_method` field specifies the method used to authenticate the subject — 0x05 for TAC\_PLUS\_AUTHEN\_METHOD\_LOCAL (authentication by the client).
- The `priv_lvl` field specifies the privilege level requested by the user — 0x00 for TAC\_PLUS\_PRIV\_LVL\_MIN.
- The `authen_type` field specifies the authentication methodology — 0x01 for TAC\_PLUS\_AUTHEN\_TYPE\_ASCII (simple login).
- The `authen_service` field specifies the requesting service — 0x01 for TAC\_PLUS\_AUTHEN\_SVC\_LOGIN (login service).
- The `user_len` field contains the length, in octets, of the user field.
- The `port_len` field contains the length, in octets, of the port field.
- The `rem_addr_len` field contains the length, in octets, of the `rem_addr` field.
- The `arg_cnt` field contains the number of arguments in the message body.

- The arg1\_len field contains the length, in octets, of the service AVP.
- The arg2\_len field contains the length, in octets, of the service AVP.
- The user field contains the login name of an admin user.
- The port field contains the name of the Oracle Communications Session Border Controller port on which authentication is taking place. Following Cisco Systems convention, this field contains the string tty10 .
- The rem\_addr field specifies the location of the user to be authenticated. This field contains the localhost address.
- The arg1 field contains the mandatory service AVP.
- The arg2 field contains the mandatory cmd AVP.

The TCACS+ daemon returns a authorization RESPONSE reporting the status, and terminating the authorization session.

```

+-----+
|           Common Header           |
|           type contains 0x2       |
+-----+-----+-----+
| status | arg_cnt | server_msg_len |
| 0x01  | 0      | 0              |
+-----+-----+-----+
|           data_len               |
|           0                      |
+-----+

```

- The status field specifies the authorization status — 0x01 for TAC\_PLUS\_AUTHOR\_STATUS\_PASS\_ADD (authorization approved).
- The arg\_cnt field contains a value of 0 — the authorization RESPONSE returns no arguments.
- The server\_msg\_len and data\_len fields both contain a value of 0, as required by the TACACS+ protocol.

### Authorization Fail

The Oracle Communications Session Border Controller initiates the authorization with an authorization REQUEST packet.

```

+-----+
|           Common Header           |
|           type contains 0x2       |
+-----+-----+-----+-----+
| authen_ | priv_lvl | authen_ | authen_ |
| method  |          | type    | service |
| 0x05    | 0x00    | 0x01    | 0x01    |
+-----+-----+-----+-----+
| user_len | port_len | rem_addr | arg_cnt |
|          |          | _len    |         |
| N        | N        | N       | 2       |
+-----+-----+-----+-----+
| arg1_len | arg2_len |          | user ... |
|          |          |          | login name |
+-----+-----+-----+-----+
|           port                   |
|           tty10                  |
+-----+-----+-----+-----+
|           rem_addr               |
|           localhost address      |
+-----+-----+-----+-----+
|           arg1                   |
|           AVP                    |
|           service=shell          |
+-----+-----+-----+-----+
|           arg2                   |
+-----+

```

## Getting Started

```
|          AVP          |
| cmd=configure terminal security |
+-----+
```

- The `authen_method` field specifies the method used to authenticate the administrative subject — 0x05 for `TAC_PLUS_AUTHEN_METHOD_LOCAL` (authentication by the client).
- The `priv_lvl` field specifies the privilege level requested by the user — 0x00 for `TAC_PLUS_PRIV_LVL_MIN`.
- The `authen_type` field specifies the authentication methodology — 0x01 for `TAC_PLUS_AUTHEN_TYPE_ASCII` (simple login).
- The `authen_service` field specifies the requesting service — 0x01 for `TAC_PLUS_AUTHEN_SVC_LOGIN` (login service).
- The `user_len` field contains the length, in octets, of the user field.
- The `port_len` field contains the length, in octets, of the port field.
- The `rem_addr_len` field contains the length, in octets, of the rem-addr field.
- The `arg_cnt` field contains the number of arguments in the message body.
- The `arg1_len` field contains the length, in octets, of the service AVP.
- The `arg2_len` field contains the length, in octets, of the service AVP.
- The user field contains the login name of an admin user.
- The port field contains the name of the Oracle Communications Session Border Controller port on which authentication is taking place. Following Cisco Systems convention, this field contains the string `tty10`.
- The `rem_addr` field specifies the location of the user to be authenticated. This field contains the localhost address.
- The `arg1` field contains the mandatory service AVP.
- The `arg2` field contains the mandatory cmd AVP.

The TCACS+ daemon returns an authorization RESPONSE reporting the status, and terminating the authorization session.

```
+-----+
|          Common Header          |
|          type contains 0x2      |
+-----+-----+-----+-----+
| status | arg_cnt | server_msg_len |
| 0x10  | 0     | 0             |
+-----+-----+-----+-----+
|          data_len              |
| 0                                         |
+-----+
```

- The `status` field specifies the authorization status — 0x10 for `TAC_PLUS_AUTHOR_STATUS_FAIL` (authorization rejected).
- The `arg_cnt` field contains a value of 0 — the authorization RESPONSE returns no arguments.
- The `server_msg_len` and `data_len` fields both contain a value of 0, as required by the TACACS+ protocol.

## TACACS+ Accounting

The Oracle Communications Session Border Controller uses TACACS+ accounting to log administrative actions. With accounting enabled, each individual CLI command executed by an admin user is logged by the accounting service.

### Accounting Message Exchange

All TACACS+ accounting packets consist of a common header and a message body. Accounting packets are of two types: REQUEST and REPLY.

The REQUEST packet has three variant forms. The START variant initiates an accounting session; the STOP variant terminates an accounting session; the WATCHDOG variant updates the current accounting session. REQUEST packets are always sent by the Oracle Communications Session Border Controller. Upon receipt of every REQUEST, the daemon must answer with a REPLY packet.

A TACACS+ accounting session proceeds as follows.

1. Immediately following successful authorization of an admin user, the Oracle Communications Session Border Controller sends an accounting REQUEST START packet.
2. The daemon responds with an accounting REPLY packet, indicating that accounting has started.
3. For each ACLI command executed by an admin user, the Oracle Communications Session Border Controller sends an accounting REQUEST WATCHDOG packet requesting accounting of the ACLI command. As the Oracle Communications Session Border Controller sends the WATCHDOG only after an admin user's access to the ACLI command is authorized, the accounting function records only those commands executed by the user, not those commands for which authorization was not granted.
4. The daemon responds with an accounting REPLY packet, indicating that the ACLI operation has been recorded by the accounting function.
5. Steps 3 and 4 are repeated for each authorized ACLI operation.
6. Immediately following logout (or timeout) of an admin user, the Oracle Communications Session Border Controller sends an accounting REQUEST STOP packet.
7. The daemon responds with an accounting REPLY packet, indicating that accounting has stopped.

### Accounting REQUEST Packet

The Oracle Communications Session Border Controller, acting as a TACACS+ client, sends an accounting REQUEST START variant to the TACACS+ daemon following the successful authorization of an admin user. It sends an accounting REQUEST WATCHDOG variant to the daemon following the authorization of an admin user's access to an ACLI command. It sends an accounting REQUEST STOP variant to the daemon at the conclusion of the ACLI session.

The accounting REQUEST packet format is as follows.

```

+-----+
|           Common Header           |
|           type contains 0x3       |
+-----+-----+-----+-----+
| flags | authen_ | priv_lvl | authen- |
|       | method  |         | type   |
+-----+-----+-----+-----+
| authen_ | user_len | port_len | rem_addr |
| service |         |         | _len    |
+-----+-----+-----+-----+
| arg_cnt | arg1_len | arg2_len | argN_len |
|         |         |         |         |
+-----+-----+-----+-----+
| argN_len |         | user ... |         |
+-----+-----+-----+-----+
|         |         | port ... |         |
+-----+-----+-----+-----+
|         |         | rem-addr ... |         |
+-----+-----+-----+-----+
|         |         | arg1 ... |         |
+-----+-----+-----+-----+
|         |         | arg2 ... |         |
+-----+-----+-----+-----+
|         |         | argN ... |         |
+-----+-----+-----+-----+

```

flags

This 8-bit field contains an enumerated value that identifies the accounting REQUEST variant.

0x2 — START

0x4 — STOP

## Getting Started

---

0x8 — WATCHDOG

authen\_method

This 8-bit field contains an enumerated value that identifies the method used to authenticate the accounting subject — that is, an admin user. Because an admin user is authenticated locally by the Oracle Communications Session Border Controller, this field always contains a value of 0x05 , indicating authentication by the requesting client.

priv\_lvl

This 8-bit field contains an enumerated value that identifies the privilege level associated with the accounting subject. For the current TACACS+ accounting implementation, this field always contains a value of 0x00 .

authen-type

This 8-bit field contains an enumerated value that identifies the methodology. used to authenticate the accounting subject. Because an admin user is authenticated with a simple username/password exchange, this field always contains a value of 0x01 , indicating ascii login.

authen\_service

This 8-bit field contains an enumerated value that identifies the service that requested authentication. Because an admin user is authenticated with a simple username/password exchange, this field always contains a value of 0x01 , the login service.

user\_len

This 8-bit field contains an integer that specifies the length, in octets, of the user field.

port\_len

This 8-bit field contains an integer that specifies the length, in octets, of the port field.

rem\_addr\_len

This 8-bit field contains an integer that specifies the length, in octets, of the rem\_addr field.

arg\_cnt

This 8-bit field contains an integer that specifies the number or arguments contained with the accounting REQUEST.

arg1\_len

This 8-bit field contains an integer that specifies the length, in octets, of the first argument.

Subsequent fields contain the length of each sequential argument.

user

This variable length field contains the login name of the accounting subject.

port

This variable length field contains the name of the Oracle Communications Session Border Controller port on accounting is taking place. Following Cisco System convention, this field always contains the string tty10 .

rem\_addr

This variable length contains the location of the authorization subject. This field always contains the localhost address.

arg...

This variable length field contains a TACACS+ attribute value pair (AVP); each arg field holds a single AVP.

A TACACS+ AVP is an ASCII string with a maximum length of 255 octets. The string consists of the attribute name and its assigned value separated by either an equal sign (=) or by an asterisk (\*). The equal sign (=) identifies a mandatory argument, one that must be understood and processed by the TACACS+ daemon; the asterisk (\*) identifies an optional argument that may be disregarded by either the client or daemon.

Administrative accounting requires the use of five TACACS+ AVPs: service, task-id, start\_time, and stop\_time.

The task\_id AVP, included in accounting REQUEST START, STOP, and WATCHDOG variants, correlates session initiation, watchdog updates, and termination packets; each associated START, STOP, and WATCHDOG packet must contain matching task-id AVPs.

```
task_id=13578642
```

The start\_time AVP, included in accounting REQUEST START and WATCHDOG variants, specifies the time at which a specific accounting request was initiated. The start time is expressed as the number of seconds elapsed since January 1, 1970 00:00:00 UTC.

```
start_time=1286790650
```

The stop\_time AVP, included in accounting REQUEST STOP variants, specifies the time at which a specific accounting session was terminated. The stop time is expressed as the number of seconds elapsed since January 1, 1970 00:00:00 UTC.

```
stop_time=1286794250
```

The service AVP, included in accounting REQUEST START, STOP, and WATCHDOG variants, identifies the function subject to accounting. In the case of the current implementation, the attribute value is always shell . Consequently the attribute takes the follow format:

```
service=shell
```

The cmd AVP, included in accounting REQUEST WATCHDOG variants, identifies the specific ACLI command to be processed by the accounting service. The command is passed in its entirety, from the administrative configuration root, **configure terminal**, through the final command argument. For example,

```
cmd=configure terminal security authentication type tacacsplus
```

Note the equal sign (=) used in the attribute examples, indicating that all are mandatory arguments.

## Accounting REPLY Packet

The TACACS+ daemon sends an accounting REPLY packet to the Oracle Communications Session Border Controller to report accounting results.

The accounting REPLY packet format is as follows.

```
+-----+
|           Common Header           |
|                                     |
|           type contains 0x3        |
+-----+-----+-----+
| server_msg_len | data_len         |
+-----+-----+-----+
| status | server_msg ...           |
+-----+-----+-----+
|           data ...                |
+-----+-----+-----+
```

server\_msg\_len

This 16-bit field contains the length, in octets, of the server\_msg field.

data\_len

This 16-bit field contains the length, in octets, of the data field.

status

This 8-bit field contains the status of the previous accounting request. Supported values are:

0x1 — Success

0x2 — Error/Failure

server\_msg

## Getting Started

This optional variable length field can contain a message intended for display to the user. This field is unused in the current TACACS+ implementation.

data

This optional variable length field can contain miscellaneous data. This field is unused in the current TACACS+ implementation.

### Accounting Scenario

The Oracle Communications Session Border Controller initiates the accounting session with an accounting REQUEST START.

-----			
Common Header			
type contains 0x3			
-----			
flags	authen_	priv_lvl	authen-
	method		type
0x02	0x05	0x00	0x01
-----			
authen_	user_len	port_len	rem_addr
service			_len
0x01	N	N	N
-----			
arg_cnt	arg1_len	arg2_len	arg3_len
3	N	N	N
-----			
user			
login name of an admin user			
-----			
port			
tty10			
-----			
rem_addr			
localhost address			
-----			
AVP			
task-id=13578642			
-----			
AVP			
start_time=1286790650			
-----			
AVP			
service=shell			
-----			

- The flags field contains an enumerated value ( 0x02 ) that identifies an accounting REQUEST START.
- The authen\_method field specifies the method used to authenticate the ACCOUNTING subject — 0x05 for TAC\_PLUS\_AUTHEN\_METHOD\_LOCAL (authentication by the client).
- The priv\_lvl field specifies the privilege level requested by the user — 0x00 for TAC\_PLUS\_PRIV\_LVL\_MIN.
- The authen\_type field specifies the authentication methodology — 0x01 for TAC\_PLUS\_AUTHEN\_TYPE\_ASCII (simple login).
- The authen\_service field specifies the requesting service — 0x01 for TAC\_PLUS\_AUTHEN\_SVC\_LOGIN (login service).
- The user\_len field contains the length, in octets, of the user field.
- The port\_len field contains the length, in octets, of the port field.
- The rem\_addr\_len field contains the length, in octets, of the rem\_addr field.
- The arg\_cnt field contains the number of arguments in the message body.
- The arg1\_len field contains the length, in octets, of the task\_id AVP.



- The arg2\_len field contains the length, in octets, of the start\_time AVP.
- The arg3\_len field contains the length, in octets, of the service AVP.
- The user field contains the login name of an admin user.
- The port field contains the name of the Oracle Communications Session Border Controller port on which authentication is taking place. Following Cisco Systems convention, this field contains the string tty10 .
- The rem\_addr field specifies the location of the user to be authenticated. This field contains the localhost address.
- The arg1 field contains the mandatory task\_id AVP.
- The arg2 field contains the mandatory start\_time AVP.
- The arg3 field contains the mandatory service AVP.

The TCACS+ daemon returns an accounting REPLY reporting the status, indicating that accounting has started.

```

+-----+
|               Common Header               |
|               type contains 0x3           |
+-----+-----+
| server_msg_len | data_len |
|      0         |      0   |
+-----+-----+
| status        |
| 0x01         |
+-----+

```

- The server\_msg\_len and data\_len fields both contain a value of 0 , as required by the TACACS+ protocol.
- The status field specifies the authorization status — 0x01 for TAC\_PLUS\_ACCT\_STATUS\_SUCCESS (accounting processed).

The Oracle Communications Session Border Controller reports ACLI command execution with an accounting REQUEST WATCHDOG.

```

+-----+-----+-----+-----+
|               Common Header               |
|               type contains 0x3           |
+-----+-----+-----+-----+
| flags | authen_ | priv_lvl | authen- |
|       | method  |         | type    |
| 0x08  | 0x05   | 0x00   | 0x01   |
+-----+-----+-----+-----+
| authen_ | user_len | port_len | rem_addr |
| service |          |          | _len    |
| 0x01   | N       | N       | N       |
+-----+-----+-----+-----+
| arg_cnt | arg1_len | arg2_len | arg3_len |
| 4      | N       | N       | N       |
+-----+-----+-----+-----+
| arg4_len |          user |
|          | login name of admin user |
+-----+-----+-----+-----+
|          port |
|          tty10 |
+-----+-----+-----+-----+
|          rem_addr |
|          localhost address |
+-----+-----+-----+-----+
|          AVP |
|          task-id=13578642 |
+-----+-----+-----+-----+
|          AVP |
|          start_time=1286790650 |
+-----+-----+-----+-----+
|          AVP |

```

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```
+-----+
|          service=shell          |
+-----+
|          AVP                    |
| cmd=configure terminal security |
+-----+
```

- The flags field contains an enumerated value ( 0x08 ) that identifies an accounting REQUEST WATCHDOG.
- The authen\_method field specifies the method used to authenticate the ACCOUNTING subject — 0x05 for TAC\_PLUS\_AUTHEN\_METHOD\_LOCAL (authentication by the client).
- The priv\_lvl field specifies the privilege level requested by the user — 0x00 for TAC\_PLUS\_PRIV\_LVL\_MIN.
- The authen\_type field specifies the authentication methodology — 0x01 for TAC\_PLUS\_AUTHEN\_TYPE\_ASCII (simple login).
- The authen\_service field specifies the requesting service — 0x01 for TAC\_PLUS\_AUTHEN\_SVC\_LOGIN (login service).
- The user\_len field contains the length, in octets, of the user field.
- The port\_len field contains the length, in octets, of the port field.
- The rem\_addr\_len field contains the length, in octets, of the rem\_addr field.
- The arg\_cnt field contains the number of arguments in the message body.
- The arg1\_len field contains the length, in octets, of the task\_id AVP.
- The arg2\_len field contains the length, in octets, of the start\_time AVP.
- The arg3\_len field contains the length, in octets, of the service AVP.
- The arg4\_len field contains the length, in octets, of the cmd AVP.
- The user field contains the login name of an admin user.
- The port field contains the name of the Oracle Communications Session Border Controller port on which authentication is taking place. Following Cisco Systems convention, this field contains the string tty10 .
- The rem\_addr field specifies the location of the user to be authenticated. This field contains the localhost address.
- The arg1 field contains the mandatory task\_id AVP.
- The arg2 field contains the mandatory start\_time AVP.
- The arg3 field contains the mandatory service AVP.
- The arg4 field contains the mandatory cmd AVP.

The TCACS+ daemon returns an accounting REPLY reporting the status, indicating that the ACLI operation has been processed.

```
+-----+
|          Common Header          |
|          type contains 0x3      |
+-----+
| server_msg_len | data_len |
|          0    |          0 |
+-----+
| status |
| 0x01  |
+-----+
```

- The server\_msg\_len and data\_len fields both contain a value of 0 , as required by the TACACS+ protocol.
- The status field specifies the authorization status — 0x01 for TAC\_PLUS\_ACCT\_STATUS\_SUCCESS (accounting processed).

The Oracle Communications Session Border Controller reports an admin user logout or timeout with an accounting REQUEST STOP.

```
+-----+
|          Common Header          |
|          type contains 0x3      |
+-----+
| flags | authen_ | priv_lvl | authen- |
```

0x04	method	0x05	0x00	type	0x01
authen_	user_len	port_len	rem_addr		
service			_len		
0x01	N	N	N		
arg_cnt	arg1_len	arg2_len	arg3_len		
3	N	N	N		
user					
login name of an admin user					
port					
tty10					
rem_addr					
localhost address					
AVP					
task-id=13578642					
AVP					
stop_time=1286790650					
AVP					
service=shell					

- The flags field contains an enumerated value ( 0x04 ) that identifies an accounting REQUEST STOP.
- The authen\_method field specifies the method used to authenticate the ACCOUNTING subject — 0x05 for TAC\_PLUS\_AUTHEN\_METHOD\_LOCAL (authentication by the client).
- The priv\_lvl field specifies the privilege level requested by the user — 0x00 for TAC\_PLUS\_PRIV\_LVL\_MIN.
- The authen\_type field specifies the authentication methodology — 0x01 for TAC\_PLUS\_AUTHEN\_TYPE\_ASCII (simple login).
- The authen\_service field specifies the requesting service — 0x01 for TAC\_PLUS\_AUTHEN\_SVC\_LOGIN (login service).
- The user\_len field contains the length, in octets, of the user field.
- The port\_len field contains the length, in octets, of the port field.
- The rem\_addr\_len field contains the length, in octets, of the rem\_addr field.
- The arg\_cnt field contains the number of arguments in the message body.
- The arg1\_len field contains the length, in octets, of the task\_id AVP.
- The arg2\_len field contains the length, in octets, of the start\_time AVP.
- The arg3\_len field contains the length, in octets, of the service AVP.
- The user field contains the login name of an admin user.
- The port field contains the name of the Oracle Communications Session Border Controller port on which authentication is taking place. Following Cisco Systems convention, this field contains the string tty10 .
- The rem\_addr field specifies the location of the user to be authenticated. This field contains the localhost address.
- The arg1 field contains the mandatory task\_id AVP.
- The arg2 field contains the mandatory start\_time AVP.
- The arg3 field contains the mandatory service AVP.

The TCACS+ daemon returns an accounting REPLY reporting the status, indicating that accounting has terminated.

Common Header	
type contains 0x3	

## Getting Started

```
| server_msg_len | data_len |
| 0              | 0        |
+-----+
| status        |
| 0x01         |
+-----+
```

- The `server_msg_len` and `data_len` fields both contain a value of 0 , as required by the TACACS+ protocol.
- The `status` field specifies the authorization status — 0x01 for `TAC_PLUS_ACCT_STATUS_SUCCESS` (accounting processed).

## TACACS+ Configuration

Configuration of TACACS+ consists of the following steps.

1. Enable TACACS+ client services
2. Specify one or more TACACS+ servers (daemons)



**Note:** TACACS servers must be reachable from the Oracle Communications Session Border Controller's media interfaces. Communications to and from TACACS servers can not be from the management interfaces.

### Enable TACACS+ Client Services

Use the following procedure to enable specific TACACS+ client AAA services.

1. Access the **authentication** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# authentication
ACMEPACKET(authentication)#
```

2. Use the required **type** attribute to specify the authentication protocol.

Supported values are **diameter**, **local** (the default, indicating that authentication determinations are referred to a local database), **radius**, and **tacacs**.

Specify **tacacs** to enable the TACACS+ AAA protocol.

```
ACMEPACKET(authentication)# type tacacs
ACMEPACKET(authentication)#
```

3. Use **tacacs-authorization** to enable or disable command-based authorization of admin users. Assuming **type** is set to `tacacs` , by default, authorization is enabled.

```
ACMEPACKET(authentication)# tacacs-authorization enabled
ACMEPACKET(authentication)#
```

4. Use **tacacs-accounting** to enable or disable accounting of admin CLI operations. Assuming **type** is set to `tacacs` , by default, accounting is enabled.

```
ACMEPACKET(authentication)# tacacs-accounting enabled
ACMEPACKET(authentication)#
```

5. Use the **server-assigned-privilege** attribute to enable a proprietary TACACS+ variant that, after successful user authentication, adds an additional TACACS+ request/reply exchange. During the exchange, the Security Gateway requests the privilege level of the newly authenticated user. In response, the TACACS+ daemon returns the assigned privilege level, either `user` or `admin` . `user` accounts are denied access to the **enable** command, thus barring them from configuration level commands.

By default, this attribute is **disabled** — meaning that no privilege level information is exchanged.

Set this attribute to **enabled** to initiate the proprietary variant behavior.

```
ACMEPACKET(authentication)# server-assigned-privilege enabled
ACMEPACKET(authentication)#
```

- Use the **management-strategy** attribute to identify the selection algorithm used to choose among multiple available TACACS+ daemons.

Retain the default value (hunt) if only a single daemon is available.

Available algorithms are hunt and roundrobin .

- **hunt** — for the first transaction the Security Gateway selects the initially configured TACACS+ daemon. As long as that daemon is online and operational, the Security Gateway directs all AAA transactions to it. Otherwise, the Security Gateway selects the second-configured daemon. If the first and second daemons are offline or non-operational, the next-configured daemon is selected, and so on through the group of available daemons.
- **roundrobin** — for the first transaction the Security Gateway selects the initially configured TACACS+ daemon. After completing the first transaction, selects each daemon in order of configuration — in theory, evenly distributing AAA transactions to each daemon over time.

```
ACMEPACKET (authentication) # management-strategy roundrobin
ACMEPACKET (authentication) #
```

- Type **done** to save your configuration.

## Specify TACACS+ Servers

Use the following procedure to specify one or more TACACS+ servers (daemons).

- Access the **tacacs-servers** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# authentication
ACMEPACKET(authentication)# tacacs-servers
ACMEPACKET(tacacs-servers)#
```

- Use the **address** attribute to specify the IP address of this TACACS+ daemon.

```
ACMEPACKET (tacacs-servers) # address 172.30.0.6
ACMEPACKET (tacacs-servers) #
```



**Note:** This address must be reachable from a media interface. The TACACS server should not be on the management interface's network.

- Use the **port** attribute to identify the daemon port that receives TACACS+ client requests.

Provide a port number within the range 1025 through 65535, or retain the default value, 49, the well-known TACACS+ port.

```
ACMEPACKET (tacacs-servers) # port 49
ACMEPACKET (tacacs-servers) #
```

- Use the **state** attribute to specify the availability of this TACACS+ daemon.

Select enabled (the default) or disabled.

Only TACACS+ daemons that are in the enabled state are considered when running the server-selection algorithm.

```
ACMEPACKET (tacacs-servers) # state enabled
ACMEPACKET (tacacs-servers) #
```

- Use the **realm-id** attribute to identify the realm that provides access to this TACACS+ daemon.

```
ACMEPACKET (tacacs-servers) # realm-id accounting
ACMEPACKET (tacacs-servers) #
```



**Note:** This realm must be reachable from a media interface.

- Retain the default value for the **authentication-methods** attribute to specify support for all TACACS+ authentication methods (pap, chap, and ascii).

- **ascii** — simple login, the Session Director prompts user for username and password
- **pap** — similar to ascii method, but username and password are encapsulated in a PAP header

## Getting Started

- chap — authentication based on a shared-secret, which is not passed during the authentication process

```
ACMEPACKET (tacacs-servers) # authentication-methods all
ACMEPACKET (tacacs-servers) #
```

7. Use the **secret** attribute to provide the shared-secret used by the TACACS+ client and the daemon to encrypt and decrypt TACACS+ messages. The identical shared-secret must be configured on associated TACACS+ clients and daemons.

Enter a 16-digit string, and ensure that the identical value is configured on the TACACS+ daemon.

```
ACMEPACKET (tacacs-servers) # secret 1982100754609236
ACMEPACKET (tacacs-servers) #
```

8. Use the **dead-time** attribute to specify, in seconds, the quarantine period imposed upon TACACS+ daemons that become unreachable. Quarantined servers are not eligible to participate in the server-selection algorithm.

Supported values are integers within the range 10 through 10000 seconds, with a default value of 10 .

```
ACMEPACKET (tacacs-servers) # dead-interval 120
ACMEPACKET (tacacs-servers) #
```

9. Type **done** to save your configuration.

10. Repeat Steps 1 through 10 to configure additional TACACS+ daemons.



**Note:** After configuring TACACS+ daemons, complete TACACS+ configuration by compiling a list of available daemons.

11. From superuser mode, use the following command sequence to access authentication configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure)# security
ACMEPACKET (security)# authentication
ACMEPACKET (authentication)#
```

12. Use the **management-servers** attribute to identify one or more TACACS+ daemons available to provide AAA services.

Daemons are identified by IP address and must have been previously configured as described above.

The following example identifies three available TACACS+ daemons. Note that the list is delimited by left and right parentheses, and list items are separated by space characters.

```
ACMEPACKET (tacacs-servers) # management-servers (172.30.0.6 172.30.1.8
172.30.2.10)
ACMEPACKET (tacacs-servers) #
```

The following example deletes the current list.

```
ACMEPACKET (tacacs-servers) # management-servers ()
ACMEPACKET (tacacs-servers) #
```

## Managing TACACS+ Operations

TACACS+ management is supported by the following utilities.

### TACACS+ MIB

An Oracle proprietary MIB provides external access to TACACS+ statistics.

MIB counters are contained in the apSecurityTacacsPlusStatsTable that is defined as follows.

```
SEQUENCE {
  apSecurityTacacsPlusCliCommands          Counter32
  apSecurityTacacsPlusSuccess Authentications Counter32
  apSecurityTacacsPlusFailureAuthentications Counter32
  apSecurityTacacsPlusSuccess Authorizations Counter32
  apSecurityTacacsPlusFailureAuthorizations Counter32
}
```

apSecurityTacacsPlusStats Table (1.3.6.1.4.1.9148.3.9.9.4)		
Object Name	Object OID	Description
apSecurityTacacsCliCommands	1.3.6.1.4.1.9148.3.9.1.4.3	Global counter for ACLI commands sent to TACACS+ Accounting
apSecurityTacacsSuccess Authentications	1.3.6.1.4.1.9148.3.9.1.4.4	Global counter for the number of successful TACACS+ authentications
apSecurityTacacsFailureAuthentications	1.3.6.1.4.1.9148.3.9.1.4.5	Global counter for the number of unsuccessful TACACS+ authentications
apSecurityTacacsSuccess Authorizations	1.3.6.1.4.1.9148.3.9.1.4.6	Global counter for the number of successful TACACS+ authorizations
apSecurityTacacsFailure Authorizations	1.3.6.1.4.1.9148.3.9.1.4.7	Global counter for the number of unsuccessful TACACS+ authorizations

### SNMP Trap

SNMP traps are issued when

- a TACACS+ daemon becomes unreachable
- an unreachable TACACS+ daemon becomes reachable
- an authentication error occurs
- an authorization error occurs

### ACLI show Command

The `show tacacs stats` command displays the following statistics.

- number of ACLI commands sent for TAGACS+ accounting
- number of successful TACACS+ authentications
- number of failed TACACS+ authentications
- number of successful TACACS+ authorizations
- number of failed TACACS+ authentications
- the IP address of the TACACS+ daemon used for the last transaction

### TACACS+ Logging

All messages between the Oracle Communications Session Border Controller and the TACACS+ daemon are logged in a cleartext format, allowing an admin user to view all data exchange, except for password information.

## Customizing Your ACLI Settings

This section describes several ways you can customize the way you log into the ACLI and the way the ACLI displays information. Where applicable, these descriptions also contain instructions for configuration.

### Disabling the Second Login Prompt

With this feature enabled, the Oracle logs you in as a Superuser (i.e., in administrative mode) regardless of your configured privilege level for either a Telnet or an SSH session. However, if you log via SSH, you still need to enter the password for local or RADIUS authentication.

### Disabling the Second Login Prompt Configuration

You disable the second login prompt in the authentication configuration.

To disable the second login prompt:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal  
ACMEPACKET(configure)#
```

2. Type **security** and press Enter.

```
ACMEPACKET(configure)# security  
ACMEPACKET(security)#
```

3. Type **authentication** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(security)# authentication  
ACMEPACKET(authentication)#
```

4. **login-as-admin**—Set this parameter to **enabled** if you want users to be logged automatically in Superuser (administrative) mode. The default for this parameter is **disabled**.

5. Save and activate your configuration.

### Persistent ACLI more Parameter

To make using the ACLI easier, the Oracle Communications Session Border Controller provides a paging feature controlled through the ACLI **cli more** command (which you can set to enabled or disabled). Disabled by default, this feature allows you to control how the Oracle Communications Session Border Controller displays information on your screen during a console, Telnet, or SSH session. This command sets the paging feature on a per session basis.

Customers who want to set the paging feature so that settings persist across sessions with the Oracle Communications Session Border Controller can set a configuration parameter that controls the paging feature. Enabling this parameter lets you set your preferences once rather than having to reset them each time you initiate a new session with the Oracle Communications Session Border Controller.

### Persistent ACLI more Parameter Configuration

To set the persistent behavior of the ACLI more feature across sessions:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal  
ACMEPACKET(configure)#
```

2. Type **system** and press Enter.

```
ACMEPACKET(configure)# system  
ACMEPACKET(system)#
```

3. Type **system-config** and press Enter.

```
ACMEPACKET(system)# system-config  
ACMEPACKET(system-config)#
```

If you are adding this feature to an existing configuration, you need to select the configuration (using the ACLI **select** command) before making your changes.

4. **cli-more**—Set this parameter to **enabled** if you want the ACLI more paging feature to work persistently across console, Telnet, or SSH sessions with the Oracle Communications Session Border Controller. If you want to continue to set this feature on a per session basis, leave this parameter set to **disabled** (default).

5. Save and activate your configuration.



## **Customized Login Banner**

A text file can be put on the Oracle Communications Session Border Controller to be used as a banner to be printed before each login. The file must be called `/code/banners/banner.txt`. The contents of this file will be printed before each User Access Verification login sequence. The limits are that no more than 79 characters per line and no more than 20 lines from the `banner.txt` file will be printed.

The `banner.txt` file used for the ACLI customized login banner has to be saved in the `/code/banners` directory. If that directory does not already exist on your system, you do not have to create the directory prior to placing the banner file because the Oracle Communications Session Border Controller will create it upon boot if it does not exist.



---

## System Configuration

This chapter explains how to configure system-level functionality for the Oracle Communications Session Border Controller. Both physical and network interfaces as well as general system parameters are required to configure your Oracle Communications Session Border Controller for service. Accounting functionality, SNMP configurations, trap configurations, and host routes are optional.

The following configurations are explained in this chapter:

- General system parameters—used for operating and identification purposes. In general, the informational fields have no specific effect on services, but are important to keep populated. The default gateway parameter is included here. It requires special attention since its configuration is dependent on the type of traffic the Oracle Communications Session Border Controller is servicing.
- Physical and network interfaces—enables the Oracle Communications Session Border Controller to communicate with any network element. Interfaces are one of the most basic configurations you need to create.
- SNMP—used for monitoring system health throughout a network.
- Syslogs and Process logs—used to save a list of system events to a remote server for analysis and auditing purposes.
- Host routes—used to instruct the Oracle Communications Session Border Controller host how to reach a given network that is not directly connected to a local network interface.

---

## General System Information

This section explains the parameters that encompass the general system information on a Oracle Communications Session Border Controller.

### System Identification

Global system identification is used primarily by the Oracle Communications Session Border Controller to identify itself to other systems and for general identification purposes.

### Connection Timeouts

It is important to set administrative session timeouts on the Oracle Communications Session Border Controller for security purposes. If you leave an active configuration session unattended, reconfiguration access is left open to anyone. By setting a connection timeout, only a short amount of time needs to elapse before the password is required for Oracle Communications Session Border Controller access.

## System Configuration

---

Timeouts determine the specified time period that must pass before an administrative connection is terminated. Any subsequent configuration activity can only be performed after logging in again to the Oracle Communications Session Border Controller. The timeout parameter can be individually specified for Telnet sessions and for console port sessions.

After the Telnet timeout passes, the Telnet session is disconnected. You must use your Telnet program to log in to the Oracle Communications Session Border Controller once again to perform any further configuration activity.

After the console timeout passes, the console session is disconnected. The current session ends and you are returned to the login prompt on the console connection into the Oracle Communications Session Border Controller.

## Configuring General System Information

---

This section explains how to configure the general system parameters, timeouts, and the default gateway necessary to configure your Oracle Communications Session Border Controller.

To configure general system information:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # system
```

3. Type **system-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (system) # system-config  
ACMEPACKET (system-config) #
```

The following is an example what a general system information configuration might look like. Parameters not described in this section are omitted below.

```
ACMEPACKET (system-config) # show  
system-config  
  hostname                test1  
  description             Example SD  
  location                 Row 3, Rack 4, Slot 451  
  default-gateway         10.0.2.1  
  telnet-timeout          1000  
  console-timeout         1000  
  last-modified-date      2004-12-08 20:15:43
```

When showing a single-instance configuration element such as **system-config**, you must first use the **select** command to select the configuration element prior to viewing.

## System Identification

You must specify identification parameters for this Oracle Communications Session Border Controller.

Set the following parameters to configure the system identification:

1. **hostname**—Set the primary hostname used to identify the system. This parameter is used by the software for informational purposes.
2. **description**—Enter a textual description of the system. This parameter is used for informational purposes.
3. **location**—Set a location description field for your system. This parameter is used for informational purposes. For example, you could include the site name and address of the location where the Oracle Communications Session Border Controller system chassis is located.
4. **default-gateway**—Set the default gateway for this Oracle Communications Session Border Controller. This is the egress gateway for traffic without an explicit destination. The application of your Oracle Communications Session Border Controller determines the configuration of this parameter.

## Configuring Connection and Debug Logging Timeouts

Configure the timeouts for terminal sessions on this Oracle Communications Session Border Controller. These parameters are optional.

Set the following parameters to configure the connection timeouts:

1. **telnet-timeout**—Set the Telnet timeout to the number of seconds you want the Oracle Communications Session Border Controller to wait before it disconnects a Telnet session or an SSH session. The default value is 0. The valid range is:
  - Minimum—0
  - Maximum—65535
2. **console-timeout**—Set the console timeout to the number of seconds you want the Oracle Communications Session Border Controller to wait before it ends the console session. The default value is 0. The valid range is:
  - Minimum—0
  - Maximum—65535
3. **debug-timeout**—Set the time in seconds you want to use for the debug timeout. This is the time allowed before the Oracle Communications Session Border Controller times out log levels for system processes set to debug using the ACLI **notify** and **debug** commands.

This command does not affect log levels set in your configuration (using parameters such as **system-config>process-log-level**) or those set using the ACLI **log-level** command.

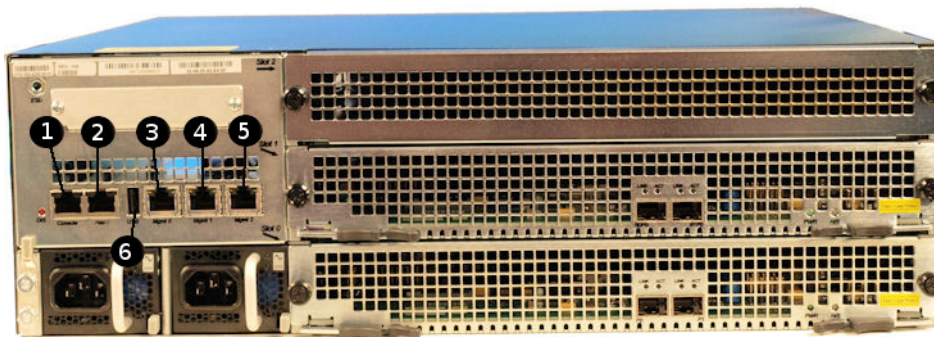
The valid range is:

- Minimum—0
- Maximum—65535

## Acme Packet 6300 Physical Interfaces

### Acme Packet 6300 Management Interfaces

The Acme Packet 6300 includes 3 x 10/100/1000 Ethernet interfaces, a console port, and an alarm port (“dry contact alarms”). Network management interfaces are referred to as wancom0, wancom1, and wancom2. There is no front facing console port. Maintenance and control ports are located on the left-rear of the chassis, above the power supplies. See the following diagram:



1. Console port
2. Alarm port
3. wancom0
4. wancom1
5. wancom2
6. USB Port (for Acme Packet personnel access only)


## System Configuration

---

### Acme Packet 6300 Media Interfaces

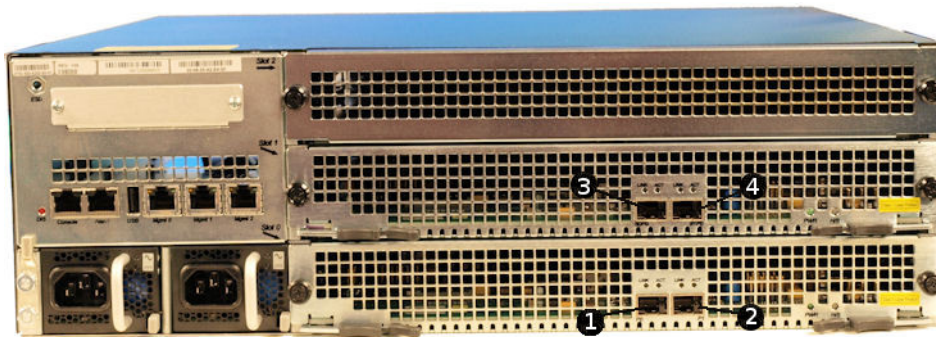
The Acme Packet 6300 has 3 PHY card slots. The bottom slot, slot0, and the middle slot, slot1, are network-facing media interfaces. The top slot, slot2, is for special processing hardware such as transcoding and is referred to as the “resource” slot. Standard Acme Packet 6300 PHY cards contain 2 x 10-gigabit Ethernet ports.

Ensure that the first 2x10Gig NIU in your system populates slot0. If you are using a second 2x10Gig NIU, it should be inserted into slot1.

 **Note:** Do not insert any 2x10Gig NIU in the resource slot (slot2).

The Acme Packet 6300 platform supports up to 4 x 10 gigabit media ports each running at full duplex line rate for all packet sizes.

Ports on the Acme Packet 6300 are numbered from left to right starting at 0. For physical interface configuration, see the following diagram



1. Slot 0, Port 0
2. Slot 0, Port 1
3. Slot 1, Port 0
4. Slot 1, Port 1

## Physical Interfaces: Acme Packet 3820 and Acme Packet 4500

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There are two sets of physical interfaces on the network interface unit (NIU) used in the Acme Packet 3820 and Acme Packet 4500. These interfaces are located on the rear of the system chassis.

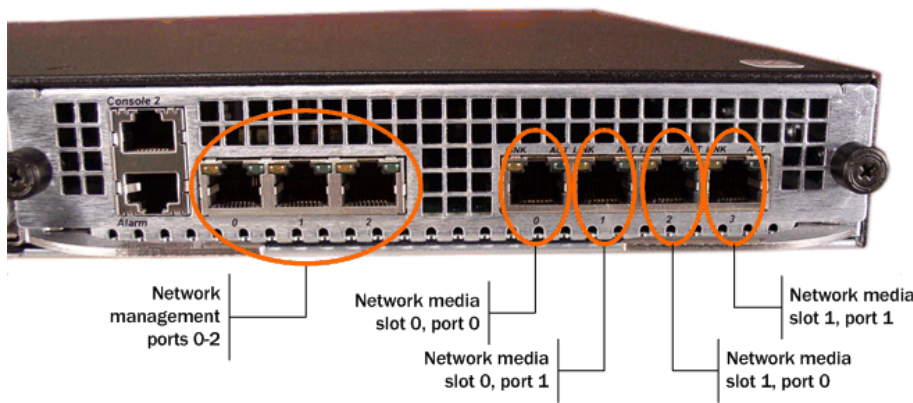
- Media interfaces are on the network interface unit (NIU); they are also referred to as network media ports
- Management interfaces are also on the NIU; they are also referred to as network management ports

The following picture of the NIU shows you how the network media and network management ports appear. These designations are an important point of reference when you set up physical interface configurations. Note that the slot parameter for network management ports will always be set to zero (0).

### Network Media Interfaces

The NIU installed on your Acme Packet 3820 or Acme Packet 4500 determines the number of interfaces, hardware protocol, and connection speed available for media and signaling traffic.

- The NIU offers either four ports , and can use single mode or multimode fiber with an LC connector.
  - 4-port GigE copper (RJ45)
  - 4-port GigE SFP (LX, SX, or Copper)
  - 4-port GigE SFP with QoS and IPsec (LX, SX, or Copper)
  - 4-port GigE SFP with IPsec (LX, SX, or Copper)
  - 4-port GigE SFP with QoS (LX, SX, or Copper)
  - 4-port GigE SFP ETC NIU (LX, SX, or Copper)



## Network Management Interfaces

The first management interface (labeled port 0 on the NIU's group of management ports) is used to carry traffic such as:

- SNMP
- Telnet
- SSH
- FTP
- ACP/XML
- Logs sent from the Oracle Communications Session Border Controller
- Boot the Oracle Communications Session Border Controller from a remote file server

The other two rear interfaces (port 1 and port 2) are used for state replication for high availability (HA). For HA, these interfaces on the Acme Packet 3820 and Acme Packet 4500 are directly connected by a crossover cable.

The following table summarizes the physical interface configuration parameters, which interface they are applicable to, and whether they are required.

Parameter	Network Media Interface	Network Management Interface
name	R	R
operation-type	R	R
port	R	R
slot	R	R
virtual-mac	O	I
admin-state	R	I
auto-negotiation	R	I
duplex-mode	R	I
speed	R	I
wancom-health-score	I	O
R = Required, O = Optional, I = Invalid		

## Before You Configure

This section describes steps you should take prior to configuring physical interfaces.

Before you configure a physical interface:

## System Configuration

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1. Decide on the number and type of physical interfaces you need.

For example, you might have one media interface connecting to a private network and one connecting to the public network. You might also need to configure maintenance interfaces for HA functionality.

2. Determine the slot and port numbering you will need to enter for the physical interfaces you want to configure. The graphic above can serve as your slot and port numbering reference.
3. If you are configuring your Acme Packet 4500 for HA, refer to the HA Nodes documentation and follow the instructions there for setting special parameters in the physical interface configuration.

## Physical Interface Configuration

This section describes how to configure physical interfaces.

1. Access the **phy-interface** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# system
ACMEPACKET(system)#phy-interface
ACMEPACKET(phy-interface)#
```

2. **name**—Set a name for the interface using any combination of characters entered without spaces. For example: **s0p0**.
3. **admin-state**—Leave the administrative state parameter set to **enabled** to receive and send traffic on this interface. Select **disabled** to prevent media and signaling from being received and sent. The default for this parameter is **enabled**.
4. **operation-type**—Select the type of physical interface connection to use. Refer to the appropriate platform section to identify how this parameter corresponds to an external interface. The default value is **control**. The valid values are:
  - **media**—Use this value for configuring the media interfaces which carry production traffic.
  - **maintenance**—Use this value for configuring the management physical interfaces, used for management protocols or HA.
  - **control**—This legacy parameter may also be used to configure the management physical interfaces.
5. **slot**—Set the slot number for this physical interface. Refer to the appropriate platform section to identify how this parameter corresponds to an external interface.
6. **port**—Set the port number for this physical interface. Refer to the appropriate platform section to identify how this parameter corresponds to an external interface.
7. **auto-negotiation**—Leave this parameter set to **enabled** so that the Oracle Communications Session Border Controller and the device to which it is linked can automatically negotiate the duplex mode and speed for the link.

If auto-negotiation is enabled, the Oracle Communications Session Border Controller begins to negotiate the link to the connected device at the duplex mode you configure. If auto-negotiation is disabled, then the Oracle Communications Session Border Controller will not engage in a negotiation of the link and will operate only at the duplex mode and speed you set. The default is **enabled**. The valid values are:

- enabled | disabled
8. **duplex-mode**—Set the duplex mode. The default is **full**; this field is only used if the auto-negotiation field is set to disabled.

Given an operating speed of 100 Mbps, full duplex mode lets both devices on a link send and receive packets simultaneously using a total bandwidth of 200 Mbps. Given the same operating speed, half duplex mode limits the devices to one channel with a total bandwidth of 100 Mbps. The valid values are:

- half | full
9. **speed**—Set the speed in Mbps of the physical interfaces; this field is only used if the auto-negotiation field is set to disabled. **100** is the default. The valid values are:
    - 10 | 100 | 1000



- 10. virtual-mac**—Refer to Oracle Communications Session Border Controller High Availability (HA) documentation to learn how to set this parameter on an HA interface.
- 11. Type done** to save your configuration.

### Auto-negotiation for Management Interfaces

For network management interfaces (wancom), set the parameters that enable or disable auto-negotiation, duplex mode, and the data rate.

Set the following parameters to configure auto-negotiation for management interfaces:

- 1. auto-negotiation**—Leave this parameter set to **enabled** so that the Oracle Communications Session Border Controller and the device to which it is linked can automatically negotiate the duplex mode and speed for the link.

If auto-negotiation is enabled, the Oracle Communications Session Border Controller begins to negotiate the link to the connected device at the duplex mode you configure. If auto-negotiation is disabled, then the Oracle Communications Session Border Controller will not engage in a negotiation of the link and will operate only at the duplex mode and speed you set. The default is **enabled**. The valid values are:

- enabled | disabled

- 2. duplex-mode**—Set the duplex mode. The default is **full**.

Given an operating speed of 100 Mbps, full duplex mode lets both devices on a link send and receive packets simultaneously using a total bandwidth of 200 Mbps. Given the same operating speed, half duplex mode limits the devices to one channel with a total bandwidth of 100 Mbps. The valid values are:

- half | full

- 3. speed**—Set the speed in Mbps of the management physical interfaces; this field is only used if the auto-negotiation field is set to disabled. **100** is the default. The valid values are:

- 10 | 100 | 1000

### HA Configuration

Refer to this guide's *HA Nodes documentation* for more information about when and how to configure **virtual-mac** and **wancom-health-score** parameters. If you are not using HA, you can leave these parameters set to their defaults.

## Interface Utilization Graceful Call Control Monitoring and Fault Management

When you enable this feature, the Oracle Communications Session Border Controller monitors network utilization of its media interfaces and sends alarms when configured thresholds are exceeded. You can also enable overload protection on a per-media interface basis, where the Oracle Communications Session Border Controller will prevent call initializations during high traffic but still allow established calls to continue if traffic passes the critical threshold you define.

### Calculation Overview

When enabled to do so, the Oracle Communications Session Border Controller performs a network utilization calculation for each of its media ports. This calculation takes into account rates of receiving and transmitting data, the speed at which each is taking place, and the quality of data traversing the interface. The Oracle Communications Session Border Controller keeps statistics for each media port so it can compare previously- and newly-retrieved data. For heightened accuracy, calculations are performed with milliseconds (rather than with seconds).

### Alarms

In the physical interface configuration, you can establish up to three alarms per media interface—one each for minor, major, and critical alarm severities. These alarms do not have an impact on your system's health score. You set the threshold for an alarm as a percentage used for receiving and transmitting data.

For example, you might configure the following alarms:

- Minor, set to 50%
- Major, set to 70%
- Critical, Set to 90%

When the utilization percentage hits 50%, the system generates a minor alarm. At 70%, the system clears the minor alarm and issues a major one. And at 90%, the system clears the major alarm and issues a critical one. At that point, if you have overload protection enabled, the system will drop call initiations but allow in-progress calls to complete normally.

To prevent alarm thrashing, utilization must remain under the current alarm threshold for 10 seconds before the system clears the alarm and rechecks the state.

### Alarm Configuration

This section shows you how to configure alarm thresholds and overload protection per media interface.

#### Configuring Utilization Thresholds for Media Interfaces

To configure utilization thresholds for media interfaces:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **system** and press Enter.

```
ACMEPACKET(configure)# system
ACMEPACKET(system)#
```

3. Type **phy-interface** and press Enter. **If you are adding this feature to an existing configuration, then remember you must select the configuration you want to edit.**

```
ACMEPACKET(system)# phy-interface
ACMEPACKET(phy-interface)#
```

4. Type **network-alarm-threshold** and press Enter.

```
ACMEPACKET(phy-interface)# network-alarm-threshold
ACMEPACKET(network-alarm-threshold)#
```

5. **severity**—Enter the severity for the alarm you want to fine for this interface: **minor** (default), **major**, or **critical**. Since the parameter defaults to minor, you must change the value if you want to define a major or critical alarm.
6. **value**—Enter the percentage of utilization (transmitting and receiving) for this interface that you want to trigger the alarm. For example, you might define a minor alarm with a utilization percentage of 50. Valid values are between 0 and 100, where 0 is the default.
7. Save your work.

#### Configuring Graceful Call Control

You can enable the Oracle Communications Session Border Controller to stop receiving session-initiating traffic on a media interface when the traffic for the interface exceeds the critical threshold you define for it.

To enable graceful call control:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **system** and press Enter.

```
ACMEPACKET(configure)# system
ACMEPACKET(system)#
```

3. Type **phy-interface** and press Enter. **If you are adding this feature to an existing configuration, then remember you must select the configuration you want to edit.**

```
ACMEPACKET(system)# phy-interface
ACMEPACKET(phy-interface)#
```

4. **overload-protection**—Change this parameter's value to enabled if you want to turn graceful call control on. Leave it set to disabled (default) if you do not want to use this feature.
5. Save your work.

## Network Interfaces

The network interface element specifies a logical network interface. In order to use a network port on a network interface, you must configure both the physical interface and the corresponding network interface configuration elements. If the network interface does not use VLANs tagging, ensure that the **sub-port-id** parameter is set to 0, the default value. When VLAN tags are used on a network interface, the valid **sub-port-id** value can range from 1-4096. The combination of the **name** parameter and the **sub-port-id** parameter must be unique in order to identify a discrete network interface.

## IP Configuration

A Oracle Communications Session Border Controller network interface has standard parameters common to nearly all IP network interfaces. There are a few fields that are unique to the Oracle Communications Session Border Controller.

### VLANs

VLANs are used to logically separate a single physical interface into multiple network interfaces. There are several applications for this like MPLS VPNs (RFC 2547), MPLS LSPs, L2VPNs (IPSec, L2TP, ATM PVCs), reusing address space, segmenting traffic, and maximizing the bandwidth into a switch or router. The range of services and management capabilities you can implement with VPNs is huge.

The primary applications of VLANs on the Oracle Communications Session Border Controller are VPNs and peering. Several peering partners may terminate their connections to a Oracle Communications Session Border Controller on a single physical interface. VLAN tags are used to segregate and correctly route the terminated traffic. The Oracle Communications Session Border Controller can support a maximum of 1024 VLANs per physical interface. Ingress packets that do not contain the correct VLAN tag will be dropped. All packets exiting on an egress interface will have the VLAN tag appended to them.

The Oracle Communications Session Border Controller can be included in an MPLS network through its connectivity to a PE router, which maps a MPLS VPN label to an 802.1q VLAN tag. Each Oracle Communications Session Border Controller can terminate different 802.1q VLANs into separate network interfaces, each of which can represent a different customer VPN.

### Overlapping Networks

Overlapping networks are when two or more private networks with the same addressing schemes terminate on one physical interface. The problem this creates can easily be solved by using VLAN tagging. For example, two 10.x.x.x networks terminating on one Oracle Communications Session Border Controller network interface will obviously not work. The Oracle Communications Session Border Controller includes the IP Address, Subnet Mask, and 802.1q VLAN tag in its Network Interface determination. This allows Oracle Communications Session Border Controller to directly interface to multiple VPNs with overlapping address space.

### Administrative Applications Over Media Interfaces

By default, the Oracle Communications Session Border Controller's FTP, ICMP, SNMP, SSH, and Telnet services cannot be accessed via the media interfaces. In order to enable these services, you must explicitly configure access by identifying valid source addresses for the specific applications. Doing such uses the Oracle Communications Session Border Controller's host-in-path (HIP) functionality.

When traffic is received on media interfaces, it is scanned for FTP, ICMP, SNMP, SSH, or Telnet packets. The configuration is set to identify the possible IP addresses where that traffic may be sourced from. When a match is made among packet type and source address, those packets are forwarded through the media interfaces to the processes running on the system's CPU.

Each media **network-interface**'s gateway should be configured so that off-subnet return traffic can be forwarded out the appropriate media interface. Also, it is advisable that no overlapping networks are configured between any media network interface and the administrative interfaces (wancom).

### Configurable MTU Size

Configurable MTU on per network-interface basis enables the user to set a different MTU on each network interface. It also enables the user to set a system wide default MTU for IPv6 and IPv4 network interfaces. System wide defaults can be set in **system-config** configuration object by setting **ipv6-signaling-mtu** or **ipv4-signaling-mtu**. Defaults are 1500 for both IPv6 and IPv4.

These settings can be overwritten for each network interface by setting **signaling-mtu** in **network-interface** configuration object. Default is 0 – meaning use the system wide MTU.

This feature applies to all Signaling packets generated by the Oracle Communications Session Border Controller. All UDP packets greater than the MTU will be fragmented. For all TCP connections we advertise MSS (Maximum Segment Size) TCP option in accordance with the configured MTU. MSS option is sent in SYN and SYN/ACK packets to let the other side of the TCP connection know what your maximum segment size is. This ensures that no TCP packet is greater than the configured MTU.

1. MTU settings do not apply to media packets.
2. UDP: MTU settings apply only to packets sent by the Oracle Communications Session Border Controller. The Oracle Communications Session Border Controller will continue to process received packets even if they exceed to the configured MTU.
3. Security Phy (IPsec) hardware only; We subtract 100 bytes from the configured MTU to allow for extra headers added by security protocols. This happens even when Security Phy (IPsec) is in clear mode (no security is being applied). Due to hardware limitations of the Security Phy (IPsec) it only allows one MTU per physical port. The maximum MTU of all network interfaces on a given physical port will be used as the MTU for that physical port.
4. The Call Recording feature is where we make a copy of a packet, encapsulate it in an IP-in-IP header and send it to a configured Call Recording Server (CRS). When Call Recording is enabled, to allow space for IP-in-IP encapsulation we reduce the MTU of the original packets to be to be the lesser of the two options listed below.
  - Original Destination network MTU minus size of IP-in-IP header.
  - CRS network interface's MTU minus size of IP-in-IP header.



**Note:** This will ensure that the traffic sent to the CRS will be within the MTU constraints of CRS' network-interface.

## Network Interface Configuration

This section explains how to access and configure network interface.

### Special Considerations

Configuration changes to network interface parameters might have an impact on boot configuration parameters. After configuring the network interface, you might receive a message indicating that you could be changing boot config parameters under the following circumstances:

- A physical interface or network interface element matches the boot interface (for example, the physical port is the same as the boot port).
- The boot configuration parameters are modified, because the IPv4 address, netmask, or gateway is different from the corresponding boot configuration parameters.

You are asked if you want to continue. If you enter yes, the configuration will be saved and then the differing boot configuration parameters will be changed. If you enter no, then the configuration is not saved and the boot configuration parameters are not changed.

Configuring the physical and network interface elements for the first management interface is optional because that interface, eth0, is implicitly created by a valid bootparam configuration that specifies the boot device, IPv4 address, subnet, and gateway.

## Network Interfaces Configuration

This section describes how to configure a network interface.

Access the **network-interface** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# system
ACMEPACKET(system)# network-interface
ACMEPACKET(network-interface)
```

## IP Configuration and Identification

You must specify the identity and address for all network interfaces.

Set the following parameters to configure a network interface:

1. **name**—Set the name for the network interface. This must be the same name as the physical interface to which it corresponds.
2. **description**—Enter a description of the network for easier identification.
3. **hostname**—Set the hostname (FQDN) of this network interface. This parameter is optional.
4. **ip-address**—Set the IP address of this network interface.
5. **netmask**—Set the netmask of this network interface in dotted decimal notation.
6. **gateway**—Set the gateway that this network interface uses to communicate with the next hop.
7. **sec-gateway**—Set an additional optional gateway for this network interface
8. **dns-ip-primary**—Set the DNS servers. You can set an additional two DNS servers by using the **dns-ip-backup1** and **dns-ip-backup2** parameters.
9. **dns-domain**—Set the default domain name.
10. **signaling-mtu**—Sets the MTU size for IPv4 or IPv6 transmission.

## VLAN Configuration

One parameter is required to configure VLANs on a Oracle Communications Session Border Controller. The **sub-port-id** parameter located in the **network-interfaces** element adds and masks for a specific VLAN tag.

**sub-port-id**—Enter the identification of a specific virtual interface in a physical interface (e.g., a VLAN tag). If this network interface is not channelized, leave this field blank, and the value will correctly default to 0. The **sub-port-id** is only required if the operation type is Media. The valid range is:

- Minimum—0
- Maximum—4095.

## HIP Address Configuration

To configure administrative service functionality on a media interface, you must first define all source IP addresses in the media-interface's network that will exchange administrative traffic with the system. Next you will identify the type of administrative traffic each of those addresses will exchange.

## System Configuration

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You must configure the **gateway** parameter on this **network-interface** for administrative traffic to successfully be forwarded. You should also ensure that this network interface is not on an overlapping network as any of the administrative networks (wancoms).

Set the following parameters to configure HIP functionality on a network interface:

1. **add-hip-ip**—Configure all possible IP address(es) from which the Oracle Communications Session Border Controller will accept administrative traffic. Entries in this element are IP addresses of media network interfaces. This parameter can accept multiple IP addresses. You can later remove this entry by typing **remove-hip-ip** followed by the appropriate IP address.
2. **add-ftp-ip**—Set the IP address(es) that will access the Oracle Communications Session Border Controller's FTP server. This allows standard FTP packets enter the Oracle Communications Session Border Controller and reach the host. You can later remove this entry by typing **remove-ftp-ip** followed by the appropriate IP address.
3. **add-icmp-ip**—Set the IP address(es) that can ping the system and expect replies. This parameter can accommodate multiple ping IP addresses. You can later remove this entry by typing **remove-icmp-ip** followed by the appropriate IP address.

For security, if the ICMP address and the hip-ip-list are not added for an address, the Oracle Communications Session Border Controller hardware discards ICMP requests or responses for the address.

4. **add-snmp-ip**—Set the IP address(es) that will access the system's SNMP process. This lets SNMP traffic enter the Oracle Communications Session Border Controller and reach the host. You can later remove this entry by typing **remove-snmp-ip** followed by the appropriate IP address.
5. **add-telnet-ip**—Set the IP address(es) that can connect and access the system through Telnet. You can later remove this entry by typing **remove-telnet-ip** followed by the appropriate IP address.
6. **add-ssh-ip**—Set the IP address(es) that can connect and access the system through SSH. You can later remove this entry by typing **remove-SSH-ip** followed by the appropriate IP address.

### Configurable MTU Size

Configurable MTU on per network-interface basis enables the user to set a different MTU on each network interface. It also enables the user to set a system wide default MTU for IPv6 and IPv4 network interfaces. System wide defaults can be set in **system-config** configuration object by setting **ipv6-signaling-mtu** or **ipv4-signaling-mtu**. Defaults are 1500 for both IPv6 and IPv4.

These settings can be overwritten for each network interface by setting **signaling-mtu** in **network-interface** configuration object. Default is 0 – meaning use the system wide MTU.

This feature applies to all Signaling packets generated by the Oracle Communications Session Border Controller. All UDP packets greater than the MTU will be fragmented. For all TCP connections we advertise MSS (Maximum Segment Size) TCP option in accordance with the configured MTU. MSS option is sent in SYN and SYN/ACK packets to let the other side of the TCP connection know what your maximum segment size is. This ensures that no TCP packet is greater than the configured MTU.

1. MTU settings do not apply to media packets.
2. UDP: MTU settings apply only to packets sent by the Oracle Communications Session Border Controller. The Oracle Communications Session Border Controller will continue to process received packets even if they exceed to the configured MTU.
3. Security Phy (IPsec) hardware only; We subtract 100 bytes from the configured MTU to allow for extra headers added by security protocols. This happens even when Security Phy (IPsec) is in clear mode (no security is being applied). Due to hardware limitations of the Security Phy (IPsec) it only allows one MTU per physical port. The maximum MTU of all network interfaces on a given physical port will be used as the MTU for that physical port.
4. The Call Recording feature is where we make a copy of a packet, encapsulate it in an IP-in-IP header and send it to a configured Call Recording Server (CRS). When Call Recording is enabled, to allow space for IP-in-IP encapsulation we reduce the MTU of the original packets to be to be the lesser of the two options listed below.
  - Original Destination network MTU minus size of IP-in-IP header.
  - CRS network interface's MTU minus size of IP-in-IP header.



**Note:** This will ensure that the traffic sent to the CRS will be within the MTU constraints of CRS' network-interface.

## System Wide MTU Size

To change system wide MTU settings:

1. Access the **system-config** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# system
ACMEPACKET(system)# system-config
ACMEPACKET(system-config)#
```

2. Type **select** to begin editing the **system-config** object.

```
ACMEPACKET(system-config)# select
ACMEPACKET(system-config)#
```

3. **ipv6-signaling-mtu** or **ipv4-signaling-mtu** Configure MTU in the system config and optionally in the network interface. Default will be 1500 bytes.

```
ACMEPACKET(system-config)# ipv6-signaling-mtu 1500
ACMEPACKET(system-config)# ipv4-signaling-mtu 1600
```

4. Type **done** to save your configuration.

## IP Identification (ID) Field

By default, non-fragmented UDP packets generated by media interfaces have the ID field set to 0. You can configure the Oracle Communications Session Border Controller to populate this field with an incrementing value by adding the **increment-ip-id** option in the media manager. Every non-fragmented packet sent will have its ID increased by one from the previous packet sent.

Using a packet trace application, egress packets from the Oracle Communications Session Border Controller will have an ID field that appears to be incrementing. Enabling the ID field can help distinguish a retransmitted non-fragmented application layer packet from a packet retransmitted by the network layer in monitoring or lab situations.

## IP Identification Field Configuration

To enable ID field generation in media-manager:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. **options**—Set the options parameter by typing **options**, a Space, the option name **increment-ip-id** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(media-manager)# options + increment-ip-id
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the realm configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

4. Save and activate your configuration.

# SNMP

---

This section explains how to configure Simple Network Management Protocol (SNMP), trap receivers, and syslog servers. These features are not essential for baseline Oracle Communications Session Border Controller service, but they are necessary to use network management systems to manage the Oracle Communications Session Border Controller. They provide important monitoring and system health information that contribute to a robust deployment of the system.

## Overview

SNMP is used to support monitoring of network-attached devices for conditions that warrant administrative attention. SNMP is comprised of three groups of settings on a Oracle Communications Session Border Controller. These settings are system-wide configurations including MIB contact information, SNMP community settings, and trap receivers.

### Basic SNMP Parameters

The Oracle Communications Session Border Controller includes several parameters that control basic SNMP functionality. The MIB-related elements are for informational purposes, and are helpful if set. The remainder of the parameters determines if certain Oracle Communications Session Border Controller events are reported to the SNMP system.

### SNMP Community

An SNMP community is a grouping of network devices and management stations used to define where information is sent and accepted. An SNMP device or agent might belong to more than one SNMP community. SNMP communities provide a type of password protection for viewing and setting management information within a community.

SNMP communities also include access level settings. They are used to define the access rights associated with a specific SNMP community. The Oracle Communications Session Border Controller lets you define two types of access levels: read-only and read-write. You can define multiple SNMP communities on a Oracle Communications Session Border Controller to segregate access modes per community and NMS host.

### SNMP Trap Receiver Uses

A trap receiver is an application used to receive, log, and view SNMP traps for monitoring the Oracle Communications Session Border Controller (SBC). An SNMP trap is the notification sent from a network device, such as the SBC, that declares a change in service. Multiple trap receivers can be defined on an SBC for either redundancy or to segregate alarms with different severity levels to individual trap receivers.

Each Oracle Communications Session Delivery Manager which manages Oracle Communications Session Border Controllers should be configured on those SBCs as trap receivers.

## Configuring SNMP

---

This section describes how to configure your Oracle Communications Session Border Controller to work with external SNMP systems. Sample configurations are also provided.

### SNMP Configuration Overview

1. Configure the SNMP identification information. This step includes configuring the MIB system contact, name, and location parameters.
2. Set the general SNMP parameters to enable or disable SNMP on the Oracle Communications Session Border Controller. This step includes setting the switches that govern how the SNMP system responds to specified events.
3. Set the syslog events. This step includes setting the parameters for handling SNMP monitoring syslog events, which can trigger SNMP syslog traps.
4. Set SNMP communities. Configuration is separated into a unique configuration element.



5. Set trap receivers. Configuration is separated into a unique configuration element.

## SNMP Configuration

To configure SNMP:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # system
```

3. Type **system-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (system) # system-config
ACMEPACKET (system-config) #
```

From this point, you can set SNMP parameters. The following is an example what an SNMP configuration might look like. Parameters not described in this section are omitted below.

```
system-config
  mib-system-contact      John Doe
  mib-system-name         Test System
  mib-system-location     Upstairs
  snmp-enabled            enabled
  enable-snmp-auth-traps  disabled
  enable-snmp-syslog-notify disabled
  enable-snmp-monitor-traps disabled
  enable-env-monitor-traps disabled
  snmp-syslog-his-table-length 1
  snmp-syslog-level       WARNING
```

### System Wide Configuration for SNMP

This section describes the system-wide SNMP parameters found in the System Configuration element. These parameters set global SNMP information.

Set the following parameters to configure system wide SNMP functionality:

1. **mib-system-contact**—Set the contact information used within the system's MIB transactions. The SNMP agent sends this information to an NMS in response to an SNMP Get for the MIB-II sysContact MIB variable. This parameter's value can be a textual identification of your company's contact person for the system and information about how to contact that person.
2. **mib-system-name**—Set the identification of this Oracle Communications Session Border Controller presented within MIB transactions. This value, along with the target name of the system (identified in the boot parameters) are the values reported for MIB-II when an SNMP GET is issued by the NMS for the MIB-II sysName variable. This parameter has no direct relation to the hostname parameter in the system configuration element.

By convention, this is the node's FQDN. For SNMP MIB-II sysName GETs, the system returns SNMP communications in the following format: <targetName>[.<mib-system-name>]

targetName is the value configured in the target name (tn) boot parameter and mib-system-name is the value configured in this field.

3. **mib-system-location**—Set the physical location of this Oracle Communications Session Border Controller that is reported within MIB transactions. This parameter is reported when an SNMP GET is issued by the NMS for the MIB-II sysLocation variable. This parameter has no direct relation to the location field in the system configuration element.
4. **snmp-enabled**—Set the SNMP system on this Oracle Communications Session Border Controller to **enabled** or **disabled**. By default, this parameter is set to **enabled**. The valid values are:
  - enabled | disabled

## System Configuration

---

5. **enable-snmp-syslog-notify**—Set whether SNMP traps are sent when the system generates an alarm message. The SNMP agent sends a trap when an alarm is generated if the following conditions are met:
  - SNMP is enabled.
  - This field is enabled.
  - The syslog severity level is equal to or greater than the severity level configured in the SNMP Syslog Level field.The default is **disabled**. Valid values are:
  - **enabled | disabled**
6. **enable-snmp-monitor-traps**—When this parameter is enabled, the Oracle Communications Session Border Controller generates traps with unique trap-IDs for each syslog event. If this parameter is disabled, a single trap-ID is used for all events, with different values in the description string. The default is **disabled**. The valid values are:
  - enabled | disabled
7. **enable-snmp-auth-traps**—Set whether the SNMP authentication traps are enabled. If an SNMP request fails authentication because of an IPv4 address and SNMP community mismatch, the SNMP request will be rejected. This field determines if an SNMP trap will be sent in response to the authentication failure. The default is **disabled**. Valid values for this parameter are:
  - **enabled | disabled**
8. **enable-env-monitor-traps**—Set whether or not the SNMP environment monitor traps are enabled. Environment traps include main board PROM temperature, CPU voltage, power supplies, fan speeds, etc. The default is **disabled**. Valid values for this parameter are:
  - enabled | disabled
9. **snmp-syslog-his-table-length**—Set the length of the syslog trap history table. When a syslog message that meets the SNMP syslog level field criteria is generated and SNMP is enabled, the SNMP agent adds that message to a history table. This parameter indicates the number of entries the table can contain. The default is **1**. The valid range is:
  - Minimum—1
  - Maximum—500Once the last table entry is filled, the oldest entry will be overwritten with a new entry.
10. **snmp-syslog-level**—Set the log severity level threshold that will cause the syslog trap to be sent to an NMS. When this criteria is met and the appropriate SNMP trap is sent, an entry is written to the SNMP Syslog History Table. The default is **warning**. The following are valid values:
  - emergency | critical | major | minor | warning | notice | info | trace | debug | detail

## SNMP Community Configuration

To configure SNMP communities:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # system
```

3. Type **snmp-community** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (system) # snmp-community  
ACMEPACKET (snmp-community) #
```

From this point, you can set SNMP community parameters.

The following is an example what an SNMP Community configuration might look like. Parameters not described in this section are omitted below.

```
snmp-community
  community-name      public
  access-mode         READ-ONLY
  ip-addresses        10.0.1.42
```

4. **community-name**—Set the SNMP community name of an active community where this Oracle Communications Session Border Controller can send or receive SNMP information. A community name value can also be used as a password to provide authentication, thereby limiting the NMSs that have access to this system. With this field, the SNMP agent provides trivial authentication based on the community name that is exchanged in plain text SNMP messages.
5. **access-mode**—Set the access level for all NMSs defined within this SNMP community. The access level determines the permissions that other NMS hosts can wield over this Oracle Communications Session Border Controller. The default is **read-only**. The valid values are:
  - **read-only**—allows GET requests.
  - **read-write**—allows both GET and SET requests.
6. **ip-addresses**—Set one or multiple IPv4 addresses that are valid within this SNMP community. These IPv4 addresses correspond with the IPv4 address of NMS applications that monitor or configure this Oracle Communications Session Border Controller. Include the IPv4 addresses of all servers where Element Management System is installed.

### Trap Receiver Configuration

To configure trap receivers:

1. Access the **trap-receiver** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# system
ACMEPACKET(system)# trap-receiver
ACMEPACKET(trap-receiver)#
```

2. Select the **trap-receiver** object to edit.

```
ACMEPACKET(trap-receiver)# select
<ip-address>:

ACMEPACKET(trap-receiver)#
```

3. **ip-address** — Set the IPv4 or IPv6 address of an authorized NMS. This parameter is the IPv4 or IPv6 address of an NMS where traps are sent. If you do not specify a port number, the default SNMP trap port of **162** will be used.
4. **filter-level** — Set the filter level threshold that indicates the severity level at which a trap is to be sent to this particular trap receiver. The default for this parameter is **critical**.

Example: When a trap with a severity level of **critical** is generated, the SNMP agent will send this trap only to NMSs that are configured in a trap-receiver element and have a **filter-level** value of **critical**.

The following table maps Syslog and SNMP alarms to trap receiver filter levels.

Filter Level	Syslog Severity Level	(SNMP) Alarm Severity Level
Critical	Emergency (1)	Emergency
	Critical (2)	Critical
Major	Emergency (1)	Emergency
	Critical (2)	Critical
	Major (3)	Major
Minor	Emergency (1)	Emergency
	Critical (2)	Critical

## System Configuration

Filter Level	Syslog Severity Level	(SNMP) Alarm Severity Level
	Major (3) Minor (4)	Major Minor
All	Emergency (1) Critical (2) Major (3) Minor (4) Warning (5) Notice (6) Info (7) Trace (8) Debug (9)	Emergency Critical Major Minor Warning

When configuring the trap-receiver element for use with Oracle Communications Session Element Manager systems, Oracle recommends that the value of **filter-level** be set to **All** for that configuration element that includes those servers.

- community-name** — Set the community name to which this trap receiver belongs. This community must be defined in the SNMP community element.
- Type **done** to save your configuration.

## Media Supervision Traps

The Oracle Communications Session Border Controller, when functioning as a border gateway, will send the following trap when the media supervision timer has expired. This behavior is disabled by default, but can be enabled by changing the **media-supervision-traps** parameter to **enabled** in the media-manager configuration element.

```
apSysMgmtMediaSupervisionTimerExpTrap    NOTIFICATION-TYPE
OBJECTS                                  { apSysMgmtCallId }
STATUS                                    current
DESCRIPTION
    " The trap will be generated when a media supervision timer
    has expired. This behavior is disabled by default but may
    be enabled by changing the 'media-supervision-traps'
    parameter of the 'media-manager' configuration element. The
    included object is the call identifier for the call which had
    the timer expire."
 ::= { apSystemManagementMonitors 34 }
```

The system does not send this trap when functioning as an integrated Oracle Communications Session Border Controller.

## Syslog and Process Logs

Logging events is a critical part of diagnosing misconfigurations and optimizing operations. Oracle Communications Session Border Controllers can send both syslog and process log data to appropriate hosts for storage and analysis.

### Overview

The Oracle Communications Session Border Controller generates two types of logs, syslogs and process logs. Syslogs conform to the standard used for logging servers and processes as defined in RFC 3164.

Process logs are Oracle proprietary logs. Process logs are generated on a per-task basis and are used mainly for debugging purposes. Because process logs are more data inclusive than syslog, their contents usually encompass syslog log data.

Syslog and process log servers are both identified by an IPv4 address and port pair.

### Process Log Messages

Process log messages are sent as UDP packets in the following format:

```
<file-name>:<log-message>
```

In this format, <filename> indicates the log filename and <log-message> indicates the full text of the log message as it would appear if it were written to the normal log file.

## Syslog and Process Logs Configuration

This section describes how to configure syslog and process log servers.

To configure syslogs and process logs:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # system
```

3. Type **system-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (system) # system-config
```

```
ACMEPACKET (system-config) #
```

From this point, you can set process log parameters. Skip to the following process log configuration section.

4. Type **syslog-server** and press Enter. The system prompt changes to let you know that you can begin configuring individual syslog parameters

```
ACMEPACKET(system-config)# syslog-server
```

```
ACMEPACKET(syslog-server)#
```

From this point, you can set syslog parameters. The following is an example what an syslog and process log configuration might look like. Parameters not described in this section are omitted below.

```
system-log-level          WARNING
syslog-server
  address                 172.15.44.12
  port                    514
  facility                 4
process-log-level        NOTICE
process-log-ip-address    0.0.0.0
process-log-port          0
```

### Syslog Configuration

The Oracle Communications Session Border Controller supports multiple syslog servers. As the number of active syslog increases, the performance level of the Oracle Communications Session Border Controller may decrease. Therefore, we recommend configuring no more than 8 syslog servers.

Set the following parameters to configure syslog servers:

1. **address**—Set the IPv4 address of a syslog server.
2. **port**—Set the port portion of the syslog server. The default is **514**.

## System Configuration

---

- 3. facility**—Set an integer to identify a user-defined facility value sent in every syslog message from the Oracle Communications Session Border Controller to the syslog server. This parameter is used only for identifying the source of this syslog message as coming from the Oracle Communications Session Border Controller. It is not identifying an OS daemon or process. The default value for this parameter is **4**. RFC 3164 specifies valid facility values.

In software release versions prior to Release 1.2, the Oracle Communications Session Border Controller would send all syslog messages with a facility marker of 4.

- 4. system-log-level**—Set which log severity levels write to the system log (filename: acmelog). The default is **WARNING**. Valid values are:
  - EMERGENCY | CRITICAL | MAJOR | MINOR | WARNING | NOTICE | INFO | TRACE | DEBUG | DETAIL

## Process Log Configuration

Set the following parameters to configure the process log server:

- 1. process-log-level**—Set the starting log level all processes running on the system use. Each individual process running on the system has its own process log. The default is **NOTICE**. Valid values are:
  - EMERGENCY | CRITICAL | MAJOR | MINOR | WARNING | NOTICE | INFO | TRACE | DEBUG | DETAIL
- 2. process-log-ip-address**—Set the IPv4 address of the process log server. The default **0.0.0.0**, which causes log messages to be written to the normal log file.
- 3. process-log-port**—Set the port number associated with the process log server. The default value for this parameter is **0**, which causes log messages to be written to the normal log file. The valid range is:
  - Minimum—0
  - Maximum—65535.

## Host Routes


---

Host routes let you insert entries into the Oracle Communications Session Border Controller's routing table. These routes affect traffic that originates at the Oracle Communications Session Border Controller's host process. Host routes are used primarily for steering management traffic to the correct network.

When traffic is destined for a network that is not explicitly defined on a Oracle Communications Session Border Controller, the default gateway (located in the **system-config**) is used. If you try to route traffic to a specific destination that is not accessible through the default gateway, you need to add a host route. Host routes can be thought of as a default gateway override.

Certain SIP configurations require that the default gateway is located on a media interface. In this scenario, if management applications are located on a network connected to an administrative network, you will need to add a host route for management connectivity.

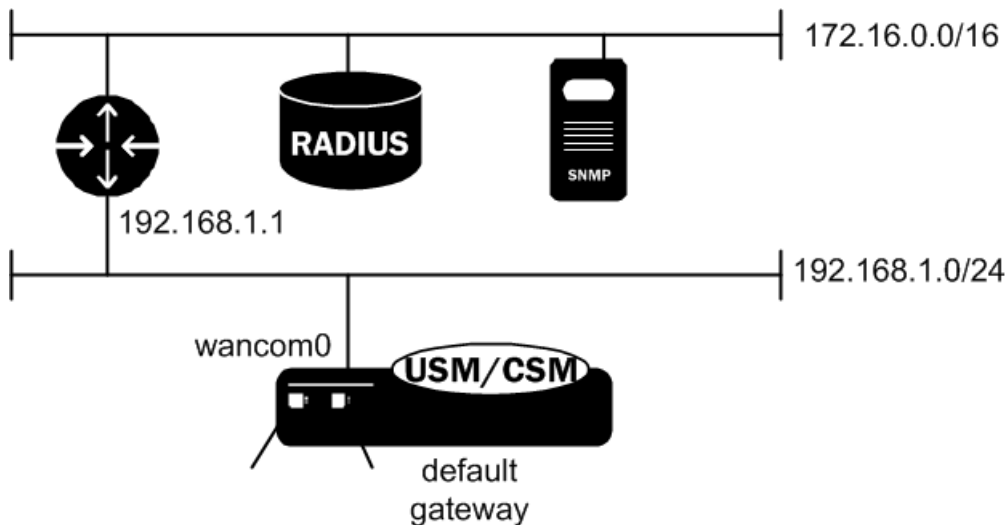
When source-based routing is used, the default gateway must exist on a media interface. Host routes might be needed to reach management applications connected to a management port in this kind of situation as well.

 **Note:** Do not configure a **host-route > gateway** with an address already used for any existing **network-interface > gateway**.

## Host Routes Example

Because SIP signaling over media interfaces is enabled, the default gateway uses an IPv4 address assigned to a media interface. Maintenance services (SNMP and Radius) are located on a network connected to, but separate from, the 192.168.1.0/24 network on wancom0. In order to route Radius or SNMP traffic to an NMS (labeled as SNMP in the following example), a host route entry must be a part of the Oracle Communications Session Border Controller

configuration. The host route tells the host how to reach the 172.16.0.0/16 network. The actual configuration is shown in the example in the next section of this guide.



## Host Route Configuration

To configure a host route:

1. Access the **host-route** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# system
ACMEPACKET(system)# host-route
ACMEPACKET(host-route)#
```

2. **dest-network**—Set the IP address of the destination network that this host route points toward.
3. **netmask**—Set the netmask portion of the destination network for the route you are creating. The netmask is in dotted decimal notation.
4. **gateway**—Set the gateway that traffic destined for the address defined in the first two elements should use as its first hop.
5. Type **done** to save your configuration.

## Setting Holidays in Local Policy

This section explains how to configure holidays on the Oracle Communications Session Border Controller.

You can define holidays that the Oracle Communications Session Border Controller recognizes. Holidays are used to identify a class of days on which a local policy is enacted. All configured holidays are referenced in the **local-policy-attributes** configuration subelement as an H in the **days-of-week** parameter. Because holidays are entered on a one-time basis per year, you must configure a new set of holidays yearly.

## Holidays Configuration

To configure holidays:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-router-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

## System Configuration

```
ACMEPACKET(session-router) # session-router-config
ACMEPACKET(session-router-config) #
```

4. Type **holidays** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router-config) # holidays
ACMEPACKET(session-router-holidays) #
```

From this point, you can configure the holidays subelement. To view all holidays parameters, enter a ? at the system prompt.

```
holiday
      date           2005-01-01
      description    New Years Day
```

To configure a holiday, add an entry for the following parameters in the holidays element:

5. **date**—Enter the holiday's date in YYYY-MM-DD format.
6. **description**—Enter a short description for the holiday you are configuring. If the description contains words separated by spaces, enter the full description surrounded by quotation marks.

## Enhanced Control of UDP and TCP Ports

This section explains how to configure the Oracle Communications Session Border Controller for finer control of the set of UDP and TCP ports that on which the Oracle Communications Session Border Controller provides services. The settings you can configure have an impact on:

- UDP/TCP port 111 (the RPC services port), which is disabled on Oracle Communications Session Border Controller startup but can be enabled in the boot parameters
- TCP ports 3000 (used when notify commands are issued remotely, i.e. via an element management system) and 3001 (used for remote configuration, i.e. via an element management system), which can now be enabled or disabled in the system configuration

Neither configuration for these features is covered by RTC, so you must reboot your Oracle Communications Session Border Controller for changes to take effect. Be aware that rebooting can cause system downtime, and plan accordingly.

### Port 111 Configuration

To enable port 111 using Oracle Communications Session Border Controller boot parameters:

1. In Superuser mode, type **configure terminal** and press Enter

```
ACMEPACKET# configure terminal
```

2. To enter the boot parameters so that you can configure them, type **bootparam** and press Enter.

```
ACMEPACKET(configure) # bootparam
```

3. Press Enter to scroll through the list of boot parameters until you reach the setting for flags.

To set this value correctly, you need to add the value 0x200000 to your existing flag setting in the boot parameters. In the example below, the existing flag value is 0x30008. When the value 0x200000 is added, the result is 0x230008. The result is the value that you need to set.

When you reach the flag setting, type the value representing the flags you need (0x230008 in the example below) and press Enter. Continue to press Enter to finish scrolling through the rest of the boot parameters.

```
'.' = clear field; '-' = go to previous field; ^D = quit
boot device           : wancom0
processor number      : 0
host name             : acmepacket8
file name             : /tffs0/sd220p9.gz
inet on ethernet (e)  : 10.0.1.57:ffff0000
inet on backplane (b) : 0.0.0.0
```



```

host inet (h)           : 10.0.1.5
gateway inet (g)       : 10.0.0.1
user (u)               : user
ftp password (pw)      : password
flags (f)              : 0x30008 0x230008
target name (tn)      : acmesystem
startup script (s)     : 0
other (o)              :

```

NOTE: These changed parameters will not go into effect until reboot. Also, be aware that some boot parameters may also be changed through the PHY and Network Interface Configurations.

```
ACMEPACKET(configure)#
```

4. Type **exit** to return to the main Superuser menu so that you can reboot your Oracle Communications Session Border Controller and apply the settings you have entered.

```
ACMEPACKET(configure)# exit
```

5. Reboot your Oracle Communications Session Border Controller. Type a **y** and press Enter to reboot.

```
ACMEPACKET# reboot
```

```
-----
WARNING: you are about to reboot this SD!
-----
```

```
Reboot this SD [y/n]?:y
```

## Port 3000 and 3001 Configuration

To control TCP ports 3000 and 3001 in the system configuration:

1. Access the **security-config** configuration element.

```

ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# security-config
ACMEPACKET(security-config)#

```

2. Type **select** to begin editing the **system-config** object.

```

ACMEPACKET(system-config)# select
ACMEPACKET(system-config)#

```

3. The parameter controlling ports 3000 and 3001 is called **remote-control**, and its default is enabled. To disable the ports, set this parameter to disabled.

```
ACMEPACKET(system-config)# remote-control disabled
```

4. Type **done** to save your configuration.

5. Reboot your Oracle Communications Session Border Controller. Type a **y** and press Enter to reboot.

```
ACMEPACKET# reboot
```

```
-----
WARNING: you are about to reboot this SD!
-----
```

```
Reboot this SD [y/n]?:y
```

## DNS Transaction Timeout

This section explains how to configure the DNS transaction timeout interval on a per network-interface basis. You can currently configure the Oracle Communications Session Border Controller with a primary and two optional backup DNS servers. The Oracle Communications Session Border Controller queries the primary DNS server and upon not receiving a response within the configured number of seconds, queries the backup1 DNS server and if that times out as well, then contacts the backup2 DNS server.

### Retransmission Logic

The retransmission of DNS queries is controlled by three timers. These timers are derived from the configured DNS timeout value and from underlying logic that the minimum allowed retransmission interval should be 250 milliseconds; and that the Oracle Communications Session Border Controller should retransmit 3 times before timing out to give the server a chance to respond.

- **Init-timer** is the initial retransmission interval. If a response to a query is not received within this interval, the query is retransmitted. To safeguard from performance degradation, the minimum value allowed for this timer is 250 milliseconds.
- **Max-timer** is the maximum retransmission interval. The interval is doubled after every retransmission. If the resulting retransmission interval is greater than the value of max-timer, it is set to the max-timer value.
- **Expire-timer**: is the query expiration timer. If a response is not received for a query and its retransmissions within this interval, the server will be considered non-responsive and the next server in the list will be tried.

The following examples show different timeout values and the corresponding timers derived from them.

```
timeout >= 3 seconds
Init-timer = Timeout/11
Max-Timer = 4 * Init-timer
Expire-Timer = Timeout
timeout = 1 second
Init-Timer = 250 ms
Max-Timer = 250 ms
Expire-Timer = 1 sec
timeout = 2 seconds
Init-Timer = 250 ms
Max-Timer = 650 ms
Expire-Timer = 2sec
```

### DNS Transaction Timeout Configuration

To configure DNS transaction timeout:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# system
```

3. Type **network-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(system)# network-interface
ACMEPACKET(network-interface)#
```

From this point, you can configure network interface parameters. To view all network interface parameters, enter a **?** at the system prompt.

4. **dns-timeout**—Enter the total time in seconds you want to elapse before a query (and its retransmissions) sent to a DNS server would timeout. The default is **11** seconds. The valid range is:
  - Minimum—1
  - Maximum—999999999.

If a query sent to the primary DNS server times out, the backup1 DNS server is queried. If the query times out after the same period of time elapses, the query continues on to the backup2 DNS server.

5. Save and activate your configuration.

## DNS Server Status via SNMP

The Oracle Communications Session Border Controller monitors the status of all configured DNS servers used by a SIP daemon. If a DNS server goes down, a major alarm is sent. If all DNS servers used by a SIP daemon are down, a critical alarm is sent. The `apAppsDnsServerStatusChangeTrap` is sent for both events.

You can poll the status of a DNS server using the `apAppsDNSServerStatusTable` in the `ap-apps.mib`.

Once the `apAppsDnsServerStatusChangeTrap` has been sent, a 30 second window elapses until the server status is checked again. At the 30 second timer expiration, if the server is still down, another trap and alarm are sent. If the server has been restored to service, the `apAppsDnsServerStatusChangeClearTrap` is sent.

## Persistent Protocol Tracing

This section explains how to configure persistent protocol tracing to capture specific SIP protocol message logs and persistently send them off the Oracle Communications Session Border Controller, even after rebooting the system. This feature is not applicable to log for H.323 or IWF.

### About Persistent Protocol Tracing

You can configure sending protocol message logs off of the Oracle Communications Session Border Controller, and have that persist after a reboot. You no longer have to manually issue the `notify` command each time you reboot.

To support persistent protocol tracing, you configure the following system-config parameters:

- **call-trace**—Enable/disable protocol message tracing (currently only `sipmsg.log` and `alg.log`) regardless of the `process-log-level` setting. If the `process-log-level` is set to `trace` or `debug`, `call-trace` will not disable.
- **internal-trace**—Enable/disable internal ACP message tracing for all processes, regardless of `process-log-level` setting. This applies to all `*.log` (internal ACP message exchange) files other than `sipmsg.log` and `alg.log`. If the `process-log-level` is set to `trace` or `debug`, `call-trace` will not disable.
- **log-filter**—Determine what combination of protocol traces and logs are sent to the log server defined by the `process-log-ip` parameter value. You can also fork the traces and logs, meaning that you keep trace and log information in local storage as well as sending it to the server. You can set this parameter to any of the following values: `none`, `traces`, `traces-fork`, `logs`, `logs`, `all`, or `all-fork`.

The Oracle Communications Session Border Controller uses the value of this parameter in conjunction with the `process-log-ip` and `process-log-port` values to determine what information to send. If you have configured the `proc-log-ip` and `proc-log-port` parameters, choosing `traces` sends just the trace information (provided they are turned on), `logs` sends only process logs (`log.*`), and `all` sends everything (which is the default).

### About the Logs

When you configure persistent protocol tracing, you affect the following types of logs.

 **Note:** Enabling logs can have an impact on Oracle Communications Session Border Controller performance.

#### Process Logs

Events are logged to a process log flow from tasks and are specific to a single process running on the Oracle Communications Session Border Controller. By default they are placed into individual files associated with each process with the following name format:

```
log.<taskname>
```

By setting the new `log-filter` parameter, you can have the logs sent to a remote log server (if configured). If you set `log-filter` to `logs` or `all`, the logs are sent to the log server. Otherwise, the logs are still captured at the level the `process-log-level` parameter is set to, but the results are stored on the Oracle Communications Session Border Controller's local storage.

## System Configuration

---

### Communication Logs

These are the communication logs between processes and system management. The logs are usually named <name>.log, with <name> being the process name. For example, sipd.log.

This class of log is configured by the new internal-trace parameter.

### Protocol Trace Logs

The only protocol trace logs included at this time are sipmsg.log for SIP. The H.323 system tracing is not included. All of the logs enabled with the call-trace parameter are sent to remote log servers, if you also set the log-filter parameter to logs or all.

## Persistent Protocol Tracing Configuration

Before you configure persistent protocol tracing, ensure you have configured the process logs by setting the system configuration's **process-log-ip** parameter.

To configure persistent protocol tracing:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # system
```

3. Type **system-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (system) # system-config  
ACMEPACKET (system-config) #
```

4. **call-trace**—Set to **enabled** to enable protocol message tracing for sipmsg.log for SIP. The default is **disabled**. The valid values are:

- enabled | disabled

5. **internal-trace**—Set to **enabled** to enable internal ACP message tracing for all processes. The default is **disabled**. The valid values are:

- enabled | disabled

6. **log-filter**—Choose the appropriate setting for how you want to send and/or store trace information and process logs. The valid values are:

- **none**—No information will be sent or stored.
- **traces**—Sends the trace information to both the log server; includes <name>.log files that contain information about the Oracle Communications Session Border Controller's internal communication processes (<name> is the name of the internal process)
- **traces-fork**—Sends the trace information to both the log server and also keeps it in local storage; includes <name>.log files that contain information about the Oracle Communications Session Border Controller's internal communication processes (<name> is the name of the internal process)
- **logs**—Sends the process logs to both the log server; includes log.\* files, which are Oracle Communications Session Border Controller process logs
- **logs-fork**—Sends the process logs to both the log server and also keeps it in local storage; includes log.\* files, which are Oracle Communications Session Border Controller process logs
- **all**—Sends all logs to the log servers that you configure
- **all-fork**—Sends all logs to the log servers that you configure, and it also keeps the logs in local storage

7. Save and activate your configuration.

## System Access Control

You can configure a system access control list (ACL) for your Oracle Communications Session Border Controller that determines what traffic the Oracle Communications Session Border Controller allows over its management interface (wancom0). By specifying who has access to the Oracle Communications Session Border Controller via the management interface, you can provide DoS protection for this interface.

Using a list of IP addresses and subnets that are allowable as packet sources, you can configure what traffic the Oracle Communications Session Border Controller accepts and what it denies. All IP packets arriving on the management interface are subject; if it does not match your configuration for system ACL, then the Oracle Communications Session Border Controller drops it.

 **Note:** All IP addresses configured in the SNMP community table are automatically permitted.

### Adding an ACL for the Management Interface

The new subconfiguration **system-access-list** is now part of the system configuration, and its model is similar to host routes. For each entry, you must define an IP destination address and mask; you can specify either the individual host or a unique subnet.

If you do not configure this list, then there will be no ACL/DoS protection for the Oracle Communications Session Border Controller's management interface.

You access the **system-access-list** via system path, where you set an IP address and netmask. You can configure multiple system ACLs using this configuration.

To add an ACL for the management interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure) # system
ACMEPACKET (system) #
```

3. Type **system-access-list** and press Enter.

```
ACMEPACKET (system) # system-access-list
ACMEPACKET (system-access-list) #
```

4. **source-address**—Enter the IP address representing for the source network for which you want to allow traffic over the management interface.

5. **netmask**—Enter the netmask portion of the source network for the traffic you want to allow. The netmask is in dotted decimal notation.

### Notes on Deleting System ACLs

If you delete a system ACL from your configuration, the Oracle Communications Session Border Controller checks whether or not there are any active FTP or Telnet client was granted access when the entry was being removed. If such a client were active during ACL removal, the Oracle Communications Session Border Controller would warn you about the condition and ask you to confirm the deletion. If you confirm the deletion, then the Oracle Communications Session Border Controller's session with the active client is suspended.

The following example shows you how the warning message and confirmation appear. For this example, an ACLI has been deleted, and the user is activating the configuration that reflects the change.

```
ACMEPACKET # activate-config
Object deleted will cause service disruption:
system-access-list: identifier=172.30.0.24
** WARNING: Removal of this system-ACL entry will result
in the lockout of a current FTP client
```

## System Configuration

```
Changes could affect service, continue (y/n) y
Activate-Config received, processing.
```

## System TCP Keepalive Settings

You can configure the Oracle Communications Session Border Controller to control TCP connections by setting:

- The amount of time the TCP connection is idle before the Oracle Communications Session Border Controller starts sending keepalive messages to the remote peer
- The number of keepalive packets the Oracle Communications Session Border Controller sends before terminating the TCP connection

If TCP keepalive fails, then the Oracle Communications Session Border Controller will drop the call associated with that TCP connection.

In the ALCI, a configured set of network parameters appears as follows:


```
network-parameters
tcp-keepinit-timer          75
tcp-keepalive-count        4
tcp-keepalive-idle-timer   400
tcp-keepalive-interval-timer 75
tcp-keepalive-mode         0
```

Then you apply these on a per-interface basis. For example, the H.323 interface (stack) configuration allows you to enable or disabled use of the network parameters settings.

## System TCP Keepalive Configuration

TCP setting are global, and then enabled or disabled on a per-interface basis.

To configure TCP keepalive parameters on your Oracle Communications Session Border Controller:

 **Note:** If you want to use the default values for TCP keepalive, you do not need to take Steps 1 through 4. You can simply set the TCP keepalive function in the H.323 stack configuration, and the defaults for network parameters will be applied.

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-related configurations.

```
ACMEPACKET(configure)# system
```

3. Type **network-parameters** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(system)# network-parameters
ACMEPACKET(network-parameters)#
```

4. **tcp-keepinit-timer**—If a TCP connection cannot be established within some amount of time, TCP will time out the connect attempt. It can be used to set the initial timeout period for a given socket, and specifies the number of seconds to wait before the connect attempt is timed out. For passive connections, this value is inherited from the listening socket. The default is 75. The valid range is:

- Minimum—0
- Maximum—999999999.

5. **tcp-keepalive-count**—Enter the number of packets the Oracle Communications Session Border Controller sends to the remote peer before it terminates the TCP connection. The default is 8. The valid range is:

- Minimum—0
- Maximum—223-1

6. **tcp-keepalive-idle-timer**—Enter the number of seconds of idle time before TCP keepalive messages are sent to the remote peer if the **SO-KEEPALIVE** option is set. This option is set via the **h323-stack** configuration element. The default is **7200**. The valid range is:
  - Minimum—30
  - Maximum—7200
7. **tcp-keepalive-interval-timer**—When the **SO\_KEEPALIVE** option is enabled, TCP probes a connection that has been idle for some amount of time. If the remote system does not respond to a keepalive probe, TCP retransmits the probe after a set amount of time. This parameter specifies the number of seconds to wait before retransmitting a keepalive probe. The default value is **75** seconds. **The valid range is:**
  - Minimum—15
  - Maximum—75
8. **tcp-keepalive-mode**—Set the TCP keepalive response sequence number. The default is **0**. The valid values are:
  - 0—The sequence number is sent un-incremented
  - 1—The number is incremented
  - 2—No packets are sent

## Configurable TCP Timers

You can configure your Oracle Communications Session Border Controller to detect failed TCP connections more quickly so that data can be transmitted via an alternate connection before timers expire. Across all protocols, you can now control the following for TCP:

- Connection establishment
- Data retransmission
- Timer for idle connections

These capabilities all involve configuring an **options** parameter that appears in the network parameters configuration.

## Configuring TCP Connection Establishment

To establish connections, TCP uses a three-way handshake during which two peers exchange TCP SYN messages to request and confirm the active open connection. In attempting this connection, one peer retransmits the SYN messages for a defined period of time if it does not receive acknowledgement from the terminating peer. You can configure the amount of time in seconds between the retries as well as how long (in seconds) the peer will keep retransmitting the messages.

You set two new options in the network parameters configuration to specify these amounts of time: **atcp-syn-rxmt-interval** and **atcp-syn-rxmt-maxtime**.

Note that for all configured options, any values entered outside of the valid range are silently ignored during configuration and generate a log when you enter the **activate** command.

To configure TCP connection establishment:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter.

```
ACMEPACKET(configure)# system
ACMEPACKET(system)#
```

3. Type **network-parameters** and press Enter.


```
ACMEPACKET(system)# network-parameters
ACMEPACKET(network-parameters)#
```

4. **options**—Set the options parameter by typing **options**, a Space, the option name **atcp-syn-rxmt-interval=x** (where x is a value in seconds between 2 and 10) with a plus sign in front of it. Then press Enter. This value will be used as the interval between TCP SYN messages when the Oracle Communications Session Border Controller is trying to establish a connection with a remote peer.

Now enter a second option to set the maximum time for trying to establish a TCP connection. Set the options parameter by typing **options**, a Space, the option name **atcp-syn-rxmt-maxtime=x** (where x is a value in seconds between 5 and 75) with a plus sign in front of it. Then press Enter.

```
ACMEPACKET(network-parameters)# options +atcp-syn-rxmt-interval=5
ACMEPACKET(network-parameters)# options +atcp-syn-rxmt-maxtime=30
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

 **Note:** **atcp-syn-rxmt-maxtime=x** option is equivalent to the **tcp-keepinit-timer** parameter, but only affects ATCP.

5. Save and activate your configuration.

## Configuring TCP Data Retransmission

TCP is considered reliable in part because it requires that entities receiving data must acknowledge transmitted segments. If data segments go unacknowledged, then they are retransmitted until they are finally acknowledged or until the maximum number of retries has been reached. You can control both the number of times the Oracle Communications Session Border Controller tries to retransmit unacknowledged segments and the periodic interval (how often) at which retransmissions occur.

You set two new options in the network parameters configuration to specify how many retransmissions are allowed and for how long: **atcp-rxmt-interval** and **atcp-rxmt-count**.

To configure TCP data retransmission:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter.

```
ACMEPACKET(configure)# system
ACMEPACKET(system)#
```

3. Type **network-parameters** and press Enter.

```
ACMEPACKET(system)# network-parameters
ACMEPACKET(network-parameters)#
```

4. **options**—Set the options parameter by typing **options**, a Space, the option name **atcp-rxmt-interval=x** (where x is a value in seconds between 2 and 60) with a plus sign in front of it. Then press Enter. This value will be used as the interval between retransmission of TCP data segments that have not been acknowledged.

Now enter a second option to set the number of times the Oracle Communications Session Border Controller will retransmit a data segment before it declares the connection failed. Set the options parameter by typing **options**, a Space, the option name **atcp-rxmt-count=x** (where x is a value between 4 and 12 representing how many retransmissions you want to enable) with a plus sign in front of it. Then press Enter.

```
ACMEPACKET(network-parameters)# options +atcp-rxmt-interval=30
ACMEPACKET(network-parameters)# options +atcp-rxmt-count=6
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the



configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

## Timer for Idle Connections

When enabled to do so, the Oracle Communications Session Border Controller monitors inbound TCP connections for inactivity. These are inbound connections that the remote peer initiated, meaning that the remote peer sent the first SYN message. You can configure a timer that sets the maximum amount of idle time for a connection before the Oracle Communications Session Border Controller consider the connection inactive. Once the timer expires and the connection is deemed inactive, the Oracle Communications Session Border Controller sends a TCP RST message to the remote peer.

To configure the timer for TCP idle connections:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter.

```
ACMEPACKET (configure) # system
ACMEPACKET (system) #
```

3. Type **network-parameters** and press Enter.

```
ACMEPACKET (system) # network-parameters
ACMEPACKET (network-parameters) #
```

4. **options**—Set the options parameter by typing **options**, a Space, the option name **atcp-idle-timer=x** (where x is a value in seconds between 120 and 7200) with a plus sign in front of it. Then press Enter. This value will be used to measure the activity of TCP connections; when the inactivity on a TCP connection reaches this value in seconds, the Oracle Communications Session Border Controller declares it inactive and drops the session.

```
ACMEPACKET (network-parameters) # options +atcp-idle-timer=900
```

**If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.**

5. Save and activate your configuration.

## Historical Data Recording (HDR)

Updated HDR and HDR configuration information resides in the Net-Net® C-Series Historical Data Recording (HDR) Resource Guide Version C6.3.0, 400-0141-63 (Net-Net 4000 S-CX6.3.0 HDR Resource Guide.pdf). This document is available with the complete Version S-CX6.3.0 documentation set.

## RAMdrive Log Cleaner

The RAMdrive log cleaner allows the Oracle Communications Session Border Controller to remove log files proactively and thereby avoid situations where running low on RAMdrive space is a danger. Because even a small amount of logging can consume a considerable space, you might want to enable the RAMdrive log cleaner.

The RAMdrive cleaner periodically checks the remaining free space in the RAMdrive and, depending on the configured threshold, performs a full check on the /ramdrv/logs directory. During the full check, the RAMdrive cleaner determines the total space logs files are using and deletes log files that exceed the configured maximum lifetime. In addition, if the cleaner finds that the maximum log space has been exceeded or the minimum free space is not sufficient, it deletes older log files until the thresholds are met.

## System Configuration

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Not all log files, however, are as active as others. This condition affects which log files the log cleaner deletes to create more space in RAMdrive. More active log files rotate through the system more rapidly. So, if the log cleaner were to delete the oldest of these active files, it might not delete less active logs files that could be older than the active ones. The log cleaner thus deletes files that are truly older, be they active or inactive.

### Applicable Settings

In the system configuration, you establish a group of settings in the options parameter that control the log cleaner's behavior:

- **ramdrv-log-min-free**—Minimum percent of free space required when rotating log files.

When the amount of free space on the RAMdrive falls below this value, the log cleaner deletes the oldest copy of the log file. The log cleaner also uses this setting when performing period cleaning.

- **ramdrv-log-max-usage**—Maximum percent of the RAMdrive the log files can use.

The log cleaner removes old log files to maintain this threshold.

- **ramdrv-log-min-check**—Minimum percent of free space on the RAMdrive that triggers the log cleaner to perform a full check of log files.
- **ramdrv-min-log-check**—Minimum time (in seconds) between log cleaner checks.
- **ramdrv-max-log-check**—Maximum time (in seconds) between log cleaner checks. This value must be greater than or equal to the **ramdrv-min-log-check**.
- **ramdrv-log-lifetime**—Maximum lifetime (in days) for log files. You give logs unlimited lifetime by entering a value of 0.

### Clean-Up Procedure

The log cleaner checks the amount of space remaining in the RAMdrive and performs a full check of the logs directory when:

- Free space is less than the minimum percent of the RAMdrive that triggers a full check of log files
- The amount of free space has changed by more than 5% of the RAMdrive capacity since the last full check
- A full check of the logs directory has not been performed in the last hour

When it checks the logs directory, the log cleaner inventories the collected log files. It identifies each files as one of these types:

- Process log—Files beginning with log.
- Internal trace file—A <task>.log file
- Protocol trace file—Call trace including sipmsg.log, dns.log, sipddns.log, and alg.log
- CDR file—File beginning with cdr

Next, the log cleaner determines the age of the log files using the number of seconds since the log files were created. Then it orders the files from oldest to newest. The age adjusts such that it always increases as the log file sequence number (a suffix added by file rotation) increases. The log cleaner applies an additional weighting factor to produce a weighted age that favors the preservation of protocol traces files over internal trace files, and internal trace files over process log files. The base log file and CDR files are excluded from the age list and so will not be deleted; the accounting configuration controls CDR file aging.

With the age list constructed, the log cleaner examines the list from highest weighted age to lowest. If the actual file age exceeds the RAMdrive maximum log lifetime, the log cleaner deletes it. Otherwise, the log cleaner deletes files until the maximum percent of RAMdrive that logs can use is no longer exceeded and until the minimum percent of free space required when rotating logs is available.

### Clean-Up Frequency

The minimum free space that triggers a full check of log files and the maximum time between log file checks control how often the log cleaner performs the clean-up procedure. When it completes the procedure, the log cleaner determines the time interval until the next required clean-up based on the RAMdrive's state.

If a clean-up results in the deletion of one or more log files or if certain thresholds are exceeded, frequency is based on the minimum time between log cleaner checks. Otherwise, the system gradually increases the interval up to the maximum time between log cleaner checks. The system increases the interval by one-quarter of the difference between the minimum and maximum interval, but not greater than one-half the minimum interval or smaller than 10 seconds. For example, using the default values, the interval would be increased by 30 seconds.

## RAMdrive Log Cleaner Configuration

You configure the log cleaner's operating parameters and thresholds in the system configuration. Note that none of these settings is RTC-supported, so you must reboot your Oracle Communications Session Border Controller in order for them to take effect. If you are using this feature on an HA node, however, you can add this feature without impact to service by activating the configuration, rebooting the standby, switching over to make the newly booted standby active, and then rebooting the newly standby system.

Unlike other values for **options** parameters, the Oracle Communications Session Border Controller validates these setting when entered using the ACLI. If any single value is invalid, they all revert to their default values.

To configure the RAMdrive log cleaner:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **system** and press Enter.

```
ACMEPACKET (configure) # system
ACMEPACKET (system) #
```

3. Type **system-config** and press Enter.

```
ACMEPACKET (system) # system-config
ACMEPACKET (system-config) #
```

4. **options**—Set the options parameter by typing options, a Space, <option name>=X (where X is the value you want to use) with a plus sign in front of it. Then press Enter.

Remember that if any of your settings are invalid, the Oracle Communications Session Border Controller changes the entire group of these options back to their default settings.

Option Name	Description
ramdrv-log-min-free	Minimum percent of free space required when rotating log files.  When the amount of free space on the RAMdrive falls below this value, the log cleaner deletes the oldest copy of the log file. The log cleaner also uses this setting when performing period cleaning.  Default=40; Minimum=15; Maximum=75
ramdrv-log-max-usage	Maximum percent of the RAMdrive the log files can use.  The log cleaner removes old log files to maintain this threshold.  Default=40; Minimum=15; Maximum=75
ramdrv-log-min-check	Minimum percent of free space on the RAMdrive that triggers the log cleaner to perform a full check of log files.  Default=50; Minimum=25; Maximum=75
ramdrv-min-log-check	Maximum time (in seconds) between log cleaner checks. This value must be greater than or equal to the ramdrv-min-log-check.  Default=180; Minimum=40; Maximum=1800

## System Configuration

Option Name	Description
ramdrv--log-lifetime	Maximum lifetime (in days) for log files. You give logs unlimited lifetime by entering a value of 0.  Default=30; Minimum=2; Maximum=9999

```
ACMEPACKET (system-config) # options +ramdrv-log-min-free=50
ACMEPACKET (system-config) # options +ramdrv-log-max-usage=50
ACMEPACKET (system-config) # options +ramdrv-log-min-check=35
ACMEPACKET (system-config) # options +ramdrv-min-log-check=120
ACMEPACKET (system-config) # options +ramdrv-max-log-free=1500
ACMEPACKET (system-config) # options +ramdrv-log-lifetime=7
```

If you type **options** and then the option value for either of these entries without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Reboot your Oracle Communications Session Border Controller.

## Configurable Alarm Thresholds and Traps

The Oracle Communications Session Border Controller supports user-configurable threshold crossing alarms. These configurations let you identify system conditions of varying severity which create corresponding alarms of varying severity. You configure an alarm threshold type which indicates the resource to monitor. The available types are:

- **cpu** — CPU utilization monitored as a percentage of total CPU capacity
- **memory** — memory utilization monitored as a percentage of total memory available
- **sessions** — license utilization monitored as a percentage of licensed session capacity
- **space** — remaining disk space (configured in conjunction with the volume parameter - see the Storage Expansion Module Monitoring section of the *Accounting Guide* for more information.)
- **deny-allocation** — denied entry utilization monitored as a percentage of reserved, denied entries.

For the alarm type you create, the Oracle Communications Session Border Controller can monitor for 1 through 3 severity levels as minor, major, and critical. Each of the severities is configured with a corresponding value that triggers that severity. For example the configuration for a CPU alarm that is enacted when CPU usage reaches 50%:

```
alarm-threshold
  type          cpu
  severity      minor
  value         50
```

You may create addition CPU alarms for increasing severities. For example:

```
alarm-threshold
  type          cpu
  severity      critical
  value         90
```

The alarm state is enacted when the resource defined with the type parameter exceeds the value parameter. When the resource drops below the value parameter, the alarm is cleared.

## SNMP Traps

When a configured alarm threshold is reached, the Oracle Communications Session Border Controller sends an `apSysMgmtGroupTrap`. This trap contains the resource type and value for the alarm configured in the `alarm-threshold` configuration element. The trap does not contain information associated with configured severity for that value.

```
apSysMgmtGroupTrap          NOTIFICATION-TYPE
  OBJECTS                    { apSysMgmtTrapType, apSysMgmtTrapValue }
  STATUS                      current
  DESCRIPTION
```

```
" The trap will generated if value of the monitoring object
exceeds a certain threshold. "
::= { apSystemManagementNotifications 1 }
```

When the resource usage retreats below a configured threshold, the Oracle Communications Session Border Controller sends an apSysMgmtGroupClearTrap.

```
apSysMgmtGroupClearTrap          NOTIFICATION-TYPE
OBJECTS                          { apSysMgmtTrapType }
STATUS                            current
DESCRIPTION
    " The trap will generated if value of the monitoring object
    returns to within a certain threshold. This signifies that
    an alarm caused by that monitoring object has been cleared. "
::= { apSystemManagementNotifications 2 }
```

The alarm and corresponding traps available through the User Configurable Alarm Thresholds functionality are summarized in the following table.

Alarm	Severity	Cause	Actions
CPU	minor major critical	high CPU usage	apSysMgmtGroupTrap sent with apSysCPUUtil apSysMgmtTrapValue
memory	minor major critical	high memory usage	apSysMgmtGroupTrap sent with apSysMemoryUtil apSysMgmtTrapValue
sessions	minor major critical	high license usage	apSysMgmtGroupTrap sent with apSysLicenseCapacity apSysMgmtTrapValue
space	minor major critical	high HDD usage, per volume	apSysMgmtStorageSpaceAvailThresholdTrap sent with: apSysMgmtSpaceAvailCurrent apSysMgmtSpaceAvailMinorThreshold apSysMgmtSpaceAvailMajorThreshold apSysMgmtSpaceAvailCriticalThreshold apSysMgmtPartitionPath
deny allocation	minor major critical	high usage of denied ACL entries	apSysMgmtGroupTrap sent with apSysCurrentEndptsDenied apSysMgmtTrapValue

## Alarm Thresholds Configuration

To configure alarm thresholds:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

## System Configuration

---

```
ACMEPACKET(configure)# system
```

3. Type system-config and press Enter.

```
ACMEPACKET(system)# system-config
ACMEPACKET(system-config)#
```

4. Type alarm-threshold and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(system-config)# alarm-threshold
ACMEPACKET(alarm-threshold)#
```

5. type — Enter the type of resource which this alarm monitors. Valid values include:
  - cpu
  - memory
  - sessions
  - space
  - deny-allocation
6. volume — Enter the logical disk volume this alarm monitors (used only in conjunction when type = space).
7. severity — Set the severity of the threshold. Valid values include:
  - minor
  - major
  - critical
8. value — Enter the value from 1 to 99, indicating the percentage, which when exceeded generates an alarm.
9. Save and activate your configuration.

## Alarm Synchronization

---

Two trap tables in the ap-smgmt.mib record trap information for any condition on the Oracle Communications Session Border Controller that triggers an alarm condition. You can poll these two tables from network management systems, OSS applications, and the Session Delivery Manager to view the fault status on one or more Oracle Communications Session Border Controller s.

The two trap tables that support alarm synchronization, and by polling them you can obtain information about the current fault condition on the Oracle Communications Session Border Controller . These tables are:

- apSysMgmtTrapTable—You can poll this table to obtain a summary of the Oracle Communications Session Border Controller 's current fault conditions. The table records multiples of the same trap type that have occurred within a second of one another and have different information. Each table entry contains the following:
  - Trap identifier
  - System time (synchronized with an NTP server)
  - sysUpTime
  - Instance number
  - Other trap information for this trap identifier
- apSysMgmtTrapInformationTable—You can poll this table to obtain further details about the traps recorded in the apSysMgmtTrapTable table. The following information appears:
  - Data index
  - Data type
  - Data length
  - The data itself (in octets)

Trap tables do not record information about alarm severity.

The apSysMgmtTrapTable can hold up to 1000 entries, and you can configure the number of days these entries stay in the table for a maximum of seven days. If you set this parameter to 0 days, the feature is disabled. And if you change

the setting to 0 days from a greater value, then the Oracle Communications Session Border Controller purges the tables.

## Caveats

Note that the Oracle Communications Session Border Controller does not replicate alarm synchronization table data across HA nodes. That is, each Oracle Communications Session Border Controller in an HA node maintains its own tables.

## Alarm Synchronization Configuration

You turn on alarm synchronization in the system configuration.

To use alarm synchronization:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **system** and press Enter.

```
ACMEPACKET (configure) # system
ACMEPACKET (system) #
```

3. Type **system-config** and press Enter.

```
ACMEPACKET (system) # system-config
ACMEPACKET (system-config) #
```

4. **trap-event-lifetime**—To enable alarm synchronization—and cause the Oracle Communications Session Border Controller to record trap information in the apSysMgmtTrapTable and the apSysMgmtTrapInformationTable—set this parameter to the number of days you want to keep the information. Leaving this parameter set to 0 (default) turns alarm synchronization off, and you can keep information in the tables for up to 7 days. 7 is the maximum value for this parameter.

## Accounting Configuration

The Oracle Communications Session Border Controller offers support for RADIUS, an accounting, authentication, and authorization (AAA) system. In general, RADIUS servers are responsible for receiving user connection requests, authenticating users, and returning all configuration information necessary for the client to deliver service to the user.

You can configure your Oracle Communications Session Border Controller to send call accounting information to one or more RADIUS servers. This information can help you to see usage and QoS metrics, monitor traffic, and even troubleshoot your system.

This guide contains all RADIUS information, as well as information about:

- Accounting for SIP and H.323
- Local CDR storage on the Oracle Communications Session Border Controller, including CSV file format settings
- The ability to send CDRs via FTP to a RADIUS sever (the FTP push feature)
- Per-realm accounting control
- Configurable intermediate period
- RADIUS CDR redundancy
- RADIUS CDR content control

## Stream Control Transfer Protocol Overview

The Stream Control Transmission Protocol (SCTP) was originally designed by the Signaling Transport (SIGTRAN) group of IETF for Signalling System 7 (SS7) transport over IP-based networks. It is a reliable transport protocol operating on top of an unreliable connectionless service, such as IP. It provides acknowledged, error-free, non-

## System Configuration

---

duplicated transfer of messages through the use of checksums, sequence numbers, and selective retransmission mechanism.

SCTP is designed to allow applications, represented as endpoints, communicate in a reliable manner, and so is similar to TCP. In fact, it has inherited much of its behavior from TCP, such as association (an SCTP peer-to-peer connection) setup, congestion control and packet-loss detection algorithms. Data delivery, however, is significantly different.

SCTP delivers discrete application messages within multiple logical streams within the context of a single association. This approach to data delivery is more flexible than the single byte-stream used by TCP, as messages can be ordered, unordered or even unreliable within the same association.

## SCTP Packets

SCTP packets consist of a common header and one or more chunks, each of which serves a specific purpose.

- DATA chunk — carries user data
- INIT chunk — initiates an association between SCTP endpoints
- INIT ACK chunk — acknowledges association establishment
- SACK chunk — acknowledges received DATA chunks and informs the peer endpoint of gaps in the received subsequences of DATA chunks
- HEARTBEAT chunk — tests the reachability of an SCTP endpoint
- HEARTBEAT ACK chunk — acknowledges reception of a HEARTBEAT chunk
- ABORT chunk — forces an immediate close of an association
- SHUTDOWN chunk — initiates a graceful close of an association
- SHUTDOWN ACK chunk — acknowledges reception of a SHUTDOWN chunk
- ERROR chunk — reports various error conditions
- COOKIE ECHO chunk — used during the association establishment process
- COOKIE ACK chunk — acknowledges reception of a COOKIE ECHO chunk
- SHUTDOWN COMPLETE chunk — completes a graceful association close

## SCTP Terminology

This section defines some terms commonly found in SCTP standards and documentation.

### SCTP Association

is a connection between SCTP endpoints. An SCTP association is uniquely identified by the transport addresses used by the endpoints in the association. An SCTP association can be represented as a pair of SCTP endpoints, for example, `assoc = { [IPv4Addr : PORT1], [IPv4Addr1, IPv4Addr2: PORT2]}`.

Only one association can be established between any two SCTP endpoints.

### SCTP Endpoint

is a sender or receiver of SCTP packets. An SCTP endpoint may have one or more IP address but it always has one and only one SCTP port number. An SCTP endpoint can be represented as a list of SCTP transport addresses with the same port, for example, `endpoint = [IPv6Addr, IPv6Addr: PORT]`.

An SCTP endpoint may have multiple associations.

### SCTP Path

is the route taken by the SCTP packets sent by one SCTP endpoint to a specific destination transport address or its peer SCTP endpoint. Sending to different destination transport addresses does not necessarily guarantee separate routes.

### SCTP Primary Path

is the default destination source address, the IPv4 or IPv6 address of the association initiator. For retransmissions however, another active path may be selected, if one is available.

### SCTP Stream



is a unidirectional logical channel established between two associated SCTP endpoints. SCTP distinguishes different streams of messages within one SCTP association. SCTP makes no correlation between an inbound and outbound stream.

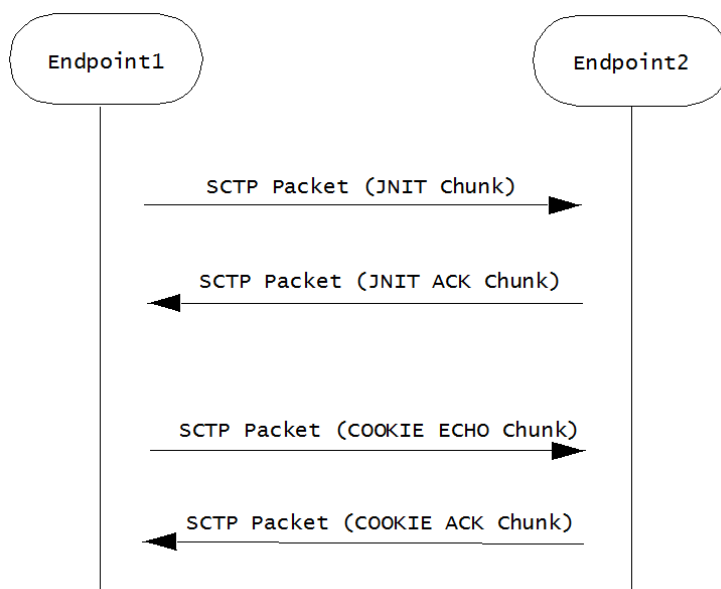
#### SCTP Transport Address

is the combination of an SCTP port and an IP address. For the current release, the IP address portion of an SCTP Transport Address must be a routable, unicast IPv4 or IPv6 address.

An SCTP transport address binds to a single SCTP endpoint.

## SCTP Message Flow

Before peer SCTP users (commonly referred to as endpoints) can send data to each other, an association (an SCTP connection) must be established between the endpoints. During the association establishment process a cookie mechanism is employed to provide protection against security attacks. The following figure shows a sample SCTP association establishment message flow.



Endpoint1 initiates the association by sending Endpoint2 an SCTP packet that contains an INIT chunk, which can include one or more IP addresses used by the initiating endpoint. Endpoint2 acknowledges the initiation of an SCTP association with an SCTP packet that contains an INIT\_ACK chunk. This chunk can also include one or more IP addresses at used by the responding endpoint.

Both the INIT chunk (issued by the initiator) and INIT ACK chunk (issued by the responder) specify the number of outbound streams supported by the association, as well as the maximum inbound streams accepted from the other endpoint.

Association establishment is completed by a COOKIE ECHO/COOKIE ACK exchange that specifies a cookie value used in all subsequent DATA exchanges.

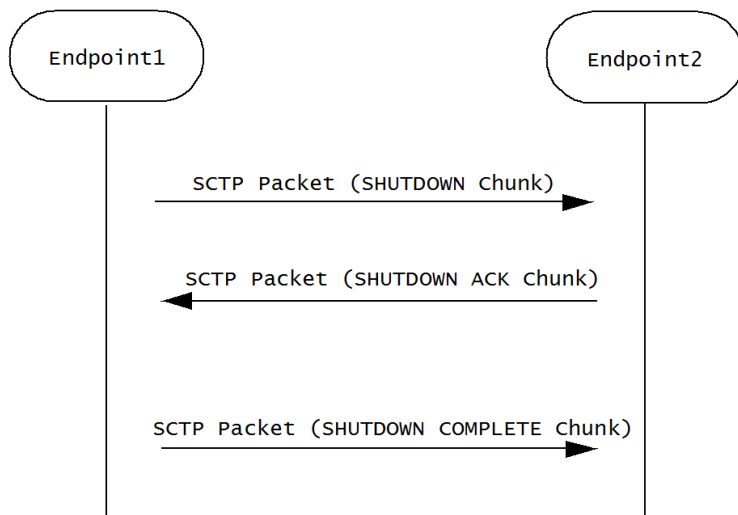
Once an association is successfully established, an SCTP endpoint can send unidirectional data streams using SCTP packets that contain DATA chunks. The recipient endpoint acknowledges with an SCTP packet containing a SACK chunk.

SCTP monitors endpoint reachability by periodically sending SCTP packets that contain HEARTBEAT chunks. The recipient endpoint acknowledges receipt, and confirms availability, with an SCTP packet containing a HEARBEAT ACK chunk.

Either SCTP endpoint can initiate a graceful association close with an SCTP packet that contains a SHUTDOWN chunk. The recipient endpoint acknowledges with an SCTP packet containing a SHUTDOWN ACK chunk. The

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initiating endpoint concludes the graceful close with an SCTP packet that contains a SHUTDOWN COMPLETE chunk.

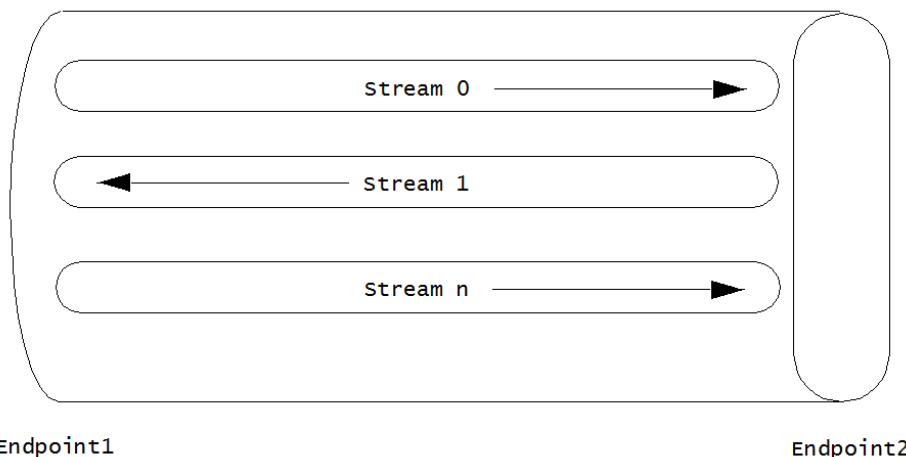


## Congestion Control

SCTP congestion control mechanism is similar to that provided by TCP, and includes slow start, congestion avoidance, and fast retransmit. In SCTP, the initial congestion window ( cwnd ) is set to the double of the maximum transmission unit (MTU) while in TCP, it is usually set to one MTU. In SCTP, cwnd increases based on the number of acknowledged bytes, rather than the number of acknowledgements in TCP. The larger initial cwnd and the more aggressive cwnd adjustment provided by SCTP result in a larger average congestion window and, hence, better throughput performance than TCP.

## Multi-Streaming

SCTP supports streams as depicted in the following figure which depicts an SCTP association that supports three streams.



The multiple stream mechanism is designed to solve the head-of-the-line blocking problem of TCP. Therefore, messages from different multiplexed flows do not block one another.

A stream can be thought of as a sub-layer between the transport layer and the upper layer. SCTP supports multiple logical streams to improve data transmission throughput. As shown in the above figure, SCTP allows multiple unidirectional streams within an association. This multiplexing/de-multiplexing capability is called multi-streaming

and it is achieved by introducing a field called Stream Identifier contained in every DATA chunk) that is used to differentiate segments in different streams.

SIP transactions are mapped into SCTP streams as described in Section 5.1 of RFC 4168. In what it describes as the simplest way, the RFC suggests (keyword SHOULD) that all SIP messages be transmitted via Stream 0 with the U bit set to 1.

On the transmit side, the current SCTP implementation follows the RFC 4168 recommendation. On the receiving side, a SIP entity must be prepared to receive SIP messages over any stream.

## **Delivery Modes**

SCTP supports two delivery modes, ordered and unordered. Delivery mode is specified by the U bit in the DATA chunk header — if the bit is clear (0), ordered delivery is specified; if the bit is set (1), unordered delivery is specified.

Within a stream, an SCTP endpoint must deliver ordered DATA chunks (received with the U bit set to 0) to the upper layer protocol according to the order of their Stream Sequence Number. Like the U bit, the Stream Sequence Number is a field within the DATA chunk header, and serves to identify the chunk's position with the message stream. If DATA chunks arrive out of order of their Stream Sequence Number, the endpoint must delay delivery to the upper layer protocol until they are reordered and complete.

Unordered DATA chunks (received with the U bit set to 1) are processed differently. When an SCTP endpoint receives an unordered DATA chunk, it must bypass the ordering mechanism and immediately deliver the data to the upper layer protocol (after reassembly if the user data is fragmented by the sender). As a consequence, the Stream Sequence Number field in an unordered DATA chunk has no significance. The sender can fill it with arbitrary value, but the receiver must ignore any value in field.

When an endpoint receives a DATA chunk with the U flag set to 1, it must bypass the ordering mechanism and immediately deliver the data to the upper layer (after reassembly if the user data is fragmented by the data sender).

Unordered delivery provides an effective way of transmitting out-of-band data in a given stream. Note also, a stream can be used as an unordered stream by simply setting the U bit to 1 in all DATA chunks sent through that stream.

## **Multi-Homing**

Call control applications for carrier-grade service require highly reliable communication with no single point of failure. SCTP can assist carriers with its multi-homing capabilities. By providing different paths through the network over separate and diverse means, the goal of no single point of failure is more easily attained.

SCTP built-in support for multi-homed hosts allows a single SCTP association to run across multiple links or paths, hence achieving link/path redundancy. With this capability, an SCTP association can be made to achieve fast failover from one link/path to another with little interruption to the data transfer service.

Multi-homing enables an SCTP host to establish an association with another SCTP host over multiple interfaces identified by different IP addresses. With specific regard to the Oracle Communications Session Border Controller these IP addresses need not be assigned to the same physical interface, or to the same physical Network Interface Unit.

If the SCTP nodes and the according IP network are configured in such a way that traffic from one node to another travels on physically different paths if different destination IP address are used, associations become tolerant against physical network failures and other problems of that kind.

An endpoint can choose an optimal or suitable path towards a multi-homed destination. This capability increases fault tolerance. When one of the paths fails, SCTP can still choose another path to replace the previous one. Data is always sent over the primary path if it is available. If the primary path becomes unreachable, data is migrated to a different, affiliated address — thus providing a level of fault tolerance. Network failures that render one interface of a server unavailable do not necessarily result in service loss. In order to achieve real fault resilient communication between two SCTP endpoints, the maximization of the diversity of the round-trip data paths between the two endpoints is encouraged.

### Multi-Homing and Path Diversity

As previously explained, when a peer is multi-homed, SCTP can automatically switch the subsequent data transmission to an alternative address. However, using multi-homed endpoints with SCTP does not automatically guarantee resilient communications. One must also design the intervening network(s) properly.

To achieve fault resilient communication between two SCTP endpoints, one of the keys is to maximize the diversity of the round-trip data paths between the two endpoints. Under an ideal situation, one can make the assumption that every destination address of the peer will result in a different, separate path towards the peer. Whether this can be achieved in practice depends entirely on a combination of factors that include path diversity, multiple connectivity, and the routing protocols that glue the network together. In a normally designed network, the paths may not be diverse, but there may be multiple connectivity between two hosts so that a single link failure will not fail an association.

In an ideal arrangement, if the data transport to one of the destination addresses (which corresponds to one particular path) fails, the data sender can migrate the data traffic to other remaining destination address(es) (that is, other paths) within the SCTP association.

### Monitoring Failure Detection and Recovery

When an SCTP association is established, a single destination address is selected as the primary destination address and all new data is sent to that primary address by default. This means that the behavior of a multi-homed SCTP association when there are no network losses is similar to behavior of a TCP connection. Alternate, or secondary, destination addresses are only used for redundancy purposes, either to retransmit lost packets or when the primary destination address cannot be reached.

A failover to an alternate destination is performed when the SCTP sender cannot elicit an acknowledgement — either a SACK for a DATA chunk, or a HEARTBEAT ACK for a HEARTBEAT chunk — for a configurable consecutive number of transmissions. The SCTP sender maintains an error-counter is maintained for each destination address and if this counter exceeds a threshold (normally six), the address is marked as inactive, and taken out of service. If the primary destination address is marked as inactive, all data is then switched to a secondary address to complete the failover.

If no data has been sent to an address for a specified time, that endpoint is considered to be idle and a HEARTBEAT packet is transmitted to it. The endpoint is expected to respond to the HEARTBEAT immediately with a HEARTBEAT ACK. As well as monitoring the status of destination addresses, the HEARTBEAT is used to obtain RTT measurements on idle paths. The primary address becomes active again if it responds to a heartbeat.

The number of events where heartbeats were not acknowledged within a certain time, or retransmission events occurred is counted on a per association basis, and if a certain limit is exceeded, the peer endpoint is considered unreachable, and the association is closed.

The threshold for detecting an endpoint failure and the threshold for detecting a failure of a specific IP addresses of the endpoint are independent of each other. Each parameter can be separately configured by the SCTP user. Careless configuration of these protocol parameters can lead the association onto the dormant state in which all the destination addresses of the peer are found unreachable while the peer still remains in the reachable state. This is because the overall retransmission counter for the peer is still below the set threshold for detecting the peer failure.

### Configuring SCTP Support for SIP

---

RFC 4168, *The Stream Control Transfer Protocol (SCTP) as a Transport for the Session Initiation Protocol (SIP)*, specifies the requirements for SCTP usage as a layer 4 transport for SIP. Use the following steps to:

- configure SCTP as the layer 4 transport for a SIP interface
- create an SCTP-based SIP port
- associate physical interfaces/network interfaces with SIP realms
- identify adjacent SIP servers that are accessible via SCTP
- set SCTP timers and counters (optional)

## Configuring an SCTP SIP Port

SIP ports are created as part of the SIP Interface configuration process.

1. From superuser mode, use the following command sequence to access sip-port configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)# sip-ports
ACMEPACKET(sip-port)#
```

2. Use the **address** parameter to provide the IPv4 or IPv6 address of the network interface that supports the SIP port.

This is the primary address of a the local multi-homed SCTP endpoint.

```
ACMEPACKET(sip-port)# address 172.16.10.76
ACMEPACKET(sip-port)#
```

3. Retain the default value, 5060 (the well-known SIP port) for the **port** parameter.

```
ACMEPACKET(sip-port)# port 5060
ACMEPACKET(sip-port)#
```

4. Use the **transport-protocol** parameter to identify the layer 4 protocol.

Supported values are UDP, TCP, TLS, and SCTP.

Select SCTP.

```
ACMEPACKET(sip-port)# transport-protocol sctp
ACMEPACKET(sip-port)#
```

5. Use the **multi-homed-addr**s parameter to specify one or more local secondary addresses of the SCTP endpoint.

Multi-homed addresses must be of the same type (IPv4 or IPv6) as that specified by the **address** parameter. Like the address parameter, these addresses identify SD physical interfaces.

To specify multiple addresses, bracket an address list with parentheses.

```
ACMEPACKET(sip-port)# multi-homed-addr 182.16.10.76
ACMEPACKET(sip-port)#
ACMEPACKET(sip-port)# multi-homed-addr (182.16.10.76 192.16.10.76
196.15.32.108)
ACMEPACKET(sip-port)#
```

6. Remaining parameters can be safely ignored.

7. Use **done**, **exit**, and **verify-config** to complete configuration of this SCTP-based SIP port.

```
ACMEPACKET(sip-port)# done
ACMEPACKET(sip-interface)# exit
ACMEPACKET(session-router)# exit
ACMEPACKET(configure)# exit
ACMEPACKET# verify-config
-----
Verification successful! No errors nor warnings in the configuration
ACMEPACKET#
```

## Configuring the Realm

After configuring a SIP port which identifies primary and secondary multi-homed transport addresses, you identify the network interfaces that support the primary address and secondary addresses to the realm assigned during SIP Interface configuration.

1. From superuser mode, use the following command sequence to access realm-config configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
```

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---

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

2. Use the **select** command to access the target realm.
3. Use the **network-interfaces** command to identify the network interfaces that support the SCTP primary and secondary addresses.

Network interfaces are identified by their name.

Enter a list of network interface names using parentheses as list brackets. The order of interface names is not significant.

```
ACMEPACKET(realm-config)# network-interfaces (mo1 M10)
ACMEPACKET(realm-config)#
```

4. Use **done**, **exit**, and **verify-config** to complete realm configuration.

```
ACMEPACKET(realm-config)# done
ACMEPACKET(media-manager)# exit
ACMEPACKET(configure)# exit
ACMEPACKET# verify-config
```

```
-----
Verification successful! No errors nor warnings in the configuration
ACMEPACKET#
```

## Configuring Session Agents

After configuring the realm, you identify adjacent SIP servers who will be accessed via the SCTP protocol.

1. From superuser mode, use the following command sequence to access session-agent configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

2. Use the **select** command to access the target session-agent.
3. Use the **transport-method** parameter to select the layer 4 transport protocol.

Select staticSCTP for SCTP transport

```
ACMEPACKET(session-agent)# transport-method staticSCTP
ACMEPACKET(session-agent)#
```

4. Set the **reuse-connections** parameter to none.

Select staticSCTP for SCTP transport

```
ACMEPACKET(session-agent)# reuse-connections none
ACMEPACKET(session-agent)#
```

5. Use **done**, **exit**, and **verify-config** to complete session agent configuration.

```
ACMEPACKET(session-agent)# done
ACMEPACKET(session-router)# exit
ACMEPACKET(configure)# exit
ACMEPACKET# verify-config
```

```
-----
Verification successful! No errors nor warnings in the configuration
ACMEPACKET#
```

6. Repeat Steps 1 through 5 as necessary to configure additional session agents who will be accessed via SCTP transport.

## Setting SCTP Timers and Counters

Setting SCTP timers and counters is optional. All configurable timers and counters provide default values that conform to recommended values as specified in RFC 4960, Stream Control Transmission Protocol.

Management of Retransmission Timer, section 6.3 of RFC 4960 describes the calculation of a Retransmission Timeout (RTO) by the SCTP process. This calculation involves three SCTP protocol parameters: RTO.Initial, RTO.Min, and RTO.Max. Suggested SCTP Protocol Parameter Values section 15 of RFC 4960 lists recommended values for these parameters.

The following shows the equivalence of recommended values and ACLI defaults.

RTO.Initial = 3 seconds **sctp-rto-initial = 3000 ms (default value)**

RTO.Min = 1 second **sctp-rto-min = 1000 ms (default value)**

RTO.Max = 60 seconds **sctp-rto-max = 60000 ms (default value)**

Path Heartbeat, section 8.3 of RFC 4960 describes the calculation of a Heartbeat Interval by the SCTP process. This calculation involves the current calculated RTO and a single SCTP protocol parameter — HB.Interval.

The following shows the equivalence of recommended the value and ACLI default.

HB.Interval = 30 seconds **sctp-hb-interval = 3000 ms (default value)**

Acknowledgement on Reception of DATA Chunks, section 6.2 of RFC 4960 describes requirements for the timely processing and acknowledgement of DATA chunks. This section requires that received DATA chunks must be acknowledged within 500 milliseconds, and recommends that DATA chunks should be acknowledged with 200 milliseconds. The interval between DATA chunk reception and acknowledgement is specific by the ACLI **sctp-sack-timeout** parameter, which provides a default value of 200 milliseconds and a maximum value of 500 milliseconds.

Transmission of DATA Chunks, section 6.1 of RFC 4960 describes requirements for the transmission of DATA chunks. To avoid network congestion the RFC recommends a limitation on the volume of data transmitted at one time. The limitation is expressed in terms of DATA chunks, not in terms of SCTP packets.

The maximum number of DATA chunks that can be transmitted at one time is specified by the ACLI **sctp-max-burst** parameter, which provides a default value of 4 chunks, the limit recommended by the RFC.

## Setting the RTO

An SCTP endpoint uses a retransmission timer to ensure data delivery in the absence of any feedback from its peer. RFC 4960 refers to the timer itself as T3-rtx and to the timer duration as RTO (retransmission timeout).

When an endpoint's peer is multi-homed, the endpoint calculates a separate RTO for each IP address affiliated with the peer. The calculation of RTO in SCTP is similar to the way TCP calculates its retransmission timer. RTO fluctuates over time in response to actual network conditions. To calculate the current RTO, an endpoint maintains two state variables per destination IP address — the SRTT (smoothed round-trip time) variable, and the RTTVAR (round-trip time variation) variable.

Use the following procedure to assign values used in RTO calculation.

1. From superuser mode, use the following command sequence to access network-parameters configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# system
ACMEPACKET(system)# network-parameters
ACMEPACKET(network-parameters)#
```

2. Use the **sctp-rto-initial** parameter to assign an initial timer duration.

Allowable values are integers within the range 0 through 4294967295 that specify the initial duration in milliseconds. In the absence of an explicitly configured integer value, **sctp-rto-initial** defaults to 3000 milliseconds (3 seconds, the recommended default value from RFC 4960).

As described in Section 6.3 of RFC 4960, the value specified by **sctp-rto-initial** is assigned to the SCTP protocol parameter RTO.Initial, which provides a default RTO until actual calculations have derived a fluctuating duration based on network usage. The value specified by the **sctp-rto-initial** parameter seeds these calculations.

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```
ACMEPACKET(network-parameters) # sctp-rto-initial 3000
ACMEPACKET(network-parameters) #
```

3. Use the **sctp-rto-min** and **sctp-rto-max** parameters to assign an RTO floor and ceiling.

Allowable values are integers within the range 0 through 4294967295 that specify the minimum and maximum durations in milliseconds. In the absence of an explicitly configured integer value, **sctp-rto-min** defaults to 1000 ms (1 second, the recommended default value from RFC 4960), and **sctp-rto-max** defaults to 60000 ms (60 seconds, the recommended default value from RFC 4960.)

As described in Section 6.3 of RFC 4960, the values specified by **sctp-rto-min** and **sctp-rto-max** are assigned to the SCTP protocol parameters, RTO.min and RTO.max that limit RTO calculations. If a calculated RTO duration is less than RTO.min, the parameter value is used instead of the calculated value; likewise, if a calculated RTO duration is greater than RTO.max, the parameter value is used instead of the calculated value.

```
ACMEPACKET(network-parameters) # sctp-rto-min 1000
ACMEPACKET(network-parameters) # sctp-rto-max 60000
ACMEPACKET(network-parameters) #
```

4. Use **done**, **exit**, and **verify-config** to complete RTO configuration.

```
ACMEPACKET(network-parameters) # done
ACMEPACKET(system) # exit
ACMEPACKET(configure) # exit
ACMEPACKET(configure) # exit
ACMEPACKET# verify-config
-----
Verification successful! No errors nor warnings in the configuration
ACMEPACKET#
```

### Setting the Heartbeat Interval

Both single-homed and multi-homed SCTP endpoints test the reachability of associates by sending periodic HEARTBEAT chunks to UNCONFIRMED or idle transport addresses.

Use the following procedure to assign values used in Heartbeat Interval calculation.

1. From superuser mode, use the following command sequence to access network-parameters configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) # system
ACMEPACKET(system) # network-parameters
ACMEPACKET(network-parameters) #
```

2. Use the **sctp-hb-interval** parameter to assign an initial Heartbeat Interval duration.

Allowable values are integers within the range 0 through 4294967295 that specify the initial Heartbeat Interval in milliseconds. In the absence of an explicitly configured integer value, **sctp-hb-interval** defaults to 30000 milliseconds (30 seconds, the recommended default value from RFC 4960).

As described in Section 8.3 of RFC 4960, the value specified by **sctp-hb-interval** is assigned to the SCTP protocol parameter HB.Interval, which provides a default interval until actual calculations have derived a fluctuating interval based on network usage. The value specified by the **sctp-hb-interval** parameter is used during these calculations.

```
ACMEPACKET(network-parameters) # sctp-hb-interval 30000
ACMEPACKET(network-parameters) #
```

3. Use **done**, **exit**, and **verify-config** to complete Heartbeat Interval configuration.

```
ACMEPACKET(network-parameters) # done
ACMEPACKET(system) # exit
ACMEPACKET(configure) # exit
ACMEPACKET(configure) # exit
ACMEPACKET# verify-config
-----
Verification successful! No errors nor warnings in the configuration
ACMEPACKET #
```



## Setting the SACK Delay Timer

An SCTP Selective Acknowledgement (SACK) is sent to the peer endpoint to acknowledge received DATA chunks and to inform the peer endpoint of gaps in the received subsequences of DATA chunks. Section 6.2 of RFC 4960 sets a specific requirement for a SACK Delay timer that specifies the maximum interval between the reception of an SCTP packet containing one or more DATA chunks and the transmission of a SACK to the packet originator.

Use the following procedure to set the SACK Delay timer.

1. From superuser mode, use the following command sequence to access network-parameters configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) # system
ACMEPACKET (system) # network-parameters
ACMEPACKET (network-parameters) #
```

2. Use the **sctp-sack-timeout** parameter to assign a value to the SACK Delay timer.

Allowable values are integers within the range 0 through 500 which specify the maximum delay (in milliseconds) between reception of a SCTP packet containing one or more Data chunks and the transmission of a SACK to the packet source. The value 0 indicates that a SACK is generated immediately upon DATA chunk reception

In the absence of an explicitly configured integer value, **sctp-sack-timeout** defaults to 200 ms (the recommended default value from RFC 4960).

```
ACMEPACKET (network-parameters) # sctp-sack-timeout 200
ACMEPACKET (network-parameters) #
```

3. Use **done**, **exit**, and **verify-config** to complete configuration of the SACK Delay timer.

```
ACMEPACKET (network-parameters) # done
ACMEPACKET (system) # exit
ACMEPACKET (configure) # exit
ACMEPACKET (configure) # exit
ACMEPACKET # verify-config
-----
Verification successful! No errors nor warnings in the configuration
ACMEPACKET #
```

## Limiting DATA Bursts

Section 6.1 of RFC 4960 describes the SCTP protocol parameter, Max.Burst, used to limit the number of DATA chunks that are transmitted at one time.

Use the following procedure to assign a value to the SCTP protocol parameter, Max.Burst.

1. From superuser mode, use the following command sequence to access network-parameters configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) # system
ACMEPACKET (system) # network-parameters
ACMEPACKET (network-parameters) #
```

2. Use the **sctp-max-burst** parameter to assign a value to the SCTP protocol parameter, Max.Burst.

Allowable values are integers within the range 0 through 4294967295 that specify the maximum number of DATA chunks that will be sent at one time. In the absence of an explicitly configured integer value, **sctp-max-burst** defaults to 4 (DATA chunks, the recommended default value from RFC 4960).

```
ACMEPACKET (network-parameters) # sctp-max-burst 4
ACMEPACKET (network-parameters) #
```

3. Use **done**, **exit**, and **verify-config** to complete configuration of DATA burst limitations.

```
ACMEPACKET (network-parameters) # done
ACMEPACKET (system) # exit
ACMEPACKET (configure) # exit
ACMEPACKET (configure) # exit
ACMEPACKET # verify-config
```

```
-----  
Verification successful! No errors nor warnings in the configuration  
ACMEPACKET#
```

### Setting Endpoint Failure Detection

As described in Monitoring, Failure Detection and Recovery, a single-homed SCTP endpoint maintains a count of the total number of consecutive failed (unacknowledged) retransmissions to its peer. Likewise, a multi-homed SCTP endpoint maintains a series of similar, dedicated counts for all of its destination transport addresses. If the value of these counts exceeds the limit indicated by the SCTP protocol parameter `Association.Max.Retrans`, the endpoint considers the peer unreachable and stops transmitting any additional data to it, causing the association to enter the CLOSED state.

The endpoint resets the counter when (1) a DATA chunk sent to that peer endpoint is acknowledged by a SACK, or (2) a HEARTBEAT ACK is received from the peer endpoint.

Use the following procedure to configure endpoint failure detection.

1. From superuser mode, use the following command sequence to access network-parameters configuration mode.

```
ACMEPACKET# configure terminal  
ACMEPACKET (configure) # system  
ACMEPACKET (system) # network-parameters  
ACMEPACKET (network-parameters) #
```

2. Use the `sctp-assoc-max-retrns` to assign a value to the SCTP protocol parameter `Association.Max.Retrans`.

Allowable values are integers within the range 0 through 4294967295 which specify the maximum number of transmission requests. In the absence of an explicitly configured integer value, `sctp-assoc-max-retrns` defaults to 10 (transmission re-tries, the recommended default value from RFC 4960).

```
ACMEPACKET (network-parameters) # sctp-assoc-max-retrns 10  
ACMEPACKET (network-parameters) #
```

3. Use `done`, `exit`, and `verify-config` to complete endpoint failure detection configuration.

```
ACMEPACKET (network-parameters) # done  
ACMEPACKET (system) # exit  
ACMEPACKET (configure) # exit  
ACMEPACKET (configure) # exit  
ACMEPACKET# verify-config
```

```
-----  
Verification successful! No errors nor warnings in the configuration  
ACMEPACKET#
```

### Setting Path Failure Detection

As described in Monitoring, Failure Detection and Recovery, when its peer endpoint is multi-homed, an SCTP endpoint maintains a count for each of the peer's destination transport addresses.

Each time the T3-rtx timer expires on any address, or when a HEARTBEAT sent to an idle address is not acknowledged within an RTO, the count for that specific address is incremented. If the value of a specific address count exceeds the SCTP protocol parameter `Path.Max.Retrans`, the endpoint marks that destination transport address as inactive.

The endpoint resets the counter when (1) a DATA chunk sent to that peer endpoint is acknowledged by a SACK, or (2) a HEARTBEAT ACK is received from the peer endpoint.

When the primary path is marked inactive (due to excessive retransmissions, for instance), the sender can automatically transmit new packets to an alternate destination address if one exists and is active. If more than one alternate address is active when the primary path is marked inactive, a single transport address is chosen and used as the new destination transport address.

Use the following procedure to configure path failure detection.

1. From superuser mode, use the following command sequence to access network-parameters configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# system
ACMEPACKET(system)# network-parameters
ACMEPACKET(network-parameters)#
```

2. Use the **sctp-path-max-retrans** parameter to assign a value to the SCTP protocol parameter Path.Max.Retrans.

Allowable values are integers within the range 0 through 4294967295 that specify the maximum number of RTOs and unacknowledged HEARTBEATS. In the absence of an explicitly configured integer value, **sctp-path-max-retrans** defaults to 5 (RTO and/or HEARTBEAT errors per transport address, the recommended default value from RFC 4960).

When configuring endpoint and path failure detection, ensure that the value of the **sctp-assoc-max-retrans** parameter is smaller than the sum of the **sctp-path-max-retrans** values for all the remote peer's destination addresses. Otherwise, all the destination addresses can become inactive (unable to receive traffic) while the endpoint still considers the peer endpoint reachable.

```
ACMEPACKET(network-parameters)# sctp-path-max-retrans 5
ACMEPACKET(network-parameters)#
```

3. Use **done**, **exit**, and **verify-config** to complete path failure detection configuration.

```
ACMEPACKET(network-parameters)# done
ACMEPACKET(system)# exit
ACMEPACKET(configure)# exit
ACMEPACKET(configure)# exit
ACMEPACKET# verify-config
-----
Verification successful! No errors nor warnings in the configuration
ACMEPACKET#
```

## Specifying the Delivery Mode

As described in Delivery Modes, SCTP support two delivery modes, ordered and unordered.

1. From superuser mode, use the following command sequence to access network-parameters configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# system
ACMEPACKET(system)# network-parameters
ACMEPACKET(network-parameters)#
```

2. Use the **sctp-send-mode** parameter to select the preferred delivery mode.

Choose ordered or unordered.

```
ACMEPACKET(network-parameters)# sctp-send-mode unordered
ACMEPACKET(network-parameters)#
```

3. Use **done**, **exit**, and **verify-config** to complete delivery mode configuration.

```
ACMEPACKET(network-parameters)# done
ACMEPACKET(system)# exit
ACMEPACKET(configure)# exit
ACMEPACKET(configure)# exit
ACMEPACKET# verify-config
-----
Verification successful! No errors nor warnings in the configuration
ACMEPACKET #
```

## Example Configurations

The following ACLI command sequences summarize required SCTP port configuration, and the configuration of required supporting elements.

- PHY interfaces
- Network interfaces

## System Configuration

---

- SIP ports
- realms
- session agents

Sequences show only configuration parameters essential for SCTP operations; other parameters can retain default values, or assigned other values specific to local network requirements.

### Phy Interface Configuration

The first ACLI command sequence configures a physical interface named m10, that will support an SCTP primary address; the second sequence configures an interface named m01 that will support a secondary SCTP address.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) # system
ACMEPACKET (system) # phy-interface
ACMEPACKET (phy-interface) # operation-type media
ACMEPACKET (phy-interface) # port 0
ACMEPACKET (phy-interface) # slot 1
ACMEPACKET (phy-interface) # name m10
ACMEPACKET (phy-interface) #
...
...
...
ACMEPACKET (phy-interface) #
ACMEPACKET# configure terminal
ACMEPACKET (configure) # system
ACMEPACKET (system) # phy-interface
ACMEPACKET (phy-interface) # operation-type media
ACMEPACKET (phy-interface) # port 1
ACMEPACKET (phy-interface) # slot 0
ACMEPACKET (phy-interface) # name m01
ACMEPACKET (phy-interface) #
...
...
...
ACMEPACKET (phy-interface) #
```

### Network Interface Configuration

These ACLI command sequences configure two network interfaces. The first sequence configures a network interface named m10, thus associating the network interface with the physical interface of the same name. The ACLI **ip-address** command assigns the IPv4 address 172.16.10.76 to the network interface. In a similar fashion, the second command sequence associates the m01 network and physical interfaces, and assigns an IPv4 address of 182.16.10.76.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) # system
ACMEPACKET (system) # network-interface
ACMEPACKET (network-interface) # name m10
ACMEPACKET (network-interface) # ip-address 172.16.10.76
...
...
...
ACMEPACKET (network-interface) #
ACMEPACKET# configure terminal
ACMEPACKET (configure) # system
ACMEPACKET (system) # network-interface
ACMEPACKET (network-interface) # name m01
ACMEPACKET (network-interface) # ip-address 182.16.10.76
...
...
...
ACMEPACKET (network-interface) #
```

## SIP Port Configuration

This CLI command sequence configures a SIP port for SCTP operations. It specifies the use of SCTP as the transport layer protocol, and assigns the existing network interface address, 172.16.10.76, as the SCTP primary address. Additionally, it identifies three other existing network addresses (182.16.10.76, 192.16.10.76, and 196.15.32.108) as SCTP secondary addresses.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-interface) # sip-ports
ACMEPACKET (sip-port) # address 172.16.10.76
ACMEPACKET (sip-port) # transport-protocol sctp
ACMEPACKET (sip-port) # multi-homed-addr (182.16.10.76 192.16.10.76
196.15.32.108)
...
...
...
ACMEPACKET (sip-port) #
```

## Realm Configuration

These CLI command sequences configure a realm for SCTP operations. The first CLI sequence assigns a named realm, in this example core-172, to a SIP interface during the interface configuration process. The second sequence accesses the target realm and uses the **network-interfaces** command to associate the named SCTP network interfaces with the realm.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-interface) # realm-id core-172
...
...
...
ACMEPACKET (sip-interface) #
ACMEPACKET# configure terminal
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) # select
identifier: core-172
1. core-172 ...
selection: 1
ACMEPACKET (realm-config) # network-interfaces (m01 m10 ...)
...
...
...
ACMEPACKET (ream-config) #
```

## Session Agent Configuration

The final CLI command sequence enables an SCTP-based transport connection between the Acme Packet 4500 and an adjacent network element.


```
ACMEPACKET# configure terminal
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) # session-agent
ACMEPACKET (session-agent) # select
<hostname>: core-172S1
1. core-172S1 ...
selection: 1
ACMEPACKET (session-agent) #
ACMEPACKET (session-agent) # transport-method staticSCTP
ACMEPACKET (session-agent) # reuse-connections none
...
```

```
...  
...  
ACMEPACKET (session-agent) #
```

## IPv6 Address Configuration

IPv6 can be a licensed feature on the Oracle Communications Session Border Controller. If you want to add this license to a system, then contact your Acme Packet sales engineering for information related to the license. Once you have the license information, refer to the Getting Started chapter for instructions about how to add a license.

You do not need to take action if you are working with a new system with which the IPv6 license was purchased.

 **Note:** For CLI parameters that support only IPv4, there are many references to that version as the accepted value for a configuration parameter or other IPv4-specific languages. For IPv6 support, these references have been edited. For example, rather than providing help that refers specifically to IPv4 addresses when explaining what values are accepted in an CLI configuration parameter, you will now see an <ipAddr> note.

This section calls out the configurations and parameters for which you can enter IPv6 addresses. In this first IPv6 implementation, the complete range of system configurations and their parameters are available for IPv6 use.

The Oracle Communications Session Border Controller follows RFC 3513 its definition of IPv6 address representations. Quoting from that RFC, these are the two forms supported:

- The preferred form is x:x:x:x:x:x:x, where the 'x's are the hexadecimal values of the eight 16-bit pieces of the address. Examples:

```
FEDC:BA98:7654:3210:FEDC:BA98:7654:3210
```

```
1080:0:0:0:8:800:200C:417A
```

Note that it is not necessary to write the leading zeros in an individual field, but there must be at least one numeral in every field (except for the case described in 2.).

- Due to some methods of allocating certain styles of IPv6 addresses, it will be common for addresses to contain long strings of zero bits. In order to make writing addresses containing zero bits easier a special syntax is available to compress the zeros. The use of ":" indicates one or more groups of 16 bits of zeros. The ":" can only appear once in an address. The "::" can also be used to compress leading or trailing zeros in an address. For example, the following addresses: 1080:0:0:0:8:800:200C:417A a unicast address FF01:0:0:0:0:0:101 a multicast address

0:0:0:0:0:0:1 the loopback address

0:0:0:0:0:0:0 the unspecified addresses

may be represented as:

1080::8:800:200C:417A a unicast address

FF01::101 a multicast address

::1 the loopback address

:: the unspecified addresses

## Access Control

These are the IPv6-enabled parameters in the **access-control** configuration.

Parameter	Entry Format
source-address	<ip-address>[/<num-bits>][:<port>[/<port-bits>]]
destination-address	<ip-address>[/<num-bits>][:<port>[/<port-bits>]]

## Host Route

These are the IPv6-enabled parameters in the **host-route** configuration.

Parameter	Entry Format
dest-network	<ipv4>   <ipv6>
netmask	<ipv4>   <ipv6>
gateway	<ipv4>   <ipv6>

## Local Policy

These are the IPv6-enabled parameters in the **local-policy** configuration.

Parameter	Entry Format
from-address	<ipv4>   <ipv6>   POTS Number, E.164 Number, hostname, wildcard
to-address	<ipv4>   <ipv6>   POTS Number, E.164 Number, hostname, wildcard

## Network Interface

These are the IPv6-enabled parameters in the **network-interface** configuration.

Parameter	Entry Format
hostname	<ipv4>   <ipv6>   hostname
ip-address	<ipv4>   <ipv6>
pri-utility-addr	<ipv4>   <ipv6>
sec-utility-addr	<ipv4>   <ipv6>
netmask	<ipv4>   <ipv6>
gateway	<ipv4>   <ipv6>
sec-gateway	<ipv4>   <ipv6>
dns-ip-primary	<ipv4>   <ipv6>
dns-ip-backup1	<ipv4>   <ipv6>
dns-ip-backup2	<ipv4>   <ipv6>
add-hip-ip	<ipv4>   <ipv6>
remove-hip-ip	<ipv4>   <ipv6>
add-icmp-ip	<ipv4>   <ipv6>
remove-icmp-ip	<ipv4>   <ipv6>

## ENUM Server

These are the IPv6-enabled parameters in the enum-config.

Parameter	Entry Format
enum-servers	[<ipv4>   <ipv6>]:port

## System Configuration

---

### Realm Configuration

These are the IPv6-enabled parameters in the **realm-config**.

Parameter	Entry Format
addr-prefix	[<ipv4>   <ipv6>]/prefix

### Session Agent

These are the IPv6-enabled parameters in the **session-agent** configuration.

Parameter	Entry Format
hostname	<ipv4>   <ipv6>
ip-address	<ipv4>   <ipv6>

### SIP Configuration

These are the IPv6-enabled parameters in the **session-config**.

Parameter	Entry Format
registrar-host	<ipv4>   <ipv6>   hostname   *

### SIP Interface SIP Ports

These are the IPv6-enabled parameters in the **sip-interface>sip-ports** configuration.

Parameter	Entry Format
address	<ipv4>   <ipv6>

### Steering Pool

These are the IPv6-enabled parameters in the **steering-pool** configuration.

Parameter	Entry Format
ip-address	<ipv4>   <ipv6>

### System Configuration

These are the IPv6-enabled parameters in the **system-config**.

Parameter	Entry Format
default-v6-gateway	<ipv6>

## IPv6 Support for Management and Telemetry

---

Several management-oriented parameters on the Oracle Communications Session Border Controller may be configured with IPv6 addresses to be used within IPv6 networks.

The following parameters that are configured with IP addresses accept IPv6 addresses to be used within IPv6 address space.

You may configure the wancom0/eth0 physical interface in the bootparams with an IPv6 address and complementary IPv6 gateway via the following parameters:



- **bootparams > inet on ethernet**
- **bootparams > gateway inet**

You may configure a syslog server with an IPv6 destination address via the following parameter:

- **system > system-config > syslog-servers > address**

You may configure a system access list entry with an IPv6 source address and complementary IPv6 gateway.

- **system > system-access-list > source-address**
- **system > system-access-list > netmask**

You may configure a RADIUS server with an IPv6 destination address via the following parameter:

- **security > authentication > radius-servers > address**

## IPv6 Default Gateway

In the system configuration, you configure a default gateway—a parameter that now has its own IPv6 equivalent.

To configure an IPv6 default gateway:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **system** and press Enter.

```
ACMEPACKET (configure) # system
ACMEPACKET (system) #
```

3. Type **system-config** and press Enter.

```
ACMEPACKET (system) # system-config
ACMEPACKET (system-config) #
```

4. **default-v6-gateway**—Set the IPv6 default gateway for this Oracle Communications Session Border Controller. This is the IPv6 egress gateway for traffic without an explicit destination. The application of your Oracle Communications Session Border Controller determines the configuration of this parameter.

5. Save your work.

## IPv6 Link Local Addresses

The Oracle Communications Session Border Controller supports IPv6 Link Local addresses configured for a network interface's gateway.

An IPv6 link local address is signified by its first hextet set to FE80:. Even if a network interface's first hextet is not FE80, but the gateway is, the Oracle Communications Session Border Controller will still function as expected.

### show neighbor-table

The show neighbor-table command displays the IPv6 neighbor table and validates that there is an entry for the link local address, and the gateway uses that MAC address.

```
System# show neighbor-table
LINK LEVEL NEIGHBOR TABLE
Neighbor                               Linklayer Address  Netif  Expire    S
Flags
300::100                               0:8:25:a1:ab:43    sp0    permanent ? R
871962224
400::100                               0:8:25:a1:ab:45    sp1    permanent ? R
871962516
fe80::bc02:a98f:f61e:20%sp0           be:2:ac:1e:0:20    sp0    4s        ? R
```

## System Configuration

```
871962808
fe80::bc01:a98f:f61e:20%sp1          be:1:ac:1e:0:20          sp1 4s          ? R
871963100
```

-----  
ICMPv6 Neighbor Table:  
-----

entry:	slot	port	vlan	IP	type	flag
pendBlk	Hit	MAC				
5	:	1	0	0	fe80::bc01:a98f:f61e:20/64	08-DYNAMIC 1
0		1			be:01:ac:1e:00:20	
4	:	1	0	0	0.0.0.0/64	01-GATEWAY 0
0		1			be:01:ac:1e:00:20	
3	:	1	0	0	400::/64	02-NETWORK 0
0		1			00:00:00:00:00:00	
2	:	0	0	0	fe80::bc02:a98f:f61e:20/64	08-DYNAMIC 1
0		1			be:02:ac:1e:00:20	
1	:	0	0	0	0.0.0.0/64	01-GATEWAY 0
0		1			be:02:ac:1e:00:20	
0	:	0	0	0	300::/64	02-NETWORK 0
0		1			00:00:00:00:00:00	

## Network Interfaces and IPv6

You set many IP addresses in the network interface, one of which is the specific IP address for that network interface and others that are related to different types of management traffic. This section outlines rules you must follow for these entries.

- For the **network-interface ip-address** parameter, you can set a single IP address. When you are working with an IPv6-enabled system, however, note that all other addresses related to that network-interface IP address must be of the same version.
- Heterogeneous address family configuration is prevented for the **dns-ip-primary**, **dns-ip-backup1**, and **dns-ip-backup2** parameters.
- For HIP addresses (**add-hip-ip**), you can use either IPv4 or IPv6 entries.
- For ICMP addresses (**add-icmp-ip**), you can use either IPv4 or IPv6 entries.
- For Telnet (**add-telnet-ip**), FTP (**add-ftp-ip**), and SNMP (**add-snmp-ip**), you are not allowed to use IPv6; your entries MUST use IPv4.

## IPv6 Reassembly and Fragmentation Support

As it does for IPv4, the Oracle Communications Session Border Controller supports reassembly and fragmentation for large signaling packets when you enable IPV6 on your system.

The Oracle Communications Session Border Controller takes incoming fragments and stores them until it receives the first fragment containing a Layer 4 header. With that header information, the Oracle Communications Session Border Controller performs a look-up so it can forward the packets to its application layer. Then the packets are re-assembled at the applications layer. Media fragments, however, are not reassembled and are instead forwarded to the egress interface.

On the egress side, the Oracle Communications Session Border Controller takes large signaling messages and encodes it into fragment datagrams before it transmits them.

Note that large SIP INVITE messages should be sent over TCP. If you want to modify that behavior, you can use the SIP interface's option parameter **max-udp-length=xx** for each SIP interface where you expect to receive large INVITE packets.

Other than enabling IPv6 on your Oracle Communications Session Border Controller, there is no configuration for IPv6 reassembly and fragmentation support. It is enabled automatically.

## Access Control List Support

The Oracle Communications Session Border Controller supports IPv6 for access control lists in two ways:

- For static access control lists that you configure in the **access-control** configuration, your entries can follow IPv6 form. Further, this configuration supports a prefix that enables wildcarding the source IP address.
- Dynamic ACLs are also supported; the Oracle Communications Session Border Controller will create ACLs for offending IPv6 endpoints.

## Data Entry

When you set the **source-address** and **destination-address** parameters in the **access-control** configuration, you will use a slightly different format for IPv6 than for IPv4.

For the **source-address**, your IPv4 entry takes the following format: <ip-address>[/<num-bits>][:<port>[/<port-bits>]]. And for the **destination-address**, your IPv4 entry takes this format: <ip-address>[:<port>[/<port-bits>]].

Since the colon (:) in the IPv4 format leads to ambiguity in IPv6, your IPv6 entries for these settings must have the address encased in brackets ([]): [7777::11]/64:5000/14.

In addition, IPv6 entries are allowed up to 128 bits for their prefix lengths.

The following is an example access control configuration set up with IPv6 addresses.

```
ACMEPACKET (access-control) # done
access-control
    realm-id                net7777
    description
    source-address          7777::11/64:5060/8
    destination-address     8888::11:5060/8
    application-protocol   SIP
    transport-protocol     ALL
    access                  deny
    average-rate-limit     0
    trust-level             none
    minimum-reserved-bandwidth 0
    invalid-signal-threshold 10
    maximum-signal-threshold 0
    untrusted-signal-threshold 0
    deny-period             30
```

## DNS Support

The Oracle Communications Session Border Controller supports the DNS resolution of IPv6 addresses; in other words, it can request the AAAA record type (per RFC 1886) in DNS requests. In addition, the Oracle Communications Session Border Controller can make DNS requests over IPv6 transport so that it can operate in networks that host IPv6 DNS servers.

For mixed IPv4-IPv6 networks, the Oracle Communications Session Border Controller follows these rules:

- If the realm associated with the name resolution is an IPv6 realm, the Oracle Communications Session Border Controller will send the query out using the AAAA record type.

## System Configuration

---

- If the realm associated with the name resolution is an IPv4 realm, the Oracle Communications Session Border Controller will send the query out using the A record type.

In addition, heterogeneous address family configuration is prevented for the `dns-ip-primary`, `dns-ip-backup1`, and `dns-ip-backup2` parameters.

## Homogeneous Realms

---

IPv6 is supported for realms and for nested realms, as long as the parent chain remains within the same address family. If you try to configure realms with mixed IPv4-IPv6 addressing, your system will issue an error message when you try to save your configuration. This check saves you time because you do not have to wait to run a configuration verification (using the CLI `verify-config` command) to find possible errors.

### Parent-Child Network Interface Mismatch

Your system will issue the following error message if parent-child realms are on different network interfaces that belong to different address families:

```
ERROR: realm-config [child] and parent [net8888] are on network interfaces
that belong to different address families
```

### Address Prefix-Network Interface Mismatch

If the address family and the address-prefix you configure for the realm does not match the address family of its network interface, your system will issue the following error message:

```
ERROR: realm-config [child] address prefix and network interface [1:1:0]
belong to different address families
```

## RADIUS Support for IPv6

---

The Oracle Communications Session Border Controller's RADIUS support now includes:

- RADIUS CDR generation for SIPv6-SIPv6 and SIPv6-SIPv4 calls
- IPv6-based addresses in RADIUS CDR attributes

This means that for the CDR attributes in existence prior to the introduction of IPv6 to the Acme Packet 4500 are mapped to the type `ipaddr`, which indicates four-byte field. The sixteen-byte requirement for IPv6 addresses is now supported, and there are a parallel set of attributes with the type `ipv6addr`. Attributes 155-170 are reserved for the IPv6 addresses.

NAS addresses use the number 95 to specify the NAS-IPV6-Address attribute. And local CDRs now contain IPv6 addresses.

### Supporting RADIUS VSAs

The following VSAs have been added to the Oracle RADIUS dictionary to support IPv6.

<code>Acme-Flow-In-Src-IPv6_Addr_FS1_F</code>	155	<code>ipv6addr</code>	Acme
<code>Acme-Flow-In-Dst-IPv6_Addr_FS1_F</code>	156	<code>ipv6addr</code>	Acme
<code>Acme-Flow-Out-Src-IPv6_Addr_FS1_F</code>	157	<code>ipv6addr</code>	Acme
<code>Acme-Flow-Out-Dst-IPv6_Addr_FS1_F</code>	158	<code>ipv6addr</code>	Acme
<code>Acme-Flow-In-Src-IPv6_Addr_FS1_R</code>	159	<code>ipv6addr</code>	Acme
<code>Acme-Flow-In-Dst-IPv6_Addr_FS1_R</code>	160	<code>ipv6addr</code>	Acme
<code>Acme-Flow-Out-Src-IPv6_Addr_FS1_R</code>	161	<code>ipv6addr</code>	Acme
<code>Acme-Flow-Out-Dst-IPv6_Addr_FS1_R</code>	162	<code>ipv6addr</code>	Acme
<code>Acme-Flow-In-Src-IPv6_Addr_FS2_F</code>	163	<code>ipv6addr</code>	Acme
<code>Acme-Flow-In-Dst-IPv6_Addr_FS2_F</code>	164	<code>ipv6addr</code>	Acme
<code>Acme-Flow-Out-Src-IPv6_Addr_FS2_F</code>	165	<code>ipv6addr</code>	Acme
<code>Acme-Flow-Out-Dst-IPv6_Addr_FS2_F</code>	166	<code>ipv6addr</code>	Acme

## System Configuration

---

Acme-Flow-In-Src-IPv6_Addr_FS2_R	167	ipv6addr	Acme
Acme-Flow-In-Dst-IPv6_Addr_FS2_R	168	ipv6addr	Acme
Acme-Flow-Out-Src-IPv6_Addr_FS2_R	169	ipv6addr	Acme
Acme-Flow-Out-Dst-IPv6_Addr_FS2_R	170	ipv6addr	Acme



---

## Realms and Nested Realms

This chapter explains how to configure realms and nested realms, and specialized media-related features.

A realm is a logical definition of a network or groups of networks made up in part by devices that provide real-time communication sessions comprised of signaling messages and possibly media flows. These network devices might be call agents, softswitches, SIP proxies, H.323 gatekeepers, IP PBXs, etc., that are statically defined by IPv4 addresses. These network devices might also be IPv4 endpoints: SIP phones, IADs, MAs, media gateways, etc., that are defined by an IPv4 address prefix.

Realms support bandwidth-based call admission control and QoS marking for media. They are the basis for defining egress and ingress traffic to the Oracle Communications Session Border Controller—which supports the Oracle Communications Session Border Controller’s topology hiding capabilities.

This chapter also explains how to configure media ports (steering pools). A steering pool exists within a realm and contains a range of ports that have a common address (for example, a target IPv4 address). The range of ports contained in the steering pool are used to steer media flows from one realm, through the Oracle Communications Session Border Controller, to another.

Finally, in this chapter you can learn about TOS/DiffServ functionality for realm-based packet marking by media type.

---

### Overview

Realms are a logical distinction representing routes (or groups of routes) reachable by the Oracle Communications Session Border Controller and what kinds of resources and special functions apply to those routes. Realms are used as a basis for determining ingress and egress associations to network interfaces, which can reside in different VPNs. The ingress realm is determined by the signaling interface on which traffic arrives. The egress realm is determined by the following:

- Routing policy—Where the egress realm is determined in the session agent configuration or external address of a SIP-NAT
- Realm-bridging—As applied in the SIP-NAT configuration and H.323 stack configurations
- Third-party routing/redirect (i.e., SIP redirect or H.323 LCF)

Realms also provide configuration support for denial of service (DoS)/access control list (ACL) functionality.

Realms can also be nested in order to form nested realm groups. Nested realms consist of separate realms that are arranged within a hierarchy to support network architectures that have separate backbone networks and VPNs for signaling and media. This chapter provides detailed information about nested realms after showing you how to configure realms on your Oracle Communications Session Border Controller.

### About Realms and Network Interfaces

All realms reference network interfaces on the Oracle Communications Session Border Controller. This reference is made when you configure a list of network interfaces in the realm configuration.

You configure a network interface to specify logical network interfaces that correspond existing physical interfaces on the Oracle Communications Session Border Controller. Configuring multiple network interfaces on a single physical interface creates a channelized physical interface, a VLAN. VLANs, in turn, allow you to reuse address space, segment traffic, and maximize bandwidth.

In order to reach the realms you configure, you need to assign them network interfaces. The values you set for the name and port in the network interface you select then indicate where the realm can be reached.

### About the SIP Home Realm

The realm configuration is also used to establish what is referred to as the SIP home realm. This is the realm where the Oracle Communications Session Border Controller's SIP proxy sits.

In peering configurations, the SIP home realm is the internal network of the SIP proxy. In backbone access configurations, the SIP home realm typically interfaces with the backbone connected network. In additions, the SIP home realm is usually exposed to the Internet in an HNT configuration.

Although you configure a SIP home realm in the realm configuration, it is specified as the home realm in the main SIP configuration by the home realm identifier parameter. Specifying the SIP home realm means that the Oracle Communications Session Border Controller's SIP proxy can be addressed directly by connected entities, but other connected network signaling receives layer 3 NAT treatment before reaching the internal SIP proxy.

### About Realms and Other Oracle Communications Session Border Controller Functions

Realms are referenced by other configurations in order to support this functionality across the protocols the Oracle Communications Session Border Controller supports and to make routing decisions. Other configurations' parameters that point to realms are:

- SIP configuration: home realm identifier, egress realm identifier
- SIP-NAT configuration: realm identifier
- H.323 stack configuration: realm identifier
- Session agent configuration: realm identifier
- Media manager: home realm identifier
- Steering ports: realm identifier
- Static flow: in realm identifier, out realm identifier

## Realms

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Realm configuration is divided into the following functional areas, and the steps for configuring each are set out in this chapter: identity and IP address prefix, realm interfaces, realm service profiles, QoS measurement, QoS marking, address translation profiles, and DNS server configuration.

### Before You Configure

Before you configure realms, you want to establish the physical and network interfaces with which the realm will be associated.

- Configure a physical interface to define the physical characteristics of the signaling line.
- Configure a network interface to define the network in which this realm is participating and optionally to create VLANs.

If you wish to use QoS, you should also determine if your Oracle Communications Session Border Controller is QoS enabled.



Remember that you will also use this realm in other configurations to accomplish the following:

- Set a signaling port or ports at which the Oracle Communications Session Border Controller listens for signaling messages.
- Configure sessions agents to point to ingress and egress signaling devices located in this realm in order to apply constraint for admission control.
- Configure session agents for defining trusted sources for accepting signaling messages.

## Realm Configuration

To access the realm configuration parameters in the ACLI:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

From this point, you can configure realm parameters. To view all realm configuration parameters, enter a **?** at the system prompt.

### Identity and IP Address Prefix

The first parameters you configure for a realm are its name (a unique identifier) and an IP address prefix and subnet mask.

The IP address and subnet mask establish a set of matching criteria for the realm, and distinguishes between realms that you assign to the same network interface.

To configure a realm's identity and IP address prefix in the ACLI:


1. **identifier**—Enter the name of the realm. This parameter uniquely identifies the realm. You will use this parameter in other configurations when asked for a realm identifier value.
2. **addr-prefix**—Enter the IPv4 or IPv6 address and subnet mask combination to set the criteria the Oracle Communications Session Border Controller uses to match packets sent or received on the network interface associated with this realm. This matching determines the realm, and subsequently what resources are used for that traffic. This setting determines whether the realm is an IPv4 or IPv6 realm.

This parameter must be entered in the correct format where the IPv4 or IPv6 address comes first and is separated by a slash (/) from the subnet mask value. For example, 172.16.0.0/24.

The default for this parameter is **0.0.0.0**. When you leave this parameter set to the default, all addresses match.

### Realm Interfaces

The realm points to one network interface on the Oracle Communications Session Border Controller.

 **Note:** Only one network-interface can be assigned to a single realm-config object, except for Local multi-homing SCTP deployments.

To assign interfaces to a realm:

**network-interfaces**—Enter the physical and network interface(s) that you want this realm to reference. These are the network interfaces through which this realm can be reached by ingress traffic, and through which this traffic exits the system as egress traffic.

Enter the name and port in the correct format where the name of the interface comes first and is separated by a colon (:) from the port number. For example, f10:0.

## Realms and Nested Realms

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The parameters you set for the network interfaces must be unique.

Enter multiple network interfaces for this list by typing an open parenthesis, entering each field value separated by a Space, typing a closed parenthesis, and then pressing Enter.

```
ACMEPACKET (realm-config) # network-interfaces fe1:0
```

You must explicitly configure a realm's network interface as either IPv4 or IPv6 when the applicable interface is either dual-stack or IPv6. You do this by appending the realm's network-interface with a **.4** or a **.6**, as shown below.

```
ACMEPACKET (realm-config) # network-interfaces fe1:0.6
```

For single-stack interface configurations that do not specify this format, the Oracle Communications Session Border Controller assumes an IPv4 interface. Dual stack interface configurations fail if this IP version family suffix is not specified.

### Realm Service Profile

The parameters you configure to establish the realm service profile determine how bandwidth resources are used and how media is treated in relation to the realm. Bandwidth constraints set for realm service profiles support the Oracle Communications Session Border Controller's admission control feature.

Peer-to-peer media between endpoints can be treated in one of three different ways:

- Media can be directed between sources and destinations within this realm on this specific Oracle Communications Session Border Controller. Media travels through the Oracle Communications Session Border Controller rather than straight between the endpoints.
- Media can be directed through the Oracle Communications Session Border Controller between endpoints that are in different realms, but share the same subnet.
- For SIP only, media can be released between multiple Oracle Communications Session Border Controllers.

To enable SIP distributed media release, you must set the appropriate parameter in the realm configuration. You must also set the SIP options parameter to **media-release** with the appropriate header name and header parameter information. This option defines how the Oracle Communications Session Border Controller encodes IPv4 address and port information for media streams described by, for example, SDP.

To configure realm service profile:

1. **max-bandwidth**—Enter the total bandwidth budget in kilobits per second for all flows to/from the realm defined in this element. The default is **0** which allows for unlimited bandwidth. The valid range is:
  - Minimum—0
  - Maximum—4294967295
2. **mm-in-realm**—Enable this parameter to treat media within this realm on this Oracle Communications Session Border Controller. The default is **disabled**. Valid values are:
  - enabled | disabled
3. **mm-in-network**—Enable this parameter to treat media within realms that have the same subnet mask on this Oracle Communications Session Border Controller. The default is **enabled**. Valid values are:
  - enabled | disabled
4. **msm-release**—Enable or disable the inclusion of multi-system (multiple Oracle Communications Session Border Controllers) media release information in the SIP signaling request sent into the realm identified by this realm-config element. If this field is set to **enabled**, another Oracle Communications Session Border Controller is allowed to decode the encoded SIP signaling request message data sent from a SIP endpoint to another SIP endpoint in the same network to restore the original SDP and subsequently allow the media to flow directly between those two SIP endpoints in the same network serviced by multiple Oracle Communications Session Border Controllers. If this field is **disabled**, the media and signaling will pass through both Oracle Communications Session Border Controllers. Remember that for this feature to work, you must also set the options parameter in the SIP configuration accordingly. The default is **disabled**. Valid values are:

- enabled | disabled

## QoS Measurement

This chapter provides detailed information about when to configure the **qos-enable** parameter. If you are not using QoS or a QoS-capable Oracle Communications Session Border Controller, then you can leave this parameter set to disabled (default).

## QoS Marking

QoS marking allows you to apply a set of TOS/DiffServ mechanisms that enable you to provide better service for selected networks

You can configure a realm to perform realm-based packet marking by media type, either audio/voice or video.

The realm configuration references a set of media policies that you configure in the media policy configuration. Within these policies, you can establish TOS/DiffServ values that define an individual type (or class) of service, and then apply them on a per-realm basis. In the media profiles, you can also specify:

- One or more audio media types for SIP and/or H.323
- One or more video types for SIP and/or H.323
- Both audio and video media types for SIP and/or H.323

To establish what media policies to use per realm in the ACLI:

**media-policy**—Enter the name (unique identifier) of the media policy you want to apply in the realm. When the Oracle Communications Session Border Controller first sets up a SIP or H.323 media session, it identifies the egress realm of each flow and then determines the media-policy element to apply to the flow. This parameter must correspond to a valid name entry in a media policy element. If you leave this parameter empty, then QoS marking for media will not be performed for this realm.

## Address Translation Profiles

If you are not using this feature, you can leave the **in-translationid** and **out-translationid** parameters blank.

## DNS Servers

You can configure DNS functionality on a per-network-interface basis, or you can configure DNS servers to use per realm. Configuring DNS servers for your realms means that you can have multiple DNS servers in connected networks. In addition, this allows you to specify which DNS server to use for a given realm such that the DNS might actually be in a different realm with a different network interface.

This feature is available for SIP only.

To configure realm-specific DNS in the ACLI:

**dns-realm**—Enter the name of the network interface that is configured for the DNS service you want to apply in this realm. If you do not configure this parameter, then the realm will use the DNS information configured in its associated network interface.

## DoS ACL Configuration

If you are not using this functionality, you can leave the parameters at their default values: **average-rate-limit**, **peak-rate-limit**, **maximum-burst-size**, **access-control-trust-level**, **invalid-signal-threshold**, and **maximum-signal-threshold**.

## Enabling RTP-RTCP UDP Checksum Generation

You can configure the Oracle Communications Session Border Controller to generate a UDP checksum for RTP/RTCP packets on a per-realm basis. This feature is useful in cases where devices performing network address

## Realms and Nested Realms

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translation (NAT) do not pass through packets with a zero checksum from the public Internet. These packets do not make it through the NAT, even if they have the correct to and from IP address and UDP port information. When you enable this feature, the Oracle Communications Session Border Controller calculates a checksum for these packets and thereby enables them to traverse a NAT successfully.

If you do not enable this feature, then the Oracle Communications Session Border Controller will not generate a checksum for RTP or RTCP packets if their originator did not include one. If a checksum is already present when the traffic arrives at the hardware, the system will relay it.

You enable this feature on the outbound realm.

## Aggregate Session Constraints Per Realm

You can set session constraints for the Oracle Communications Session Border Controller's global SIP configuration, specified session agents, and specified SIP interfaces. This forces users who have a large group of remote agents to create a large number of session agents and SIP interfaces.

With this feature implemented, however, you can group remote agents into one or more realms on which to apply session constraints.

To enable sessions constraints on a per realm basis:

**constraint-name**—Enter the name of the constraint you want to use for this realm. You set up in the session-constraints configuration.

## UDP Checksum Generation Configuration

To enable UDP checksum generation for a realm:

**generate-udp-checksum**—Enable this parameter to generate a UDP checksum for this outbound realm. The default is **disabled**. Valid values are:

- enabled | disabled

This parameter is visible only on Acme Packet 3800s and Acme Packet 4500s that do not have an ETC card installed. The function is enabled and not configurable on all other platforms.

## Admission Control Configuration

You can set admission control based on bandwidth for each realm by setting the **max-bandwidth** parameter for the realm configuration. Details about admission control are covered in this guide's *Admission Control and QoS* chapter.

## Reserved Parameters

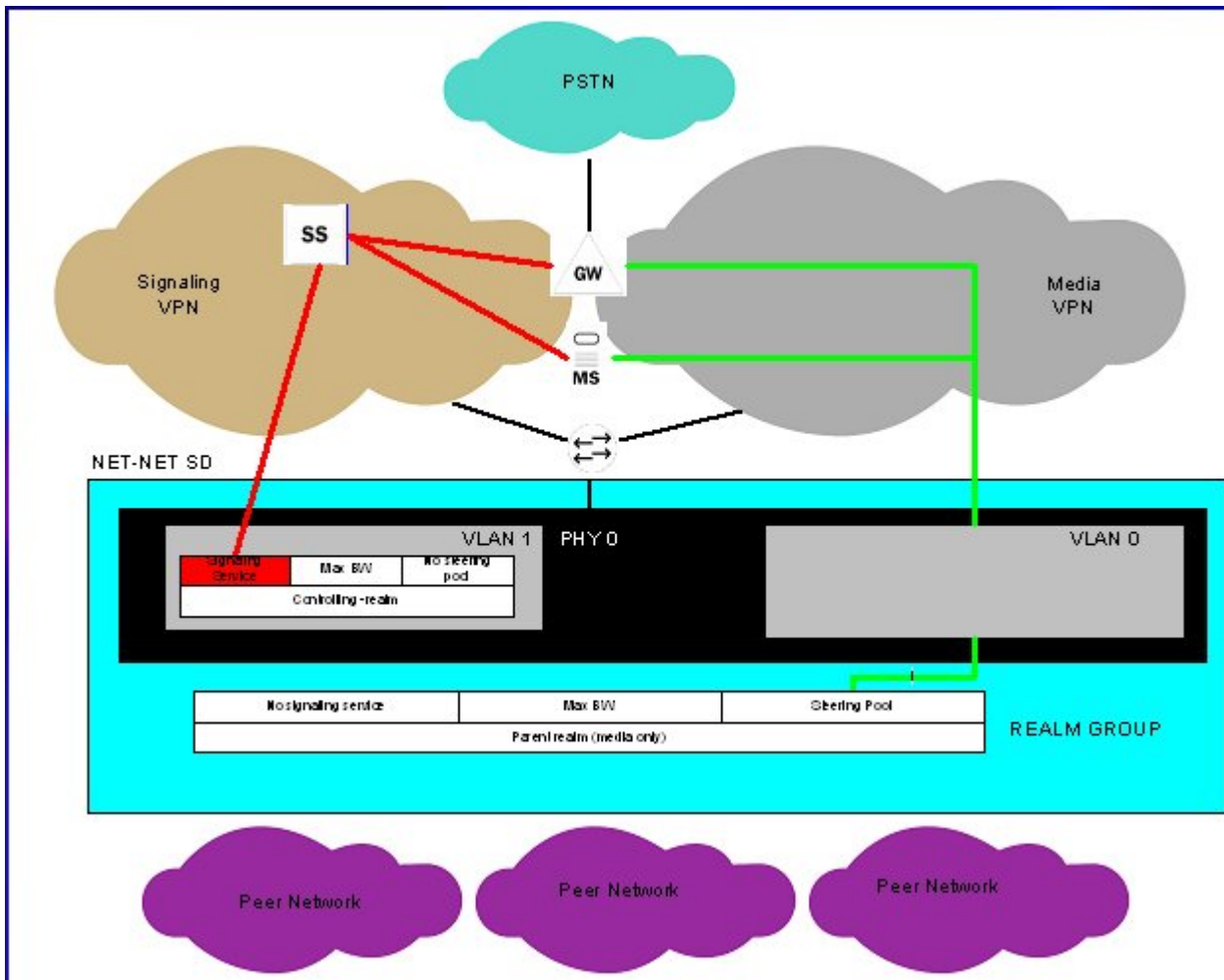
In the CLI, you do not need to configure the following parameters: **max-latency**, **max-jitter**, **max-packet-loss**, and **observ-window-size**.

## Nested Realms

---

Configuring nested realms allows you to create backbone VPN separation for signaling and media. This means that you can put signaling and media on separate network interfaces, that the signaling and media VPN can have different address spaces, and that the parent realm has one media-only sub-realm.

The following figure shows the network architecture.



In addition, you can achieve enhanced scalability by using a shared service interface. A single service address is shared across many customers/peers, customer specific policies for bandwidth use and access control are preserved, and you can achieve fine-grained policy control.

These benefits are achieved when you configure these types of realms:

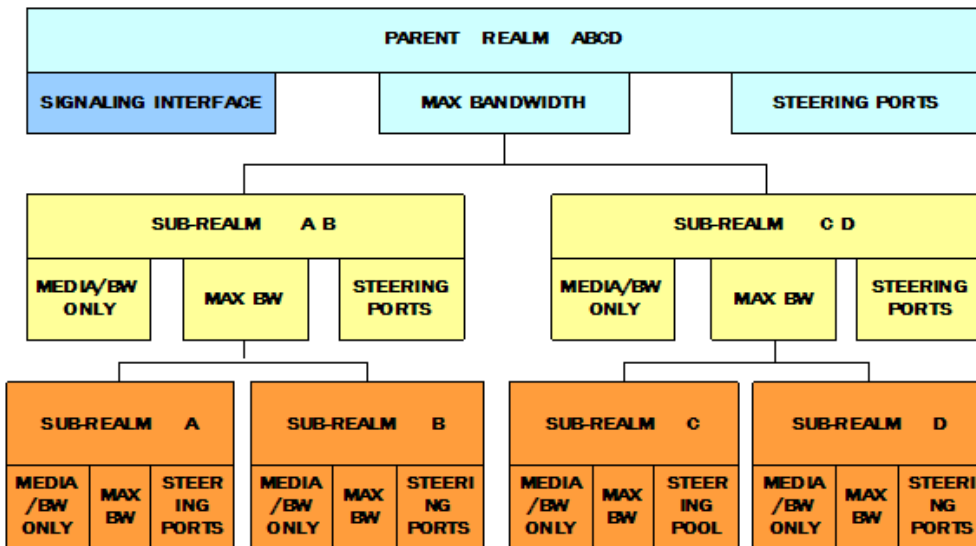
- Realm group—A hierarchical nesting of realms identified by the name of the highest order realm.
- Controlling realm—A realms for which a signaling interface is configured. For example, you might configure these signaling interfaces in the following configurations: SIP-NAT, SIP port or H.323 stack. Typically, this is the highest order realm for the parent realm in a realm group.
- Parent realm—A realm that has one or more child realms. A parent realm might also be the child realm of another realm group.
- Child realm—A realm that is associated with a single higher order parent realm. A child might also be the parent realm of another realm group. Child realms inherit all signaling and steering ports from higher order realms.
- Media-only realm—A realm for which there is no configured signaling interface directly associated. Media-only realms are nested within higher order realms.

As these definitions suggest, parent and child realms can be constructed so that there are multiple nesting levels. Lower order realms inherit the traits of the realms above them, including: signaling service interfaces, session translation tables, and steering pools.

Since realms inherit the traits of the realms above them in the hierarchy, you will probably want to map what realms should be parents and children before you start configuring them. These relationships are constructed through one parameter in the realm configuration that identifies the parent realm for the configuration. If you specify a parent realm, then the realm you are configuring becomes a child realm subject to the configured parameters you have

## Realms and Nested Realms

established for that parent. And since parent realms can themselves be children of other realm, it is important that you construct these relationships with care.



## Configuring Nested Realms

When you are configuring nested realms, you can separate signaling and media by setting realm parameters in the SIP interface configuration, the H.323 stack configuration, and the steering ports configuration.

- The realm identifier you set in the SIP interface configuration labels the associated realm for signaling.
- The realm identifier you set in the H.323 stack configuration labels the associated realm for signaling.
- The realm identifier you set in the steering ports configuration labels the associated realm for media.

Constructing a hierarchy of nested realms requires that you note which realms you want to handle signaling, and which you want to handle media.

In the SIP port configuration for the SIP interface and in the H.323 stack configuration, you will find an allow anonymous parameter that allows you to set certain access control measures. The table below outlines what each parameter means.

Allow Anonymous Parameter	Description
all	All anonymous connections allowed.
agents-only	Connections only allowed from configured session agents.
realm-prefix	Connections only allowed from addresses with the realm's address prefix and configured session agents.
registered	Connections allowed only from session agents and registered endpoints. (For SIP only, a REGISTER is allowed for any endpoint.)
register-prefix	Connections allowed only from session agent and registered endpoints. (For SIP only, a REGISTER is allowed for session agents and a matching realm prefix.)

## Parent and Child Realm Configuration

To configure nested realms, you need to set parameters in the realm configuration.

To configure parent and child realms:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

4. **parent-realm**—Enter the identifier of the realm you want to name as the parent. Configuring this parameter makes the realm you are currently configuring as the child of the parent you name. As such, the child realm is subject to the configured parameters for the parent.

### Required Signaling Service Parameters

To configure nested realms, you need to set parameters in the realm configuration and in the configurations for the signaling protocols you want to use.

To configure H.323 stack parameters for nested realms:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET # configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **h323** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # h323
ACMEPACKET (h323) #
```

4. Type **h323-stacks** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

From this point, you can configure H.323 stack parameters. To view all h323-stack configuration parameters, enter a **?** at the system prompt.

5. **allow-anonymous**—Enter the admission control of anonymous connections accepted and processed by this H.323 stack. The default is **all**. The valid values are:

- **all**—Allow all anonymous connections
- **agents-only**—Only requests from session agents allowed
- **realm-prefix**—Session agents and address matching realm prefix

### Aggregate Session Constraints Nested Realms

In addition to setting session constraints per realm for SIP and H.323 sessions, you can also enable the Oracle Communications Session Border Controller to apply session constraints across nested realms. When you set up session constraints for a realm, those constraints apply only to the realm for which they are configured without consideration for its relationship either as parent or child to any other realms.

You can also, however, enable the Oracle Communications Session Border Controller to take nested realms into consideration when applying constraints. For example, if a call enters on a realm that has no constraints but its parent does, then the constraints for the parent are applied. This parameter is global and so applies to all realms on the system. For the specific realm the call uses and for all of its parents, the Oracle Communications Session Border Controller increments the counters upon successful completion of an inbound or outbound call.

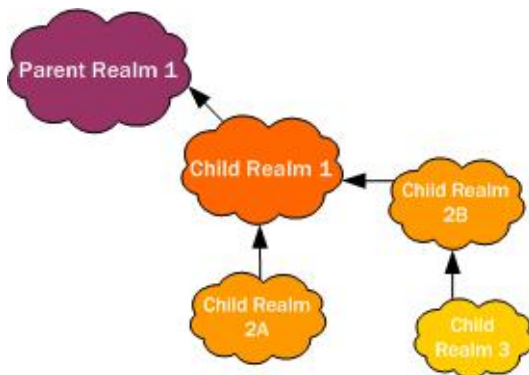
In the following example, you can see one parent realm and its multiple nested, child realms. Now consider applying these realm constraints:

- Parent Realm 1—55 active sessions
- Child Realm 1—45 active sessions
- Child Realm 2A—30 active sessions

## Realms and Nested Realms

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- Child Realm 2B—90 active sessions
- Child Realm 3—20 active sessions



Given the realm constraints outlined above, consider these examples of how global session constraints for realms. For example, a call enters the Oracle Communications Session Border Controller on Child Realm 2B, which has an unmet 90-session constraint set. Therefore, the Oracle Communications Session Border Controller allows the call based on Child Realm 2B. But the call also has to be within the constraints set for Child Realm 1 and Parent Realm 1. If the call fails to fall within the constraints for either of these two realms, then the Oracle Communications Session Border Controller rejects the call.

### Impact to Other Session Constraints and Emergency Calls

You can set up session constraints in different places in your Oracle Communications Session Border Controller configuration. Since session agents and SIP interfaces also take session constraints, it is important to remember the order in which the Oracle Communications Session Border Controller applies them:

1. Session agent session constraints
2. Realm session constraints (including parent realms)
3. SIP interface session constraints

Emergency and priority calls for each of these is exempt from session constraints. That is, any call coming into the Oracle Communications Session Border Controller marked priority is processed.

### Session Constraints Configuration

You enabled use of session constraints for nested realms across the entire system by setting the **nested-realms-stats** parameter in the session router configuration to **enabled**.

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-router** and press Enter.

```
ACMEPACKET(session-router)# session-router
ACMEPACKET(session-router-config)#
```

4. **nested-realms-stats**—Change this parameter from **disabled** (default) to **enabled** if you want the Oracle Communications Session Border Controller to apply session constraints across all nested realms (realms that are children to other realms)

5. Save and activate your configuration.



## Realm-Based Packet Marking

The Oracle Communications Session Border Controller supports TOS/DiffServ functions that allow you to

- Set up realm-based packet marking by media type, either audio-voice or video
- Set up realm-based packet marking for signaling, either SIP or H.323

Upstream devices use these markings to classify traffic in order to determine the priority level of treatment it will receive.

### About TOS DiffServ

TOS and DiffServ are two different mechanisms used to achieve QoS in enterprise and service provider networks; they are two different ways of marking traffic to indicate its priority to upstream devices in the network.

For more information about TOS (packet) marking, refer to:

- IETF RFC 1349 (<http://www.ietf.org/rfc/rfc1349.txt>)

For more information about DiffServ, refer to:

- IETF RFC 2474 (<http://www.ietf.org/rfc/rfc2474.txt>)
- IETF RFC 2475 (<http://www.ietf.org/rfc/rfc2475.txt>).

### ToS Byte

The TOS byte format is as follows:

Precedence			TOS				MBZ
0	1	2	3	4	5	6	7

The TOS byte is broken down into three components:

- Precedence—The most used component of the TOS byte, the precedence component is defined by three bits. There are eight possible precedence values ranging from 000 (decimal 0) through 111 (decimal 7). Generally, a precedence value of 000 refers to the lowest priority traffic, and a precedence value of 111 refers to the highest priority traffic.
- TOS—The TOS component is defined by four bits, although these bits are rarely used.
- MBZ—The must be zero (MBZ) component of the TOS byte is never used.


### DiffServ Byte

Given that the TOS byte was rarely used, the IETF redefined it and in doing so created the DiffServ byte.

The DiffServ byte format is as follows:

Precedence			TOS				MBZ
0	1	2	3	4	5	6	7

The DiffServ codepoint value is six bits long, compared to the three-bit-long TOS byte's precedence component. Given the increased bit length, DiffServ codepoints can range from 000000 (decimal 0) to 111111 (decimal 63).

 **Note:** By default, DiffServ codepoint mappings map exactly to the precedence component priorities of the original TOS byte specification.

### Packet Marking for Media

You can set the TOS/DiffServ values that define an individual type or class of service for a given realm. In addition, you can specify:

- One or more audio media types for SIP and/or H.323

- One or more video media types for SIP and/or H.323
- Both audio and video media types for SIP and/or H.323

For all incoming SIP and H.23 requests, the media type is determined by negotiation or by preferred codec. SIP media types are determined by the SDP, and H.323 media types are determined by the media specification transmitted during call setup.

## Configuring Packet Marking by Media Type

This section describes how to set up the media policy configuration that you need for this feature, and then how to apply it to a realm.

These are the CLI parameters that you set for the media policy:

```
name          media policy name
tos-settings  list of TOS settings
```

This is the CLI parameter that you set for the realm:

```
media-policy default media policy name
```

### Packet Marking Configuration

To set up a media policy configuration to mark audio-voice or video packets:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# media-manager
```

3. Type **media-policy** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# media-policy
ACMEPACKET(media-policy)#
```

From this point, you can configure media policy parameters. To view all configuration parameters for media profiles, enter a ? at the system prompt.

4. Type **media-policy** and press Enter.

```
ACMEPACKET(media-manager)# media-policy
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the CLI **select** command) the configuration you want to edit.

5. **name**—Create a reference name for this policy and press Enter.

6. Type **tos-settings** and press Enter.

```
ACMEPACKET(media-policy)# tos-settings
```

7. **media-type**—Enter the media type that you want to use for this group of TOS settings. You can enter any of the IANA-defined media types for this value: audio, example, image, message, model, multipart, text, and video. This value is not case-sensitive and can be up to 255 characters in length; it has no default.

```
ACMEPACKET(tos-settings)# media-type message
```

8. **media-sub-type**—Enter the media sub-type you want to use for the media type. This value can be any of the sub-types that IANA defines for a specific media type. This value is not case-sensitive and can be up to 255 characters in length; it has no default.

```
ACMEPACKET(tos-settings)# media-sub-type sip
```

9. **media-attributes**—Enter the media attribute that will match in the SDP. This parameter is a list, so you can enter more than one value. The values are case-sensitive and can be up to 255 characters in length. This parameter has no default.

If you enter more than one media attribute value in the list, then you must enclose your entry in quotation marks ().

```
ACMEPACKET(tos-settings) # media-attributes sendonly sendrecv
```

10. **tos-value**—Enter the TOS value you want applied for matching traffic. This value is a decimal or hexadecimal value. The valid range is:

- 0x00 to 0xFF.

```
ACMEPACKET(tos-settings) # tos-value 0xF0
```

11. Save and activate your configuration.

## Applying a Media Policy to a Realm

To apply a media policy to a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure) # media-manager
```

3. Type **realm** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager) # realm  
ACMEPACKET(realm) #
```

4. **media-policy**—Enter the unique name of the media policy you want to apply to this realm.

## Signaling Packet Marking Configuration

ToS marking for signaling requires you to configure a media policy and set the name of the media policy in the appropriate realm configuration.

This section shows you how to configure packet marking for signaling.

### Configuring a Media Policy for Signaling Packet Marking

To set up a media policy configuration to mark signaling packets:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure) # media-manager
```

3. Type **media-policy** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager) # media-policy  
ACMEPACKET(media-policy) #
```

From this point, you can configure media policy parameters. To view all media policy configuration parameters, enter a ? at the system prompt.

4. Type **media-policy** and press Enter.

```
ACMEPACKET(media-manager) # media-policy
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the ACLI **select** command) the configuration you want to edit.

5. **name**—Create a reference name for this policy and press Enter.

6. Type **tos-settings** and press Enter.

```
ACMEPACKET(media-policy) # tos-settings
```

## Realms and Nested Realms

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7. **media-type**—Enter the media type that you want to use for this group of TOS settings. You can enter any of the IANA-defined media types for this value: audio, example, image, message, model, multipart, text, and video. This value is not case-sensitive and can be up to 255 characters in length; it has no default.

```
ACMEPACKET(tos-settings) # media-type message
```

8. **media-sub-type**—Enter the media sub-type you want to use for the media type. This value can be any of the sub-types that IANA defines for a specific media type. This value is not case-sensitive and can be up to 255 characters in length; it has no default.

```
ACMEPACKET(tos-settings) # media-sub-type sip
```

9. **media-attributes**—Enter the media attribute that will match in the SDP. This parameter is a list, so you can enter more than one value. The values are case-sensitive and can be up to 255 characters in length. This parameter has no default.

If you enter more than one media attribute value in the list, then you must enclose your entry in quotation marks ().

```
ACMEPACKET(tos-settings) # media-attributes sendonly sendrecv
```

10. **tos-value**—Enter the TOS value you want applied for matching traffic. This value is a decimal or hexadecimal value. The valid range is:

- 0x00 to 0xFF.

```
ACMEPACKET(tos-settings) # tos-value 0xF0
```

11. Save and activate your configuration.

### Applying a Media Policy to a Realm

To apply a media policy to a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure) # media-manager
```

3. Type **realm** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager) # realm
```

```
ACMEPACKET(realm) #
```

4. **media-policy**—Enter the unique name of the media policy you want to apply to this realm.

## Using Class Profile for Packet Marking

Class profile provides an additional means of ToS marking, but only for limited circumstances. Use class-profile only if you are marking ToS on traffic destined for a specific To address, and when media-policy is not used on the same realm. Using media-policy for ToS marking is, by far, more common.

To configure a class profile, you prepare your desired media policy, create the class profile referencing the media policy and the To address, and set the name of the class profile in the appropriate realm configuration.

### Class Profile and Class Policy Configuration

This section shows you how to configure packet marking using a class profile.

To configure the class profile and class policy:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure) # session-router
```

3. Type **class-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router) # class-profile
ACMEPACKET(class-profile) #
```

4. Type **policy** and press Enter to begin configuring the class policy.

```
ACMEPACKET(class-profile) # policy
```

From this point, you can configure class policy parameters. To view all class policy configuration parameters, enter a ? at the system prompt.

5. **profile-name**—Enter the unique name of the class policy. When you apply a class profile to a realm configuration, you use this value.
6. **to-address**—Enter a list of addresses to match to incoming traffic for marking. You can use E.164 addresses, a host domain address, or use an asterisk (\*) to set all host domain addresses.
7. **media-policy**—Enter the name of the media policy you want to apply to this class policy.

### Applying a Class Policy to a Realm

To apply a class policy to a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure) # media-manager
```

3. Type **media-policy** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager) # realm
ACMEPACKET(realm) #
```

4. **class-profile**—Enter the name of the class profile to apply to this realm. This is the name you set in the **profile-name** parameter of the class-policy configuration.

## Differentiated Services for DNS and ENUM

The Oracle Communications Session Border Controller can mark DNS/ENUM packets with a configurable differentiated services code point (DSCP).

Certain service providers mandate support for Differentiated Services (DS) on all traffic streams exiting any network element. This mandate requires that network elements function as a DS marker — that is as a device that sets the Distributed Services Codepoint (DSCP) in the Differentiated Services field of the IP header.

Previous software releases provided the capabilities to mark standard SIP and Real-Time Protocol (RTP) messages. Release S-CX6.4.0M3 adds the capability to mark DNS and ENUM queries.

For basic information about DS, refer to:

- The *ACLI Configuration Guide Realm-Based Packet Marking* topic in the Realms and Nested Realms chapter, which provides information on currently supported DS marking of SIP and RTP packets
- *IETF RFC 2474, Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers* (<http://www.ietf.org/rfc/rfc2474.txt>)
- *IETF RFC 2475, An Architecture for Differentiated Services* (<http://www.ietf.org/rfc/rfc2475.txt>).

DS provides a mechanism to define and deliver multiple and unique service classifications that can be offered by a service provider. Specific service classifications are identified by a DSCP, essentially a numeric index. The DSCP maps to a per-hop-behavior (PHB) that defines an associated service class. PHBs are generally defined in terms of call admission controls, packet drop criteria, and queue admission algorithms. In theory, DS supports 64 distinct classifications. In practice, however, network offerings generally consist of a much smaller suite, which typically includes:

## Realms and Nested Realms

---

- Default PHB - required best effort service — defined in RFC 2474
- Expedited Forwarding (EF) PHB - low-loss, low-latency — defined in RFC 3246, An Expedited Forwarding PHB
- Assured Forwarding (AF) PHB - assured delivery within subscriber limits — defined in *RFC 2597, Assured Forwarding PHB Group*, and *RFC 3260, New Terminology and Clarifications for DiffServ*
- Class Selector PHBs - maintain compatibility with previous TOS (type of service) precedence usage — defined in RFC 2474

Marking of DNS and ENUM queries requires the creation of a Differentiated Services media policy, and the assignment of such a policy to a specific realm.

## Differentiated Services for DNS and ENUM Configuration

To create a Differentiated Services media policy:

1. From Superuser mode, use the following CLI command sequence to move to media-policy configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# media-policy
ACMEPACKET(media-policy)#
```

2. Use the required **name** parameter to provide a unique identifier for this media-policy instance.

```
ACMEPACKET(media-policy)# name diffServeDnsEnum
ACMEPACKET(media-policy)#
```

3. Use the **tos-settings** command to move to tos-settings configuration mode.

```
ACMEPACKET(media-policy)# tos-settings
ACMEPACKET(tos-settings)#
```

4. Use the required **media-type** parameter to identify the packet-type subject to Differentiated Services marking.

For DNS/ENUM packets, use `message/dns`.

```
ACMEPACKET(tos-settings)# media-type message/dns
ACMEPACKET(tos-settings)#
```

5. Use the required **tos-value** parameter to specify the 6-bit DSCP to be included in the Differentiated Services field of the IP header.

Specify the DSCP as an integer (within the range from 0 to 63). The DSCP can be expressed in either decimal or hexadecimal format.

```
ACMEPACKET(tos-settings)# tos-value 0x30
ACMEPACKET(tos-settings)#
```

6. Other displayed parameters, **media-attributes** and **media-sub-type**, can be safely ignored.

7. Use the **done** and **exit** commands to complete Differentiated Services media policy configuration and return to media-policy configuration-mode.

```
ACMEPACKET(tos-settings)# done
tos-settings
  media-type           message/dns
  media-sub-type
  tos-value            0x30
  media-attributes
ACMEPACKET(tos-settings)# exit
ACMEPACKET(media-policy)#
```

8. If necessary, repeat steps 2 through 7 to create additional Differentiated Services media policies.

9. Assign this media policy to the target realm via the **media-policy** parameter in a **realm-config** object.

## SIP-SDP DCSP Marking ToS Bit Manipulation

Used to indicate priority and type of requested service to devices in the network, type of service (TOS) information is included as a set of four-bit flags in the IP header. Each bit has a different purpose, and only one bit at a time can be set: There can be no combinations. Available network services are:

- Minimum delay—Used when latency is most important
- Maximum throughput—Used when the volume of transmitted data in any period of time is important
- Maximum reliability—Used when it is important to assure that data arrives at its destination without requiring retransmission
- Minimum cost—Used when it is most important to minimize data transmission costs

The Oracle Communications Session Border Controller's support for type of service (TOS) allows you to base classification on the media type as well as the media subtype. In prior releases, you can configure the Oracle Communications Session Border Controller to mark TOS bits on outgoing packets using a media policy. Supported media types include audio, video, application, data, image, text, and message; supported protocol types are H.225, H.245, and SIP. Note that, although H.225 and H.245 are not part of any IANA types, they are special cases (special subtypes) of message for the Oracle Communications Session Border Controller. When these criteria are met for an outgoing packet, the Oracle Communications Session Border Controller applies the TOS settings to the IP header. The augmented application of TOS takes matching on media type or protocol and expands it to match on media type, media-sub-type, and media attributes.

The new flexibility of this feature resolves issues when, for example, a customer needs to differentiate between TV-phone and video streaming. While both TV-phone and video streaming have the attribute "media=video," TV-phone streaming has "direction=sendrcv" prioritized at a high level and video has direction=sendonly or recvonly with middle level priority. The Oracle Communications Session Border Controller can provide the appropriate marking required to differentiate the types of traffic.

In the media policy, the **tos-values** parameter accepts values that allow you to create any media type combination allowed by IANA standards. This is a dynamic process because the Oracle Communications Session Border Controller generates matching criteria directly from messages.

The new configuration takes a media type value of any of these: audio, example, image, message, model, multipart, text, and video. It also takes a media sub-type of any value specified for the media type by IANA; however, support for T.38 must be entered exactly as **t.38** (rather than t38). Using these values, the Oracle Communications Session Border Controller creates a value Based on a combination of these values, the Oracle Communications Session Border Controller applies TOS settings.

You also configure the TOS value to be applied, and the media attributes you want to match.

You can have multiple groups of TOS settings for a media policy.

### ToS Bit Manipulation Configuration

This section provides instructions for how to configure TOS bit manipulation on your Oracle Communications Session Border Controller.

To configure TOS bit manipulation:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **media-policy** and press Enter.

```
ACMEPACKET(media-manager)# media-policy
```

## Realms and Nested Realms

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If you are adding support for this feature to a pre-existing configuration, then you must select (using the ACLI **select** command) the configuration you want to edit.

4. **name**—Create a reference name for this policy and press Enter.
5. Type **tos-settings** and press Enter.

```
ACMEPACKET(media-policy) # tos-settings
```

6. **media-type**—Enter the media type that you want to use for this group of TOS settings. You can enter any of the IANA-defined media types for this value: audio, example, image, message, model, multipart, text, and video. This value is not case-sensitive and can be up to 255 characters in length; it has no default.

```
ACMEPACKET(tos-settings) # media-type message
```

7. **media-sub-type**—Enter the media sub-type you want to use for the media type. This value can be any of the sub-types that IANA defines for a specific media type. This value is not case-sensitive and can be up to 255 characters in length; it has no default.

```
ACMEPACKET(tos-settings) # media-sub-type sip
```

8. **media-attributes**—Enter the media attribute that will match in the SDP. This parameter is a list, so you can enter more than one value. The values are case-sensitive and can be up to 255 characters in length. This parameter has no default.

If you enter more than one media attribute value in the list, then you must enclose your entry in quotation marks ().

```
ACMEPACKET(tos-settings) # media-attributes sendonly sendrecv
```

9. **tos-value**—Enter the TOS value you want applied for matching traffic. This value is a decimal or hexadecimal value. The valid range is:

- 0x00 to 0xFF.

```
ACMEPACKET(tos-settings) # tos-value 0xF0
```

10. Save and activate your configuration.

## SDP Alternate Connectivity

---

The Oracle Communications Session Border Controller can create an egress-side SDP offer containing both IPv4 and IPv6 media addresses via a mechanism which allows multiple IP addresses, of different address families (i.e., IPv4 & IPv6) in the same SDP offer. Our implementation is based on the RFC draft "draft-boucadair-mmusic-altc-09".

Each realm on the Oracle Communications Session Border Controller can be configured with an alternate family realm on which to receive media in the alt family realm parameter in the realm config. As deployed, one realm will be IPv4, and the alternate will be IPv6. The Oracle Communications Session Border Controller creates the outbound INVITE with IPv4 and IPv6 addresses to accept the media, each in an a=altc: line and each in its own realm. The IP addresses inserted into the a=altc: line are from the egress realm's and alt-realm-family realm's steering pools. Observe in the image how the red lines indicate the complementary, alternate realms.

You can configure the order in which the a=altc: lines appear in the SDP in the pref-address-type parameter in the realm-config. This parameter can be set to

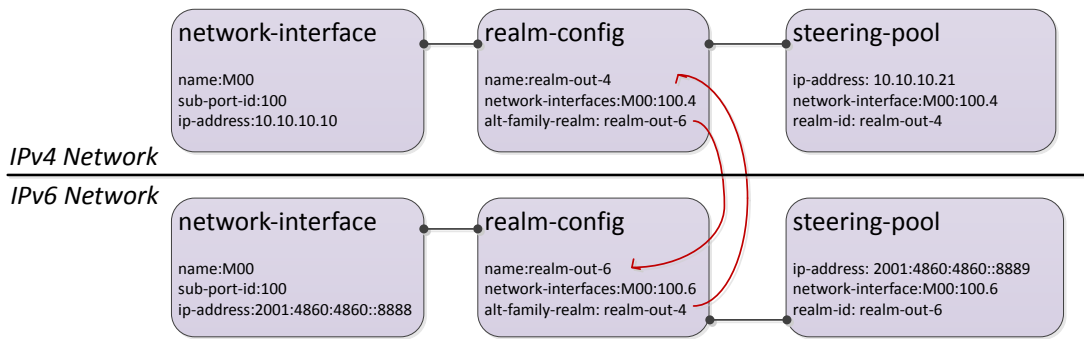
- IPv4 - SDP contains the IPv4 address first
- IPv6 - SDP contains the IPv6 address first
- NONE - SDP contains the native address family of the egress realm first

In the 200OK to the INVITE, the callee chooses either the IPv6 or IPv4 address to use for the call's media transport between itself and Oracle Communications Session Border Controller. After the Oracle Communications Session Border Controller receives the 200OK, the chosen flow is installed, and the unused socket is discarded.

For two realms from different address families to share the same physical interface and vlan, you use a .4 or .6 tag in the network-interface reference. When IPv4 and IPv6 realms share the same network-interface and VLAN, you identify them by realm name and network-interface configured as:



- IPv4 - <phy-interface>:<vlan>.4
- IPv6 - <phy-interface>:<vlan>.6



If the INVITE's egress realm is IPv6, pref-address-type = NONE, the outbound SDP has these a=altc: lines:

```
a=altc:1 IPv6 2001:4860:4860::8889 20001
a=altc:2 IPv4 10.10.10.21 20001
```

If the INVITE's egress realm is IPv6, pref-address-type = IPv4, the outbound SDP has these a=altc: lines:

```
a=altc:1 IPv4 10.10.10.21 20001
a=altc:2 IPv6 2001:4860:4860::8889 20001
```

SDP Alternate connectivity supports B2B and hairpin call scenarios. SDP Alternate connectivity also supports singleterm, B2B, and hairpin call scenarios.

When providing SDP alternate connectivity for SRTP traffic, in the security policy configuration element, the network-interface parameter's value must be configured with a .4 or .6 suffix to indicate IPv4 or IPv6 network, respectively.

## SDP Alternate Connectivity Configuration

To configure SDP alternate connectivity:

1. Access the **realm-config** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

2. Select the **realm-config** object to edit.

```
ACMEPACKET(realm-config)# select
identifier:
1: realm01 left-left:0 0.0.0.0


selection: 1
ACMEPACKET(realm-config)#
```

3. **alt-realm-family** — Enter the realm name of the alternate realm, from which to use an IP address in the other address family. If this parameter is within an IPv4 realm configuration, you will enter an IPv6 realm name.
4. **pref-address-type** — Set the order in which the a=altc: lines suggest preference. Valid values are:
  - none — address family type of egress realm signaling
  - ipv4 — IPv4 realm/address first
  - ipv6 — IPv6 realm/address first
5. Type **done** to save your configuration.

## Steering Pools

Steering pools define sets of ports that are used for steering media flows through the Oracle Communications Session Border Controller. These selected ports are used to modify the SDP to cause receiving session agents to direct their media toward this system. Media can be sent along the best quality path using these addresses and ports instead of traversing the shortest path or the BGP-4 path.

For example, when the Oracle Communications Session Border Controller is communicating with a SIP device in a specific realm defined by a steering pool, it uses the IP address and port number from the steering pool's range of ports to direct the media. The port the Oracle Communications Session Border Controller chooses to use is identified in the SDP part of the message.


 **Note:** The values entered in the steering pool are used when the system provides NAT, PAT, and VLAN translation.

### Configuration Overview

To plan steering pool ranges, take into account the total sessions available on the box, determine how many ports these sessions will use per media stream, and assign that number of ports to all of the steering pools on your Oracle Communications Session Border Controller. For example, if your Oracle Communications Session Border Controller can accommodate 500 sessions and each session typically uses 2 ports, you would assign 1000 ports to each steering pool. This strategy provides for a maximum number of ports for potential use, without using extra resources on ports your Oracle Communications Session Border Controller will never use.

The following table lists the steering pool parameters you need to configure:

Parameter	Description
IP address	IPv4 address of the steering pool.
start port	Port number that begins the range of ports available to the steering pool. You must define this port to enable the system to perform media steering and NAT operations.
end port	Port number that ends the range of ports available to the steering pool. You must define this port to enable the system to perform media steering and NAT operations.
realm id	Identifies the steering pool's realm. The steering pool is restricted to only the flows that originate from this realm.

 **Note:** The combination of entries for IP address, start port, and realm ID must be unique in each steering pool. You cannot use the same values for multiple steering pools.

Each bidirectional media stream in a session uses two steering ports, one in each realm (with the exception of audio/video calls that consume four ports). You can configure the start and end port values to provide admission control. If all of the ports in all of the steering pools defined for a given realm are in use, no additional flows/sessions can be established to/from the realm of the steering pool.

### Steering Pool Configuration

To configure a steering pool:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# media-manager
```

3. Type **steering-pool** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager) # steering-pool
ACMEPACKET(steering-pool) #
```

4. **ip-address**—Enter the target IPv4 address of the steering pool in IP address format. For example:

```
192.168.0.11
```

5. **start-port**—Enter the start port value that begins the range of ports available to this steering pool. The default is 0. The valid range is:

- Minimum—0
- Maximum—65535

You must enter a valid port number or the steering pool will not function properly.

6. **end-port**—Enter the end port value that ends the range of ports available to this steering pool. The default is 0. The valid range is:

- Minimum—0
- Maximum—65535

You must enter a valid port number or the steering pool will not function properly.

7. **realm-id**—Enter the realm ID to identify the steering pool’s realm, following the name format. The value you enter here must correspond to the value you entered as the identifier (name of the realm) when you configured the realm. For example:

```
peer-1
```

This steering pool is restricted to flows that originate from this realm.

The following example shows the configuration of a steering pool that

```
steering-pool
  ip-address          192.168.0.11
  start-port          20000
  end-port            21000
  realm-id            peer-1
  last-modified-date  2005-03-04 00:35:22
```

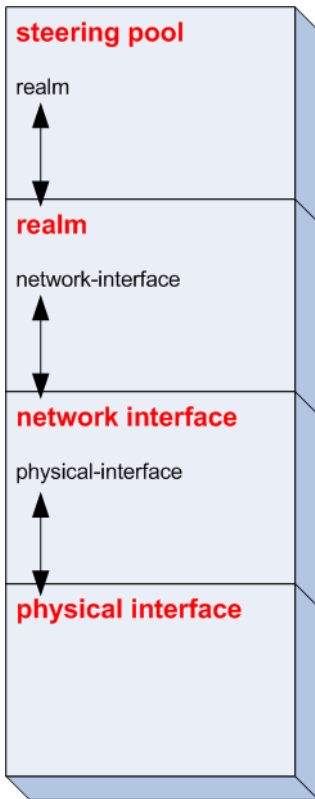
## Multiple Interface Realms

The multi-interface realm feature lets you group multiple network interfaces to aggregate their bandwidth for media flows. In effect, this feature lets you use the total throughput of the available physical interfaces on your Oracle Communications Session Border Controller for a single realm. Multi-interface realms are implemented by creating multiple steering pools, each on an individual network interface, that all reference a single realm.

Of course, you can not to use this feature and configure your Oracle Communications Session Border Controller to create a standard one-realm to one-network interface configuration.

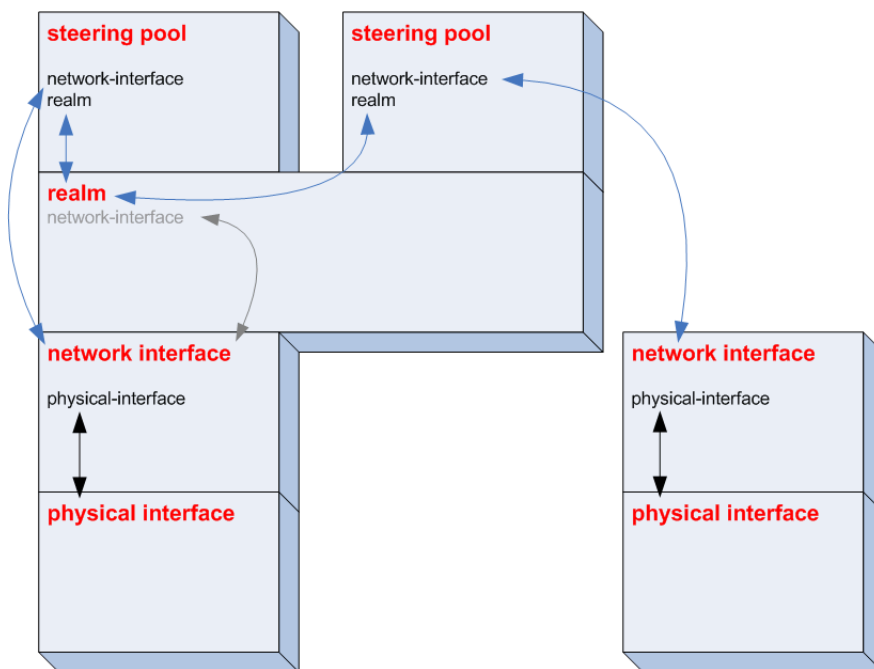
Without using multiple interface realms, the basic hierarchical configuration of the Oracle Communications Session Border Controller from the physical interface through the media steering pool looks like this:

## Realms and Nested Realms

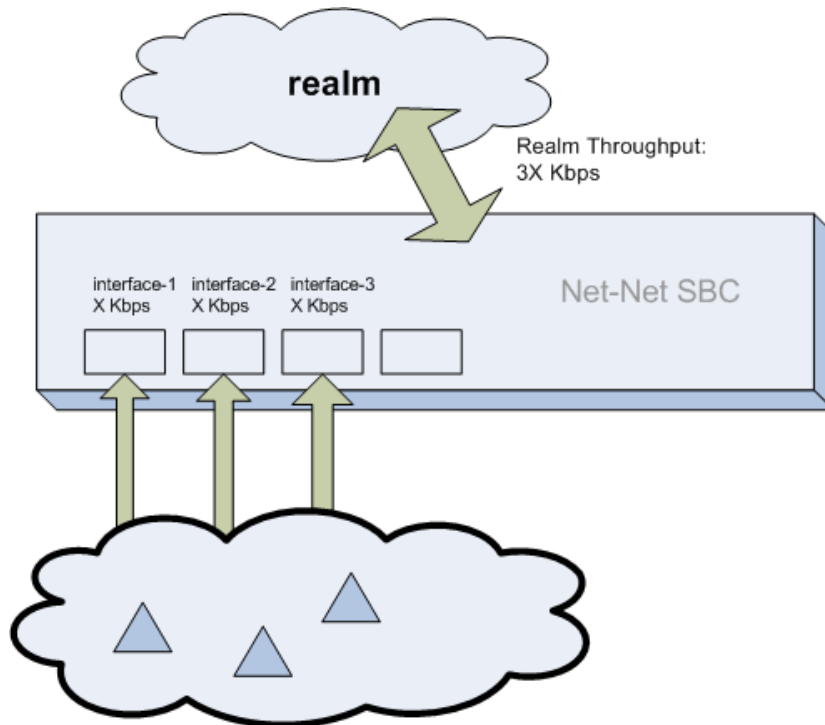


In this model, one (non-channelized) network interface exists on a physical interface. One realm exists on one network interface. One or more steering pools can exist on one realm. Within each higher level configuration element exists a parameter that references a lower level configuration element in the Oracle Communications Session Border Controller's logical network model.

The multi-interface realm feature directs media traffic entering and exiting multiple network interfaces in and out of a single realm. Since all the steering pools belong to the same realm, their assigned network interfaces all feed into the same realm as well. The following diagram shows the relationship in the new logical model:



The advantage of using multi-interface realms is the ability to aggregate the bandwidth available to multiple network interfaces for a larger-than-previously-available total bandwidth for a realm. In the illustration below, three physical interfaces each have X Kbps of bandwidth. The total bandwidth available to the realm with multiple network interfaces is now 3X the bandwidth. (In practical usage, interface-1 only contributes X - VoIP Signaling to the total media bandwidth available into the realm.)



### Steering Pool Port Allocation

Every steering pool you create includes its own range of ports for media flows. The total number of ports in all the steering pools that feed into one realm are available for calls in and out of the realm.

Steering pool ports for a given realm are assigned to media flows sequentially. When the first call enters the Oracle Communications Session Border Controller after start-up, it is assigned the first ports on the first steering pool that you configured. New calls are assigned to ports sequentially in the first steering pool. When all ports from the first steering pool are exhausted, the Oracle Communications Session Border Controller uses ports from the next configured steering pool. This continues until the last port on the last configured steering pool is used.

After the final port is used for the first time, the next port chosen is the one first returned as empty from the full list of ports in all the steering pools. As media flows are terminated, the ports they used are returned to the realm’s full steering pool. In this way, after initially exhausting all ports, the realm takes new, returned, ports from the pool in a least last used manner.

When a call enters the Oracle Communications Session Border Controller, the signaling application allocates a port from all of the eligible steering pools that will be used for the call. Once a port is chosen, the Oracle Communications Session Border Controller checks if the steering pool that the port is from has a defined network interface. If it does, the call is set up on the corresponding network interface. If a network interface is not defined for that steering pool, the network interface defined for the realm is used.

### Network Interface Configuration

This section explains how to configure your Oracle Communications Session Border Controller to use multiple interface realms.

You must first configure multiple physical interfaces and multiple network interfaces on your Oracle Communications Session Border Controller.

## Realms and Nested Realms

---

To configure the realm configuration for multi-interface realms.

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-manager path.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm-config** and press Enter. The system prompt changes.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

From this point, you can configure a realm that will span multiple network interfaces.

4. **network-interfaces**—Enter the name of the network interface where the signaling traffic for this realm will be received.

### Creating Steering Pools for Multiple Interface Realms

To configure steering pools for multi-interface realms:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-manager path.

```
ACMEPACKET (configure) # media-manager
```

3. Type **steering-pool** and press Enter. The system prompt changes.

```
ACMEPACKET (media-manager) # steering-pool
ACMEPACKET (steering-pool) #
```

From this point, you can configure steering pools which collectively bridge the multiple network interfaces they are connected to.

4. **ip-address**—Enter the IP address of the first steering pool on the first network interface.

This IP address must correspond to an IP address within the subnet of a network interface you have already configured.

This IP can not exist on a network interface other than the one you configure in the **network-interface** parameter.

5. **start-port**—Enter the beginning port number of the port range for this steering pool. The default is 0. The valid range is:

- Minimum—0
- Maximum—65535

6. **end-port**—Enter the ending port number of the port range for this steering pool. The default is 0. The valid range is:

- Minimum—0
- Maximum—65535

7. **realm-id**—Enter the name of the realm which this steering pool directs its media traffic toward.

8. **network-interface**—Enter the name of the network interface you want this steering pool to direct its media toward. This parameter will match a **name** parameter in the network-interface configuration element. If you do not configure this parameter, you can only assign a realm to a single network interface, as the behavior was in all Oracle Communications Session Border Controller Software releases pre- 2.1.

9. Create additional steering pools on this and on other network interfaces as needed. Remember to type **done** when you are finished configuring each new steering pool.

## Media over TCP

The Oracle Communications Session Border Controller now supports RFC 4145 (TCP-Based Media Transport in the SDP), also called TCP Bearer support. Media over TCP can be used to support applications that use TCP for bearer path transport.

RFC 4145 adds two new attributes, `setup` and `connection`, to SDP messages. The `setup` attribute indicates which end of the TCP connection should initiate the connection. The `connection` attribute indicates whether an existing TCP connection should be used or if a new TCP connection should be setup during re-negotiation. RFC 4145 follows the offer/answer model specified in RFC3264. An example of the SDP offer message from the end point 192.0.2.2 as per RFC4145 is as given below:

```
m=image 54111 TCP t38
c=IN IP4 192.0.2.2
a=setup:passive
a=connection:new
```

This offer message indicates the availability of t38 fax session at port 54111 which runs over TCP. Oracle Communications Session Border Controller does not take an active part in the application-layer communication between each endpoint.

The Oracle Communications Session Border Controller provides the means to set up the end-to-end TCP flow by creating the TCP/IP path based on the information learned in the SDP offer/answer process.

## TCP Bearer Conditions

The following conditions are applicable to the Oracle Communications Session Border Controller's support of RFC 4145.

1. The Oracle Communications Session Border Controller can not provide media-over-TCP for HNT scenarios (endpoints behind a NAT).
2. When media is released into the network, the TCP packets do not traverse the Oracle Communications Session Border Controller because no TCP bearer connection is created.
3. The Oracle Communications Session Border Controller does not inspect the `setup` and `connection` attributes in the SDP message since the TCP packets transparently pass through the Oracle Communications Session Border Controller. These SDP attributes are forwarded to the other endpoint. It is the other endpoint's responsibility to act accordingly.
4. After the Oracle Communications Session Border Controller receives a SYN packet, it acts as a pure pass through for that TCP connection and ignores all further TCP handshake messages including FIN and RST. The flow will only be torn down in the following instances:
  - The TCP initial guard timer, TCP subsequent guard timer, or the TCP flow time limit timer expire for that flow.
  - The whole SIP session is torn down.

## TCP Port Selection

When a call is first set up, the Oracle Communications Session Border Controller inspects the SDP message's m-line to see if any media will be transported via TCP. If the SDP message indicates that some content will use TCP, the Oracle Communications Session Border Controller allocates a configured number of steering ports for the media-over-TCP traffic. These TCP media ports are taken from the each realm's steering pool.

Each endpoint can initiate up to four end-to-end TCP flows between itself and the other endpoint. The Oracle Communications Session Border Controller assigns one port to receive the initial TCP packet (server port), and one to four ports assigned to send TCP traffic (client ports) to the receiving side of the TCP flow. The number of TCP flows for each call is configured globally.

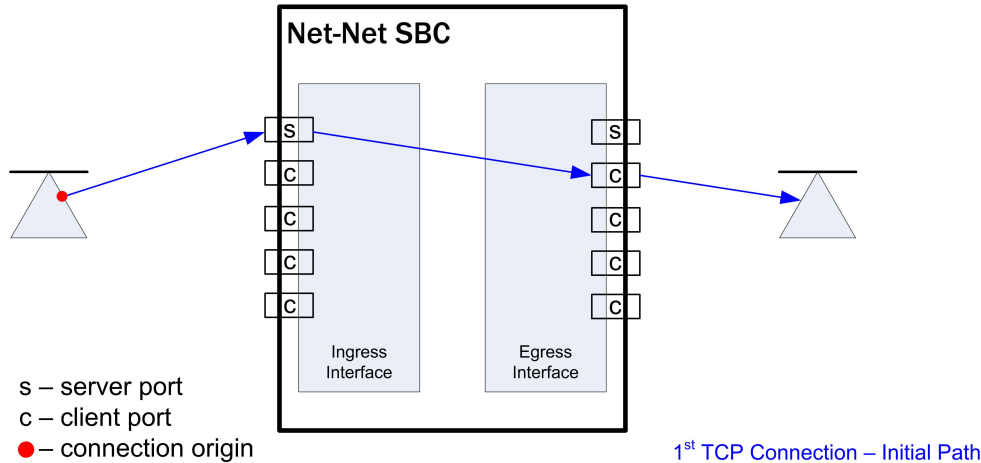
In order to configure the Oracle Communications Session Border Controller to facilitate and support this process, you need to specify the number of ports per side of the call that can transport discrete TCP flows. You can configure one

## Realms and Nested Realms

to four ports/flows. For configuration purposes, the Oracle Communications Session Border Controller counts this number as inclusive of the server port. Therefore if you want the Oracle Communications Session Border Controller to provide a maximum of one end-to-end TCP flow, you have to configure two TCP ports; one to receive, and one to send. The receiving port (server) is reused to set up every flow, but the sending port (client) is discrete per flow. For example: for 2 flows in each direction, set the configuration to 3 TCP ports per flow; for 3 flows in each direction, set the configuration to 4 TCP ports per flow, etc.

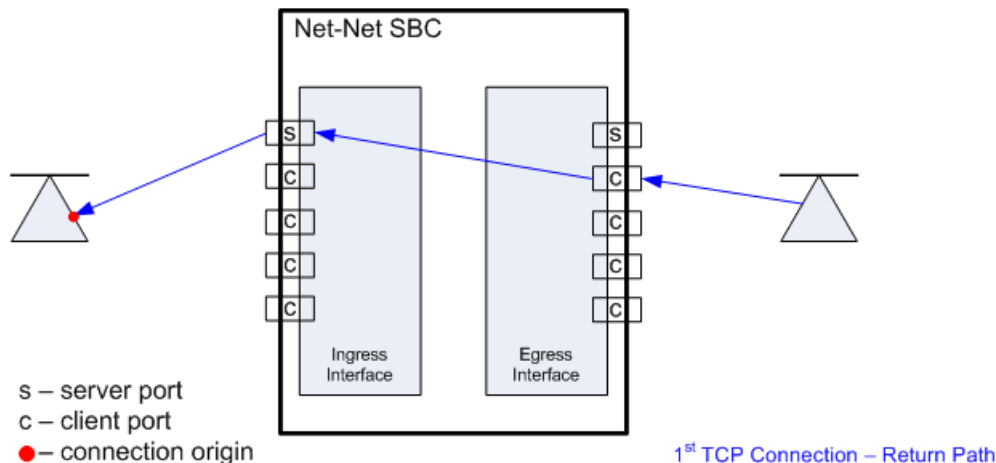
The server port is used for initiating a new TCP connection. An endpoint sends the first packet to a server port on the ingress interface. The packet is forwarded out of the Oracle Communications Session Border Controller through a client port on the egress interface toward an endpoint:

### TCP Connection 1 - Eastward Path



The endpoint responds back to the client port on the egress interface. This message traverses the Oracle Communications Session Border Controller and is forwarded out of the server port on the ingress interface where the initial packet was sent. The remainder of the TCP flow uses the server and client port pair as a tunnel through the Oracle Communications Session Border Controller:

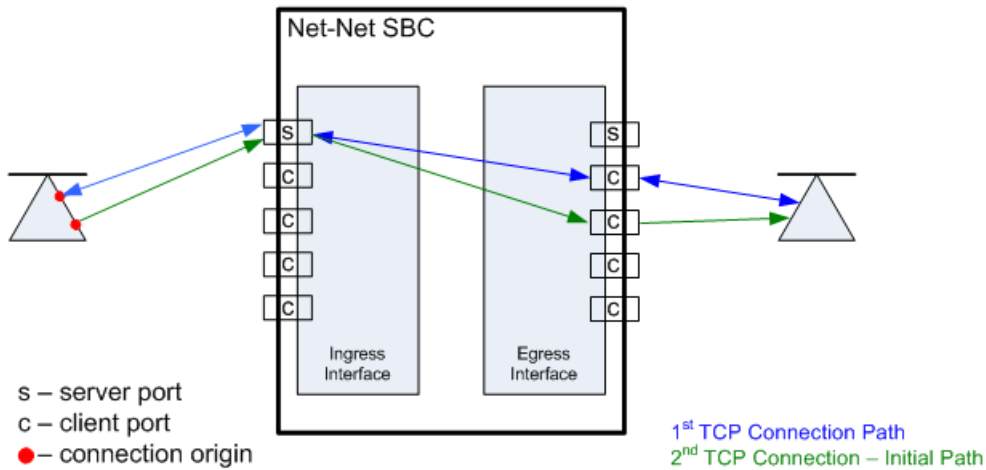
### TCP Connection 1 - Westward Path



When the second TCP connection is set up in the same direction as in the first example, the first packet is still received on the server port of the ingress interface. The next unused client port is chosen for the packet to exit the Oracle Communications Session Border Controller:

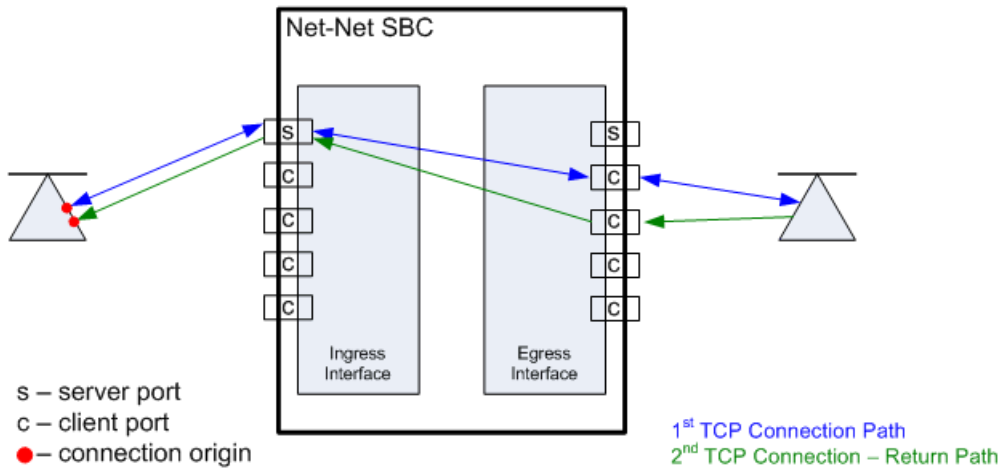


### TCP Connection 2 - Eastward Path



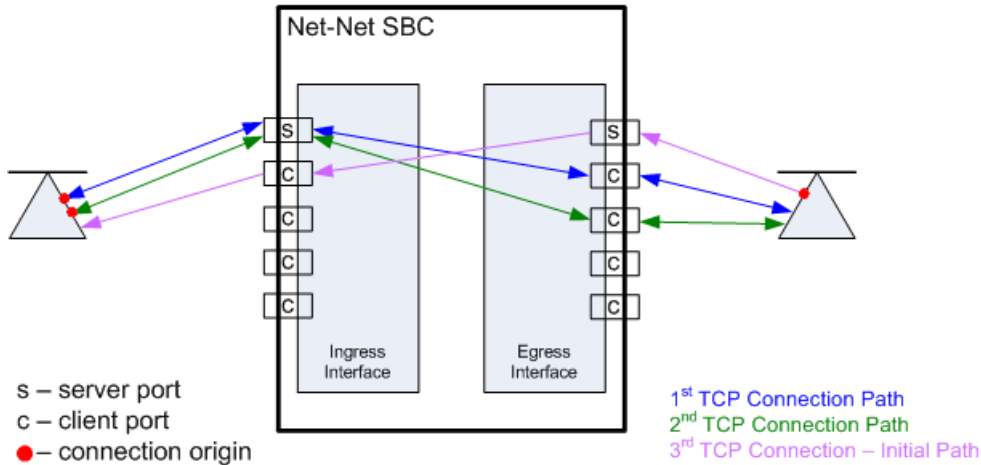
The response takes the same path back to the caller. The remainder of the second TCP connection uses this established path:

### TCP Connection 2 - Westward Path



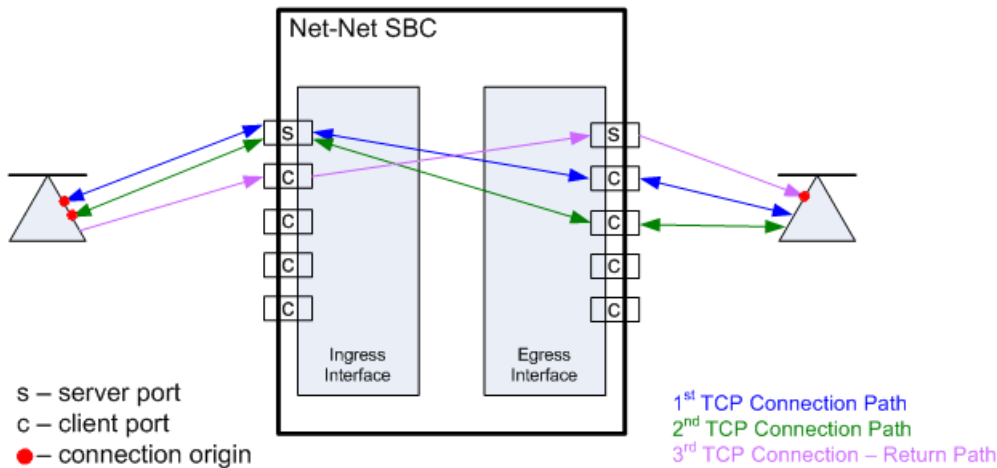
When the callee initiates a TCP connection, it must send its initial traffic to the server port on its Oracle Communications Session Border Controller ingress interface. The packet is forwarded out of the first free client port on the egress side of this TCP connection toward the caller.

TCP Connection 3 – Callee Initiates Connection



The caller’s response takes the same path back to the callee that initiated this TCP connection. The remainder of the third TCP connection uses this established path.

TCP Connection 3 – Return Path: Caller to Callee

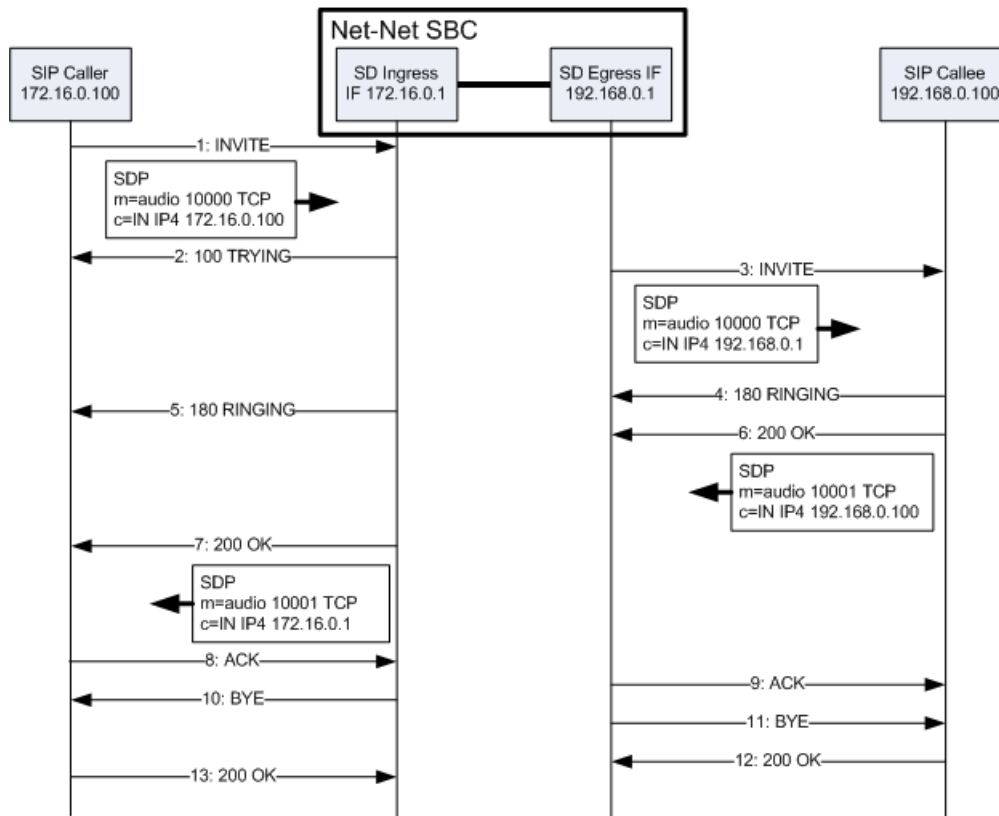


The Oracle Communications Session Border Controller can support a total of eight media-over-TCP connections per call. A maximum of 4 connections are supported as initiated from each side of the call.

**SDP Offer Example**

The following abbreviated call flow diagram sets up a media-over-TCP flow. Observe that the caller listens for audio over TCP on 172.16.0.10:10000, as described in the SDP offer (1). The Oracle Communications Session Border Controller re-writes the m and c lines in the SDP offer to reflect that it is listening for audio over TCP on its egress interface at 192.168.0.1:10000 (3). The Oracle Communications Session Border Controller then forwards the SIP invite to the callee.

The SIP callee responds with an SDP answer in a 200 OK message. The callee indicates it is listening for the audio over TCP media on 192.168.0.10:10001 (6). The Oracle Communications Session Border Controller re-writes the m and c lines in the SDP answer to reflect that it is listening for audio over TCP on the call's ingress interface at 172.16.0.1:10001 (7). The Oracle Communications Session Border Controller then forwards the SIP invite to the caller.



All interfaces involved with the end-to-end TCP flow have now established their listening IP address and port pairs.

### Timers

The Oracle Communications Session Border Controller has three guard timers that ensure a TCP media flow does not remain connected longer than configured. You can set each of these from 0 (disabled) to 999999999 in seconds.

- TCP initial guard timer — Sets the maximum time in seconds allowed to elapse between the initial SYN packet and the next packet in this flow.
- TCP subsequent guard timer — Sets the maximum time in seconds allowed to elapse between all subsequent sequential TCP packets.
- TCP flow time limit — Sets the maximum time that a single TCP flow can last. This does not refer to the entire call.

### TCP Port Configuration

To configure media over TCP:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-level configuration elements.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **media-manager** and press Enter to begin configuring media over TCP.

```
ACMEPACKET(media-manager)# media-manager
ACMEPACKET(media-manager-config)#
```

4. **tcp-number-of-ports-per-flow**—Enter the number of ports, inclusive of the server port, to use for media over TCP. The total number of supported flows is this value minus one. The default is 2. The valid range is:

## Realms and Nested Realms

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- Minimum—2
- Maximum—5

```
ACMEPACKET (realm-config) # tcp-number-of-ports-per-flow 5
```

5. **tcp-initial-guard-timer**—Enter the maximum time in seconds allowed to elapse between the initial SYN packet and the next packet in a media-over-TCP flow. The default is 300. The valid range is:

- Minimum—0
- Maximum—999999999

```
ACMEPACKET (realm-config) # tcp-initial-guard-timer 300
```

6. **tcp-subsq-guard-timer**—Enter the maximum time in seconds allowed to elapse between all subsequent sequential media-over-TCP packets. The default is 300.

- Minimum—0
- Maximum—999999999

```
ACMEPACKET (realm-config) # tcp-subsq-guard-timer 300
```

7. **tcp-flow-time-limit**—Enter the maximum time in seconds that a media-over-TCP flow can last. The default is 86400. The valid range is:

- Minimum—0
- Maximum—999999999

```
ACMEPACKET (realm-config) # tcp-flow-time-limit 86400
```

## Restricted Media Latching

---

The restricted media latching feature lets the Oracle Communications Session Border Controller latch only to media from a known source IP address, in order to learn and latch the dynamic UDP port number. The restricting IP address's origin can be either the SDP information or the SIP message's Layer 3 (L3) IP address, depending on the configuration.

### About Latching

Latching is when the Oracle Communications Session Border Controller listens for the first RTP packet from any source address/port for the destination address/port of the Oracle Communications Session Border Controller. The destination address/port is allocated dynamically and sent in the SDP. After it receives a RTP packet for that allocated destination address/port, the Oracle Communications Session Border Controller only allows subsequent RTP packets from that same source address/port for that particular Oracle Communications Session Border Controller destination address/port. Latching does not imply that the latched source address/port is used for the destination of the reverse direction RTP packet flow (it does not imply the Oracle Communications Session Border Controller will perform symmetric RTP).

### Restricted Latching

The Oracle Communications Session Border Controller restricts latching of RTP/RTCP media for all calls within a realm. It latches to media based on one of the following:

- SDP: the IP address and address range based on the received SDP c= connect address line in the offer and answer.
- Layer 3: the IP address and address range based on the received L3 IP address of the offer or answer. This option is for access registered HNT endpoints. If the L3 IP address is locally known and cached by the Oracle Communications Session Border Controller as the public SIP contact address, that information could be used instead of waiting for a response. The Oracle Communications Session Border Controller might use the L3 IP address restriction method for all calls regardless of whether the endpoint is behind a NAT or not, for the same realms.

## Symmetric Latching

A mode where a device's source address/ports for the RTP/RTCP it sends to the Oracle Communications Session Border Controller (SBC) that are latched, are then used for the destination of RTP/RTCP sent to the device.

After allocating the media session in SIP, the SBC sets the restriction mode and the restriction mask for the calling side as well as for the called side. It sets the source address and address prefix bits in the flow. It also parses and loads the source flow address into the MIBOCO messages. After receiving the calling SDP, the SBC sets the source address (address and address prefix) in the appropriate flow (the flow going from calling side to the called side). After receiving the SDP from the called side, the SBC sets the source address in the flow going from the called side to the calling side.

The SBC uses either the address provided in the SDP or the layer 3 signaling address for latching. You also configure the SBC to enable latching so that when it receives the source flow address, it sets the address and prefix in the NAT flow. When the NAT entry is installed, all the values are set correctly. In addition, sipd sends the information for both the incoming and outgoing flows. After receiving SDP from the called side sipd, the SBC sends information for both flows to the MIBOCO so that the correct NAT entries are installed.

Enabling restricted latching may make the SBC wait for a SIP/SDP response before latching, if the answerer is in a restricted latching realm. This is necessary because the SBC does not usually know what to restrict latching to until the media endpoint is reached. The only exception could be when the endpoint's contact/IP is cached.

## Relationship to Symmetric Latching

The current forced HNT symmetric latching feature lets the Oracle Communications Session Border Controller assume devices are behind NATs, regardless of their signaled IP/SIP/SDP layer addresses. The Oracle Communications Session Border Controller latches on any received RTP destined for the specific IP address/port of the Oracle Communications Session Border Controller for the call, and uses the latched source address/port for the reverse flow destination information.

If both restricted latching and symmetric latching are enabled, the Oracle Communications Session Border Controller only latches if the source matches the restriction, and the reverse flow will only go to the address/port latched to, and thus the reverse flow will only go to an address of the same restriction.

- Symmetric latching is enabled.

If symmetric latching is enabled, the Oracle Communications Session Border Controller sends the media in the opposite direction to the same IP and port, after it latches to the source address of the media packet.

- Symmetric latching is disabled.

If symmetric latching is disabled, the Oracle Communications Session Border Controller only latches the incoming source. The destination of the media in the reverse direction is controlled by the SDP address.

## Example 1

A typical example is when the Oracle Communications Session Border Controller performs HNT and non-HNT registration access for endpoints. Possibly the SDP might not be correct, specifically if the device is behind a NAT. Therefore the Oracle Communications Session Border Controller needs to learn the address for which to restrict the media latching, based on the L3 IP address. If the endpoint is not behind a NAT, then the SDP could be used instead if preferred. However, one can make some assumptions that access-type cases will require registration caching, and the cached fixed contact (the public FW address) could be used instead of waiting for any SDP response.

## Example 2

Another example is when a VoIP service is provided using symmetric-latching. A B2BUA/proxy sits between HNT endpoints and the Oracle Communications Session Border Controller, and calls do not appear to be behind NATs from the Oracle Communications Session Border Controller's perspective. The Oracle Communications Session Border Controller's primary role, other than securing softswitches and media gateways, is to provide symmetric latching so that HNT media will work from the endpoints.

To ensure the Oracle Communications Session Border Controller's latching mechanism is restricted to the media from the endpoints when the SIP Via and Contact headers are the B2BUA/proxy addresses and not the endpoints', the

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endpoint's real (public) IP address in the SDP of the offer/answer is used. The B2BUA/proxy corrects the c= line of SDP to that of the endpoints' public FW address.

The Oracle Communications Session Border Controller would then restrict the latching to the address in the SDP of the offer from the access realm (for inbound calls) or the SDP answer (for outbound calls).

## Restricted Latching Configuration

To configure restricted latching:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-level configuration elements.

```
ACMEPACKET (configure) # media-manager  
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config  
ACMEPACKET (realm-config) #
```

4. Select the realm where you want to apply this feature.

```
ACMEPACKET (realm-config) # select  
identifier:  
1: Acme_Realm <none>          0.0.0.0  
2: H323REALM <none>         0.0.0.0  
selection:1  
ACMEPACKET (realm-config) #
```

5. **restricted-latching**— Enter the restricted latching mode. The default is **none**. The valid values are:
  - **none**—No restricted-latching used
  - **sdp**—Use the address provided in the SDP for latching
  - **peer-ip**—Use the layer 3 signaling address for latching
6. **restriction-mask**— Enter the number of address bits you want used for the source latched address. This field will be used only if the restricted-latching is used. The default is **32**. When this value is used, the complete IP address is matched for IPv4 addresses. The valid range is:
  - Minimum—1
  - Maximum—128

## Media Release Across SIP Network Interfaces

---

This feature lets the Oracle Communications Session Border Controller release media between two SIP peers, between two realms on two network interfaces of the same Oracle Communications Session Border Controller. Use this feature when you want the Oracle Communications Session Border Controller to release media for specific call flows, regardless of the attached media topology.

## Media Release Configuration

To configure media release across network interfaces:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-level configuration elements.

```
ACMEPACKET (configure) # media-manager  
ACMEPACKET (media-manager) #
```

- Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

- Select the realm where you want to apply this feature.

```
ACMEPACKET(realm-config)# select
identifier:
1: Acme_Realm <none>          0.0.0.0
2: H323REALM <none>         0.0.0.0
selection:1
ACMEPACKET(realm-config)#
```

- mm-in-system**—Set this parameter to **enabled** to manage/latch/steer media in the Oracle Communications Session Border Controller. Set this parameter to **disabled** to release media in the Oracle Communications Session Border Controller.



**Note:** Setting this parameter to disabled will cause the Oracle Communications Session Border Controller to NOT steer media through the system (no media flowing through this Oracle Communications Session Border Controller).

The default is **enabled**. The valid values are:

- enabled | disabled

## Media Release Behind the Same IP Address

The media management behind the same IP feature lets the Oracle Communications Session Border Controller release media when two endpoints are behind the same IP address, in the same realm. Using this feature prevents the media for intra-site calls from going through the Oracle Communications Session Border Controller. You can use this feature for both hosted NAT traversal (HNT) and non-HNT clients. It works with NATed endpoints and for non-NATed ones that are behind the same IP.

## Additional Media Management Options

Additional media management options include:

- Media directed between sources and destinations within this realm on this specific Oracle Communications Session Border Controller. Media travels through the Oracle Communications Session Border Controller rather than straight between the endpoints.
- Media directed through the Oracle Communications Session Border Controller between endpoints that are in different realms, but share the same subnet.
- For SIP only, media released between multiple Oracle Communications Session Border Controllers.

To enable SIP distributed media release, you must set the appropriate parameter in the realm configuration. You must also set the SIP options parameter to media-release with the appropriate header name and header parameter information. This option defines how the Oracle Communications Session Border Controller encodes IPv4 address and port information for media streams described by, for example, SDP.

## Configuring Media Release Behind the Same IP Address

You need to configure both the mm-in-realm and mm-same-ip parameters for the realm:

- If the mm-in-realm parameter is disabled, the mm-same-ip parameter is ignored.
- If the mm-in-realm parameter is enabled and the mm-same-ip parameter is disabled, media will be managed in the realm but released if the two endpoints are behind the same IP address.

To configure media management:

- In Superuser mode, type **configure terminal** and press Enter.

## Realms and Nested Realms

---

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config  
ACMEPACKET (realm-config) #
```

From this point, you can configure realm parameters. To view all realm configuration parameters, enter a **?** at the system prompt.

4. **mm-in-realm**—Enable if you plan to use **mm-same-ip**. If this parameter is disabled, the **mm-same-ip** parameter is ignored. If you set this to **enabled** and **mm-same-ip** to **disabled**, media is managed in the realm but released if the two endpoints are behind the same IP address. The default is **disabled**. The valid values are:
  - enabled | disabled
5. **mm-same-ip**—Enable if you want media to go through this Oracle Communications Session Border Controller, if **mm-in-realm** is **enabled**. When **disabled**, the media will not go through the Oracle Communications Session Border Controller for endpoint that are behind the same IP. The default is **enabled**. The valid values are:
  - enabled | disabled

## Bandwidth CAC for Media Release

---

The bandwidth CAC for media release feature adds per-realm configuration that determines whether or not to include inter-realm calls in bandwidth calculations. When you use this feature, the Oracle Communications Session Border Controller's behavior is to count and subtract bandwidth from the used bandwidth for a realm when a call within a single site has its media released. When you do not enable this feature (and the Oracle Communications Session Border Controller's previous behavior), the Oracle Communications Session Border Controller does not subtract the amount of bandwidth.

In other words:

- When you enable this feature, an inter-realm media-released call will decrement the maximum bandwidth allowed in that realm with the bandwidth used for that call.
- When you disable this feature (default behavior), and inter-realm media-released call will not decrement the maximum bandwidth allowed for that call.

## Bandwidth CAC Configuration

To enable bandwidth CAC for media release:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager  
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config  
ACMEPACKET (realm-config) #
```

4. Select the realm where you want to want to add this feature.

```
ACMEPACKET (realm-config) # select
```

5. **bw-cac-non-mm**—Enable this parameter to turn on bandwidth CAC for media release. The default is **disabled**. The valid values are:



- enabled | disabled
6. Save and activate your configuration.

## Media Release between Endpoints with the Same IP Address

You can configure your Oracle Communications Session Border Controller to release media between two endpoints even when one of them:

- Is directly addressable at the same IP address as a NAT device, but is not behind a NAT device
- Is at the same IP address of a NAT device the other endpoint is behind

You enable this feature on a per-realm basis by setting an option in the realm configuration.

When this option is not set, the Oracle Communications Session Border Controller will (when configured to do so) release media between two endpoints sharing one NAT IP address in the same realm or network.

## Media Release Configuration

In order for this feature to work properly, the following conditions apply for the realm configuration:

- Either the `mm-in-realm` or the `mm-in-network` parameter must be disabled; you can have one of these enabled as long as the other is not. The new option will apply to the parameter that is disabled.
- If either the `mm-in-realm` or `mm-in-network` parameter is enabled, then the `mm-same-ip` parameter must be disabled.

To enable media release between endpoints with the same IP address:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type `media-manager` and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type `realm-config` and press Enter.

```
ACMEPACKET(media-manager)# realm-config
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI `select` command) the realm that you want to edit.

4. **options**—Set the options parameter by typing `options`, a Space, the option name `release-media-at-same-nat` with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(realm-config)# options +release-media-at-same-nat
```

**If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the realm configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.**

5. Save and activate your configuration.

## Media Release Behind the Same NAT IP Address

You can now configure your Oracle Communications Session Border Controller to release media between endpoints sharing the same NAT IP address, even if one endpoint is at—but not behind—the same NAT. This feature expands on the Oracle Communications Session Border Controller's pre-existing ability to release media between calling and called parties behind the same IP address/NAT device in the same realm or network.

### Media Release Configuration

For this feature to work properly, your realm configuration should either have the `mm-in-realm` or `mm-in-network` parameter set to disabled, unless the `mm-same-ip` parameter is set to disabled. If the `mm-same-ip` parameter is enabled, then `mm-in-realm` or `mm-in-network` can both be enabled.

To set the option that enables media release behind the same IP address:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type `media-manager` and press Enter.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type `realm-config` and press Enter.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI `select` command) the realm that you want to edit.

4. **options**—Set the options parameter by typing `options`, a Space, the option name `release-media-at-same-nat` with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (realm-config) # options +release-media-at-same-nat
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the realm configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

### Codec Reordering

---

Certain carriers deploy voice services where their peering partners do not use the carriers' preferred codecs. The Oracle Communications Session Border Controller can now reorder the codecs so that the preferred one is selected first.

Take the example of a carrier that deploys a voice service using G.729 rather than G.711. If that carrier has a peering partner providing call origination for the VoIP customers with G.711 used as the preferred codec, there can be issues with codec selection.

The Oracle Communications Session Border Controller resolves this issue by offering its codec reordering feature. Enabled for realms and session agents, this feature gives the Oracle Communications Session Border Controller the ability to reorder the default codec in an SDP offer to the preferred codec before it forwards the offer to the target endpoint. When you enable this feature, you increase the probability that the target endpoint will choose the preferred codec for its SDP answer, thereby avoiding use of the undesired codec.


You enable codec reordering feature by setting the `preferred-codec=X` (where X is the preferred codec) option in the realm and session agent configurations. You set it in the realm from which the Oracle Communications Session Border Controller receives SDP offers (in requests or responses), and for which the media format list needs to be reordered by the Oracle Communications Session Border Controller prior to being forwarded. To configure additional codec ordering support for cases when a response or request with an SDP offer is from a session agent, you can set this option in the session agent configuration.

If you enable the option, the Oracle Communications Session Border Controller examines each SDP media description before it forwards an SDP offer. And if necessary, it performs reordering of the media format list to designate that the preferred codec as the default.

The Oracle Communications Session Border Controller determines preferred codecs in the following ways:

- If the response or request with an SDP offer is from a session agent, the Oracle Communications Session Border Controller determines the preferred codec by referring to the session agent configuration. You set the preferred codec for a session agent by configuring it with the `preferred-codec=X` option.
- If the response or request with an SDP offer is not from a session agent or is from a session agent that does not have the `preferred-codec=X` option configured, the Oracle Communications Session Border Controller determines the preferred codec by referring to the `preferred-codec=X` option in the realm.
- If the Oracle Communications Session Border Controller cannot determine a preferred codec, it does not perform codec reordering.

The way that the Oracle Communications Session Border Controller performs codec reordering is to search for the preferred codec in the SDP offer's media description (`m=`) line, and designate it as the default codec (if it is not the default already). After it marks the preferred codec as the default, the Oracle Communications Session Border Controller does not perform any operation on the remaining codecs in the media format list.

 **Note:** that the Oracle Communications Session Border Controller performs codec reordering on the media format list only. If the `rtptime` attribute of the preferred codec is present, the Oracle Communications Session Border Controller does not reorder it.

## Preferred Codec Precedence


When you configure preferred codecs in session agents or realms, be aware that the codec you set for a session agent takes precedence over one you set for a realm. This means that if you set preferred codecs in both configurations, the one you set for the session agent will be used.

In the case where the Oracle Communications Session Border Controller does not find the session agent's preferred codec in the SDP offer's media format list, then it does not perform codec reordering even if the media format list contains the realm's preferred codec.

## Codec Reordering Configuration

When you configure codec ordering, the codec you set in either the session agent or realm configuration must match the name of a media profile configuration. If your configuration does not use media profiles, then the name of the preferred codec that you set must be one of the following:

- PCMU
- G726-32
- G723
- PCMA
- G722
- G728
- G729

 **Note:** If you configure this feature for a session agent, you must configure it for the associated realm as well. Otherwise, the feature will not work correctly.

### Setting a Preferred Codec for a Realm

To set a preferred codec for a realm configuration:

These instructions assume that you want to add this feature to a realm that has already been configured.

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `media-manager` and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

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---

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. Select the realm where you want to apply this feature.

```
ACMEPACKET(realm-config)# select
identifier:
1: public      media2:0      0.0.0.0
2: private    media1:0      0.0.0.0
selection:1
ACMEPACKET(realm-config)#
```

5. **options**—Set the **options** parameter by typing **options**, a Space, the option name preceded by a plus sign (+) (**preferred-codec=X**), and then press Enter. X is the codec that you want to set as the preferred codec.

```
ACMEPACKET(realm-config)# options +preferred-codec=PCMU
```

**If you type `options preferred-codec=X`, you will overwrite any previously configured options. In order to append the new option to the realm-config's options list, you must prepend the new option with a plus sign as shown in the previous example.**

6. Save and activate your configuration.

### Setting a Preferred Codec for a Session Agent

To set a preferred codec for a session agent configuration:

These instructions assume that you want to add this feature to a session agent that has already been configured.

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

4. Select the session agent where you want to apply this feature.

```
ACMEPACKET(session-agent)# select
<hostname>:
1: acmepacket.com realm=      ip=
2: sessionAgent2  realm=tester ip=172.30.1.150
selection:
selection:1
ACMEPACKET(session-agent)#
```

5. **options**—Set the **options** parameter by typing **options**, a Space, the option name preceded by a plus sign (+) (**preferred-codec=X**), and then press Enter. X is the codec that you want to set as the preferred codec.

```
ACMEPACKET(session-agent)# options +preferred-codec=PCMU
```

**If you type `options preferred-codec=X`, you will overwrite any previously configured options. In order to append the new option to the session agent's options list, you must prepend the new option with a plus sign as shown in the previous example.**


6. Save and activate your configuration.

## Media Profiles Per Realm

For different codecs and media types, you can set up customized media profiles that serve the following purposes:

- Police media values
- Define media bandwidth policies
- Support H.323 slow-start to fast-start interworking

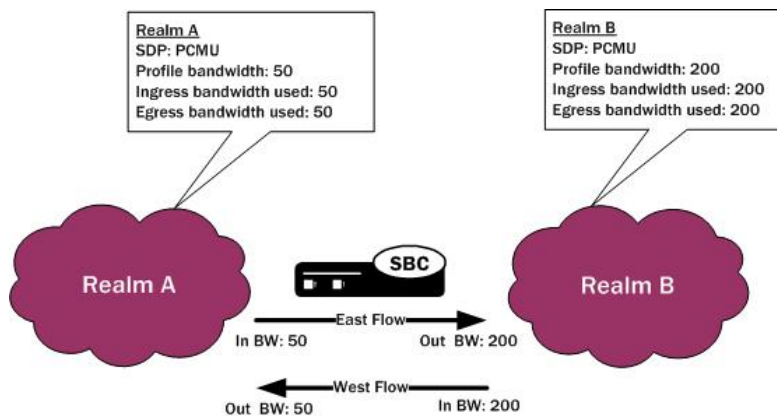
You can use media policies globally for the Oracle Communications Session Border Controller, or—starting with Release C6.1.0—you can configure them for application on a per-realm basis. For a realm, you can configure a list of media profiles you want applied. The Oracle Communications Session Border Controller matches the value you set for the **match-media-profiles** parameter, and then applies those media profiles to the realm itself and to all of its child realms (but not to its parent realms).

 **Note:** This feature has no impact on the ways the Oracle Communications Session Border Controller uses media profiles non-realm applications such as: H.323 interfaces, SIP interfaces, IWF, session agents, codec policies, and policy attributes.

## Call Admission Control and Policing

The Oracle Communications Session Border Controller supports call admission control (CAC) based on realm, and it applies the limits on either ingress or egress bandwidth counters. If a call exceeds bandwidth on either the ingress or egress side, the Oracle Communications Session Border Controller rejects the call. You can also use per-user CAC, which limits the maximum bandwidth from the east and west flows for both the TO and FROM users.

When you apply media profiles to a realm, the Oracle Communications Session Border Controller applies bandwidth policing from the flow's ingress realm media profile. In the diagram below, the Oracle Communications Session Border Controller policies traffic for Realm A based Realm A's policing values, and the same is true for Realm B.



## Media Profile Configuration

This section shows you how to configure multiple media profiles per realm, and it explains how to use wildcarding.

To reference a media profile in this list, you need to enter its name and subname values in the following format `<name>::<subname>`. Releases C6.1.0 and later accept the subname so you can configure multiple media profile for the same codec; the codec **name** customarily serves and the name value for a media profile configuration.

### About Wildcarding

You can wildcard both portions (name and subname) of this value:

- When you wildcard the **name** portion of the value, you can provide a specific subname that the Oracle Communications Session Border Controller uses to find matching media profiles.
- When you wildcard the subname portion of the value, you can provide a specific **name** that the Oracle Communications Session Border Controller uses to find matching media profiles.

## Realms and Nested Realms

You can also enter the name value on its own, or wildcard the entire value. Leaving the subname value empty is also significant in that it allows the realm to use all media profile that have no specified **subname**. However, you cannot leave the **name** portion of the value unspecified (as all media profiles are required to have names).

Consider the examples in the following table:

Syntax	Example Value	Description
<name>	PCMU	Matches any and all media profiles with the name value configured as PCMU. This entry has the same meaning as a value with this syntax: <name>::*.
<name>::	PCMU::	Matches a media profile with the name with the name value configured as PCMU with an empty subname parameter.
<name>:: <subname&gt;< td=""><td>PCMU::64k</td><td>Matches a media profiles with the name with the name value configured as PCMU with the subname parameter set to 64k.</td></subname&gt;<>	PCMU::64k	Matches a media profiles with the name with the name value configured as PCMU with the subname parameter set to 64k.
*	*	Matches anything, but does not have to be a defined media profile.
*::*	*::*	Matches any and all media profiles, but requires the presence of media profile configurations.
*:: <subname&gt;< td=""><td>*::64k</td><td>Matches all media profiles with this subname. You might have a group of media profiles with different names, but the same subname value.</td></subname&gt;<>	*::64k	Matches all media profiles with this subname. You might have a group of media profiles with different names, but the same subname value.
*::	*::	Matches any media profiles with an empty subname parameter.
::	::	Invalid
::*	::*	Invalid

The Oracle Communications Session Border Controller performs matching for wildcarded **match-media-profiles** values last. Specific entries are applied first and take precedence. When the Oracle Communications Session Border Controller must decide between media profiles matches, it selects the first match.

To use media profiles for a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **realm-config** and press Enter. If you are adding this feature to a pre-existing realm configuration, you will need to select and edit your realm.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. **match-media-profiles**—In the form **<name>::, enter the media profiles you would like applied to this realm. These values correspond to the name and subname parameters in the media profile configuration. You can wildcard either of these portions of the value, or you can leave the **<subname>** portion empty.**

This parameter has no default.

5. Save and activate your configuration.

## Multiple Media Profiles

You can use the media profiles configuration to set up:

- One media profile for a particular SIP SDP encoding (such as G729), where the name of the profile identifies it uniquely. This behavior is your only option in Oracle Communications Session Border Controller release prior to Release C6.1.0.
- Multiple media profiles for the same SIP SDP encoding. Available in Release C6.1.0 and forward, you can create multiple media profiles for the same encoding. To do so, you add a subname to the configuration, thereby identifying it uniquely using two pieces of information rather than one.

The sections below provide two descriptions of deployments where using multiple profiles for the same codec would solve codec and packetization problems for service providers.

### Use Case 1

Service Provider 1 peers with various carriers, each of which uses different packetization rates for the same codec. For example, their Peer 1 uses 10 milliseconds G.711 whereas their Peer 2 uses 30 milliseconds for the same codec. The difference in rates produces a difference in bandwidth consumption—resulting in a difference in SLA agreements and in Oracle Communications Session Border Controller call admission control (CAC) and bandwidth policing. Service Provider 1 uses the Oracle Communications Session Border Controller’s media profile configuration parameters to determine CAC (**req-bandwidth**) and bandwidth policing (**avg-rate-limit**). Because this service provider’s peers either do not use the SDP p-time attribute or use it inconsistently, it is difficult to account for bandwidth use. And so it is likewise difficult to set up meaningful media profiles.

The best solution for this service provider—given its traffic engineering and desire for the cleanest routing and provisioning structures possible—is to define multiple media profiles for the same codec.

### Use Case 2

Service Provider 2 supports H.263 video, for which the Oracle Communications Session Border Controller offers a pre-provisioned media profile with a set bandwidth value. And yet, H.263 is not a codec that has a single bandwidth value. Instead, H.263 can have different bandwidth values that correspond to various screen resolution and quality. While it is true that the Oracle Communications Session Border Controller can learn the requisite bandwidth value from SDP, not all SDP carries the bandwidth value nor do system operators always trust the values communicated.

Configuring multiple media profiles for the same codec (here, H.263) helps considerably with this problem—and moves closer to complete solution. Service Provider 2 can configure H.263 media profiles capable of handling the different bandwidth values that might appear.

## Multiple Media Profiles Configuration

Configuring the **subname** parameter in the media profiles configuration allows you to create multiple media profiles with the same name.

To configure the subname parameter for a media profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **media-profile** and press Enter. If you are adding this feature to a pre-existing media profile configuration, you will need to select and edit your media profile.

```
ACMEPACKET (session-router) # media-profile
ACMEPACKET (media-profile) #
```

## Realms and Nested Realms

---

4. **subname**—Enter the subname value for this media profile. Information such as the rate or bandwidth value make convenient subname values. For example, you might set the **name** of the media profile as PCMU and the **subname** as 64k.

This parameter is not required and has no default.

5. Save and activate your configuration.

## SIP Disable Media Inactivity Timer for Calls Placed on Hold

---

Hardware-based media flow guard timers detect when a call has lost media while it is being relayed through the Oracle Communications Session Border Controller. In response, the system tears down the call.

You can configure **disable-guard-timer-sendonly** to disable media inactivity timers for calls placed on hold. The Oracle Communications Session Border Controller disables initial and subsequent guard timers for media when the SIP or IWF call is put on hold with a 0.0.0.0 address in:

- The **c=connection** line
- An **a=inactive** attribute
- An **a=sendonly** attribute

It should be noted that disabling the media inactivity timers will also disable the guard timers for calls which are not necessarily on hold, but simply are one-way audio applications.

No license requirements to enable this feature.

## Media Inactivity Timer Configuration

To disable the media inactivity timer for calls placed on hold:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **media-manager-config** and press Enter.

```
ACMEPACKET (media-manager) # media-manager-config
ACMEPACKET (media-manager-config) #
```

4. **options**—Set the options parameter by typing **options**, a Space, the option-name **disable-guard-timer-sendonly** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (media-manager-config) # options +disable-guard-timer-sendonly
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the SIP interface configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

## Peer-to-Peer MSRP TCP Stitching

---

The Oracle Communications Session Border Controller supports peer-to-peer TCP connections for peers behind NATs, enabling Message Session Relay Protocol (MSRP) client to communicate with one another. More specifically, the Oracle Communications Session Border Controller can:



- Establish incoming TCP connections with each endpoint participating in the MSRP session using a 3-way handshake. The Oracle Communications Session Border Controller receives incoming SYNs on the local address and port provided in the SDP offer and answer to each endpoint.
- Stitch together the two TCP connections internally after successful establishment of both connections. This capability is used when the caller and the callee initiate TCP SYNs towards one another via the Oracle Communications Session Border Controller; the stitching makes both clients think they are talking to a server. To achieve this end, the Oracle Communications Session Border Controller caches SYNs from both sides so it can modify the SYN packets to SYN-Acks with the correct sequence and Ack numbers.

Note, though this case is rare, that if a user is behind a NAT offers a=passive, then this feature cannot function properly.

- Relay MSRP stream between the endpoints.
- Police bandwidth for MSRP streams based on a defined media profile for MSRP.



---

## SIP Signaling Services

This chapter explains how to configure the Oracle Communications Session Border Controller to support Session Initiation Protocol (SIP) signaling services for hosted IP services applications. SIP is a text-based application-layer signaling protocol that creates, identifies, and terminates multimedia sessions between devices.

### About the Oracle Communications Session Border Controller and SIP

---

This section describes the Oracle Communications Session Border Controller's support of SIP. It provides the basic information you need to understand before you configure the Oracle Communications Session Border Controller for SIP signaling.

#### Types of SIP Devices

There are four types of SIP devices:

- SIP user agent (UA) is an endpoint in SIP end-to-end communication. A UA is a user agent client (UAC) when it initiates a request and waits to receive a response. A UA is a user agent server (UAS) when it receives a request and generates a response. A given UA will be a UAC or a UAS depending on whether it is initiating the request or receiving the request.
- A SIP proxy (or proxy server) is an intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients. A proxy server's primary role is routing. Its job is to ensure that a request is sent to another entity closer to the targeted user. A proxy interprets, and if necessary, rewrites specific parts of a request message before forwarding it.
- A SIP redirect server is a UAS that generates redirect responses to requests it receives, directing the client to contact an alternate set of targets. Unlike a proxy which forwards the request to the alternate set of targets, the redirect response tells the UAC to directly contact the alternate targets.
- A SIP registrar is a server that accepts REGISTER requests and places the information it receives in those requests into the location service for the domain it handles. Proxies and redirect servers can use the information from the location service to determine the location of the targeted user.

A redirect server and a registrar are each a special type of UA because they act as the UAS for the requests they process.

#### Basic Service Models

The Oracle Communications Session Border Controller operates as a back-to-back user agent (B2BUA) within the following two basic service models:

## SIP Signaling Services

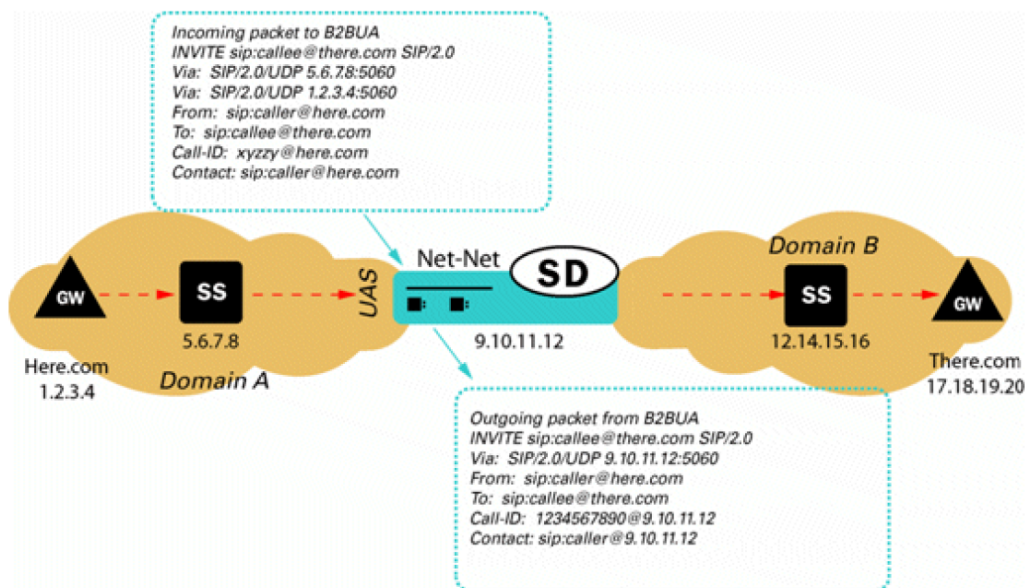
- peering
- hosted IP services

### About B2BUA

A B2BUA is a logical entity that receives a request and processes it as a user agent server (UAS). In order to determine how the request should be answered, it acts as a user agent client (UAC) and generates requests. It maintains dialog state and must participate in all requests sent on the dialogs it has established.

### SIP B2BUA Peering

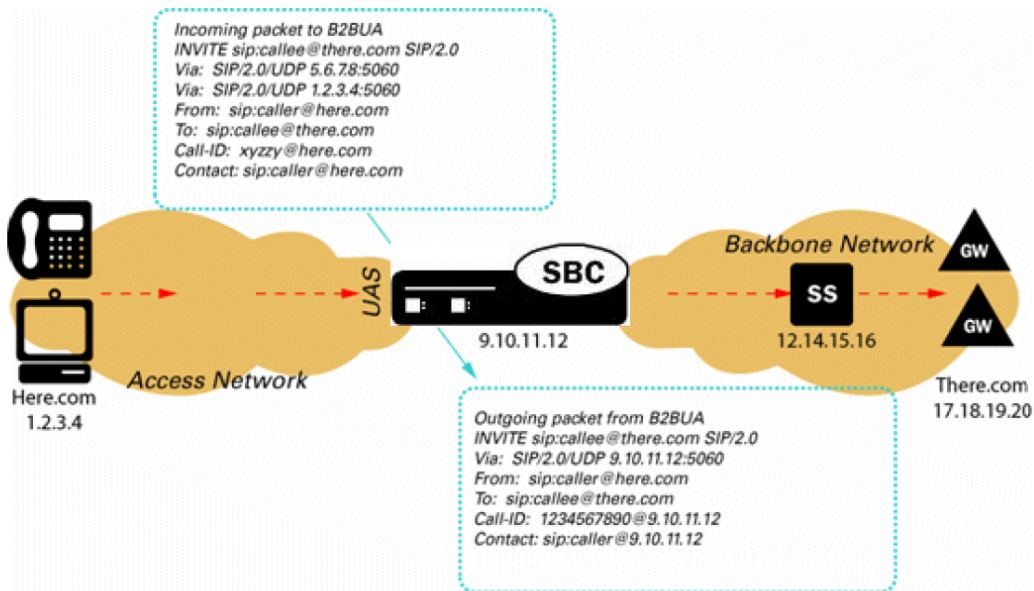
The Oracle Communications Session Border Controller operates as a SIP B2BUA. It terminates SIP sessions and re-originates them as new sessions as they are routed through the Oracle Communications Session Border Controller. For each session, it establishes NAPT translations and re-writes SDP to allow all session related media to be routed through the Oracle Communications Session Border Controller. It generates new call IDs and modifies SIP headers to prevent any protected SIP addresses and route information from being transmitted to external peers. The Oracle Communications Session Border Controller supports multiple SIP interfaces that are associated with a set of media ports, thus appearing as multiple virtual SIP gateways.



### B2BUA Hosted IP Services

The Oracle Communications Session Border Controller acts as an outbound proxy for SIP endpoints and performs the operations required to allow UAs behind NATs to initiate and terminate SIP sessions (Hosted NAT Traversal).

The Oracle Communications Session Border Controller caches registration requests from SIP endpoints and forwards them to the appropriate softswitch or registrar in its backbone network. All subsequent signaling between the endpoint and the backbone network is through the Oracle Communications Session Border Controller. Also, all calling features such as caller ID, call waiting, three-way calling, and call transfer are all supported transparently through the Oracle Communications Session Border Controller.



### SIP B2BUA and L3 L5 NAT

For each SIP session, the Oracle Communications Session Border Controller establishes NATP translations and rewrites SDP to route all session related media through the Oracle Communications Session Border Controller. These actions make the Oracle Communications Session Border Controller look like a SIP gateway. Also, the Oracle Communications Session Border Controller support of multiple SIP interfaces associated with different network interfaces makes it appear as multiple virtual SIP gateways.

This functionality enables the Oracle Communications Session Border Controller to deliver VoIP services to multiple end users, across a VPN backbone.

### About SIP Interfaces

The SIP interface defines the transport addresses (IP address and port) upon which the Oracle Communications Session Border Controller receives and sends SIP messages. You can define a SIP interface for each network or realm to which the Oracle Communications Session Border Controller is connected. SIP interfaces support both UDP and TCP transport, as well as multiple SIP ports (transport addresses). The SIP interface's SIP NAT function lets Hosted NAT Traversal (HNT) be used in any realm.

### SIP INVITE Message Processing

When the session agent element on the softswitch side of the message flow (ingress session agent) has the gateway contact parameter configured as an option, the Oracle Communications Session Border Controller looks for the URI parameter (as defined by the gateway contact parameter) in the Request-URI and decodes the gateway address.

#### Example

The following example shows a SIP INVITE message from a softswitch to a Oracle Communications Session Border Controller.

```
INVITE sip:05030205555@ss-side-ext-address;gateway=encoded-gw-address
From: "Anonymous"<sip:anonymous@anonymous.invalid>;tag=xxxx
To: <sip:05030205555@ss-side-ext-address;user=phone>
```

The following example shows a SIP INVITE message from a Oracle Communications Session Border Controller to a gateway.

```
INVITE sip:05030205555@gw-ip-address SIP/2.0
From: "Anonymous"<sip:anonymous@anonymous.invalid>;tag=SDxxxx-xxxx
To: <sip:05030205555@hostpart;user=phone>
```

## SIP Signaling Services

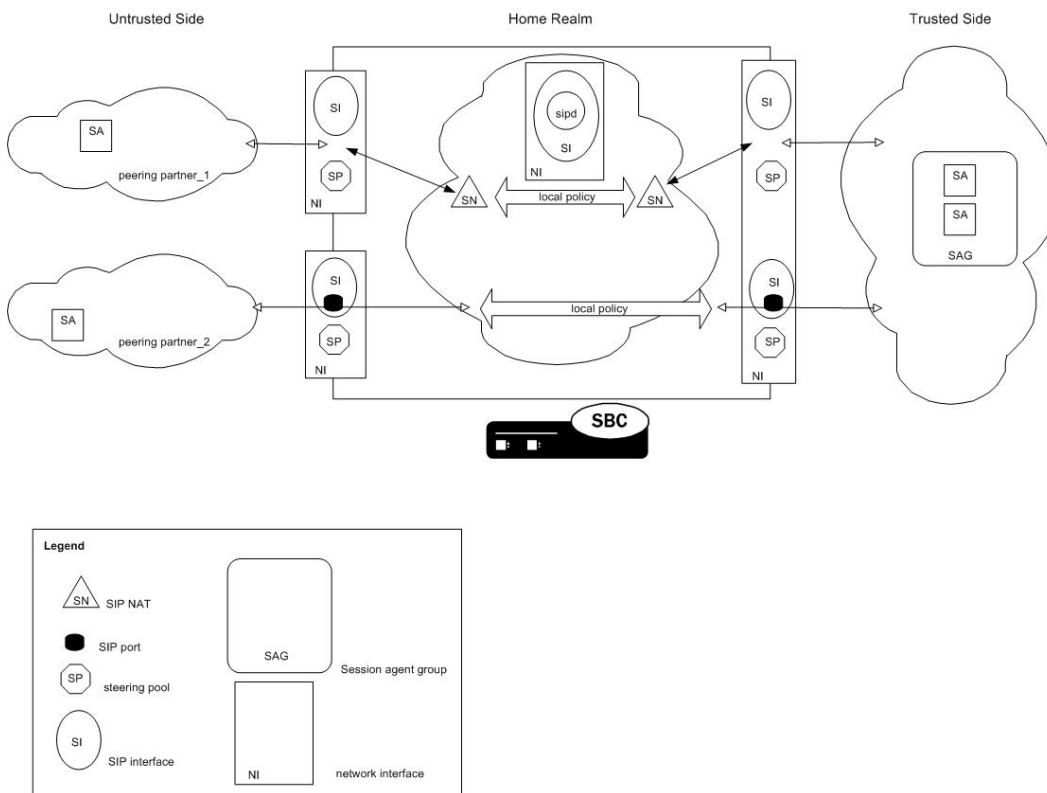
The Oracle Communications Session Border Controller converts the hostpart in the To header except in the following scenarios:

- when the original hostpart value received is an Fully Qualified Domain Name (FQDN)
- when the Oracle Communications Session Border Controller is configured not to NAT the To headers.

Acme Packet recommends configuring the Oracle Communications Session Border Controller to NAT the To headers to ensure the security of protected addresses. Otherwise, the outgoing hostpart is set to the SIP NAT's external proxy address for the SIP NAT's external realm.

## Configuring the Oracle Communications Session Border Controller for SIP Signaling

This section contains a diagram of a B2BUA peering environment that illustrates the Oracle Communications Session Border Controller components you need to configure.



## Home Realm

This section explains how to configure a home realm. The home realm applies only to a SIP configuration. It represents the internal default realm or network for the Oracle Communications Session Border Controller and is where the Oracle Communications Session Border Controller's SIP proxy is located.

## Overview

You primarily use a home realm when using the SIP NAT function to connect multiple realms/networks to the Oracle Communications Session Border Controller. You define the home realm defined as either public or private for the purposes of using the SIP NAT function. If the home realm is public, all external realms are considered private. If the home realm is private, all external networks are considered public. Usually the home realm is public.

Messages are encoded (for example, the topology is hidden) when they pass from a private to a public realm. Messages are decoded when they pass from a public realm to a private realm.

These external realms/networks might have overlapping address spaces. Because SIP messages contain IP addresses, but no layer 2 identification (such as a VLAN tag), the SIP proxy must use a single global address space to prevent confusing duplicate IP addresses in SIP URIs from different realms.

### SIP NAT Function

The SIP NAT function converts external addresses in SIP URIs to an internal home realm address. Usually the external address is encoded into a cookie that is added to the userinfo portion of the URI and the external address is replaced with a home realm address unique to the SIP NAT (the SIP NAT home address).

URIs are encoded when they pass from a private realm to a public realm. When an encoded URI passes back to the realm where it originated, it is decoded (the original userinfo and host address are restored). The encoding/decoding process prevents the confusion of duplicate addresses from overlapping private addresses. It can also be used to hide the private address when a SIP message is traversing a public network. Hiding the address occurs when it is a private address; or when the owner of the private network does not want the IP addresses of their equipment exposed on a public network or on other private networks to which the Oracle Communications Session Border Controller connects.

### Home Realm's Purpose

A home realm is required because the home address for SIP NATs is used to create a unique encoding of SIP NAT cookies. You can define the home realm as a network internal to the Oracle Communications Session Border Controller, which eliminates the need for an actual home network connected to the Oracle Communications Session Border Controller. You can define this virtual home network if the supply of IP addresses is limited (because each SIP NAT requires a unique home address), or if all networks to which the Oracle Communications Session Border Controller is connected must be private to hide addresses.

For example, you can define a public home realm using the loopback network (127.0.0.0) and using the home realm address prefix (for example, 127.0.0.0/8) for encoding addresses that do not match (all addresses outside 127.0.0.0/8) in SIP NAT cookies. The SIP NAT address prefix field can be used to accomplish this while keeping the ability to define an address prefix for the realm for ingress realm determination and admission control. By defining the SIP NAT address prefix as 0.0.0.0, the home realm address prefix is used to encode addresses that do not match.

## Home Realm Configuration

To configure the home realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-config** and press Enter. The system prompt changes.

```
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) #
```

From this point, you can configure SIP configuration parameters. To view all sip-config parameters, enter a ? at the system prompt.

4. **home-realm-id**—Enter the name of the realm you want to use for the realm ID. For example, **acme**.

The name of the realm must correspond to the identifier value you entered when you configured the realm.

5. **egress-realm-id**—Optional. Enter the egress realm ID to define the default route for SIP requests addressed to destinations outside the home realm's address prefix.

If you enter a value for this optional field, it must correspond to the identifier value you entered when you configured the realm.

## SIP Signaling Services

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**Note:** You should leave this parameter blank for access/backbone applications. When left blank, the realm specified in the home-realm-id parameter is used by default.

6. **nat-mode**—Indicate the SIP NAT mode. The default is **none**. The valid values are:
- **public**—Indicates the subnet defined in the addr-prefix-id field of the home realm is public and the subnet defined in the addr-prefix-id field of all external realms identified in the SIP NAT are private networks. IPv4 addresses are encoded in SIP messages received from the external realm defined by the SIP NAT. The IPv4 addresses are decoded in messages that are sent to the realm.
  - **private**—Indicates the subnet defined in the addr-prefix-id field of the home realm is private and the subnet defined in the addr-prefix-id field of all external realms identified in the SIP NAT are public networks. IPv4 addresses are encoded in SIP messages sent to the external realm defined by the SIP NAT and decoded in messages received from the realm.
  - **none**—No SIP NAT function is necessary.

The following example shows the SIP home realm configured for a peering network.

```
sip-config
    state                               enabled
    operation-mode                       dialog
dialog-transparency                     disabled
    home-realm-id                        acme
    egress-realm-id
    nat-mode                              Public
    registrar-domain
    registrar-host
    registrar-port                        0
    init-timer                            500
    max-timer                              4000
    trans-expire                           32
    invite-expire                           180
    inactive-dynamic-conn                  32
    red-sip-port                           1988
    red-max-trans                           10000
    red-sync-start-time                     5000
    red-sync-comp-time                      1000
    last-modified-date                     2005-03-19 12:41:28
```

## SIP Interfaces

---

This section explains how to configure a SIP interface. The SIP interface defines the transport addresses (IP address and port) upon which the Oracle Communications Session Border Controller receives and sends SIP messages.

### Overview

The SIP interface defines the signaling interface. You can define a SIP interface for each network or realm to which the Oracle Communications Session Border Controller is connected. SIP interfaces support both UDP and TCP transport, as well as multiple SIP ports (transport addresses). The SIP interface also lets Hosted NAT Traversal (HNT) be used in any realm.

The SIP interface configuration process involves configuring the following features:

- address and transport protocols (SIP ports)
- redirect action
- proxy mode
- trust mode



---

## About SIP Ports

A SIP port defines the transport address and protocol the Oracle Communications Session Border Controller will use for a SIP interface for the realm. A SIP interface will have one or more SIP ports to define the IP address and port upon which the Oracle Communications Session Border Controller will send and receive messages. For TCP, it defines the address and port upon which the Oracle Communications Session Border Controller will listen for inbound TCP connections for a specific realm.

You need to define at least one SIP port, on which the SIP proxy will listen for connections. If using both UDP and TCP, you must configure more than one port. For example, if a call is sent to the Oracle Communications Session Border Controller using TCP, which it needs to send out as UDP, two SIP ports are needed.

### Preferred SIP Port

When a SIP interface contains multiple SIP ports of the same transport protocol, a preferred SIP port for each transport protocol is selected for outgoing requests when the specific SIP port cannot be determined. When forwarding a request that matched a cached registration entry (HNT or normal registration caching), the SIP port upon which the original REGISTER message arrived is used. Otherwise, the preferred SIP port for the selected transport protocol is used. When selecting the preferred SIP port, the default SIP port of 5060 will be selected over other non-default ports.

For SIP interfaces using the SIP NAT function, the preferred SIP port address and port will take precedence over the external address of the SIP NAT when they do not match. If both TCP and UDP SIP ports are defined, the address and port of the preferred UDP port is used.

## Proxy Mode

The Oracle Communications Session Border Controller's proxy mode determines whether it forwards requests received on the SIP interface to target(s) selected from local policy; or sends a redirect response to the previous hop. Sending the redirect response causes the previous hop to contact the targets directly.

If the source of the request matches a session agent with a proxy mode already defined, that mode overrides the proxy mode defined in the SIP interface.

You can configure the proxy mode to use the Record-Route option. Requests for stateless and transaction operation modes are forwarded with a Record-Route header that has the Oracle Communications Session Border Controller's addresses added. As a result, all subsequent requests are routed through the Oracle Communications Session Border Controller.

## Redirect Action

The redirect action is the action the SIP proxy takes when it receives a SIP Redirect (3xx) response on the SIP interface. If the target of the request is a session agent with redirect action defined, its redirect action overrides the SIP interface's.

You can set the Oracle Communications Session Border Controller to perform a global redirect action in response to Redirect messages. Or you can retain the default behavior where the Oracle Communications Session Border Controller sends SIP Redirect responses back to the previous hop (proxy back to the UAC) when the UAS is not a session agent.

The default behavior of the Oracle Communications Session Border Controller is to recurse on SIP Redirect responses received from the user agent server (UAS) and send a new request to the Contact headers contained in the SIP Redirect response.

Instead of this default behavior, the Oracle Communications Session Border Controller can proxy the SIP Redirect response back to the user agent client (UAC) using the value in the session agent's redirect action field (when the UAS is a session agent). If there are too many UASes to define as individual session agents or if the UASs are HNT endpoints, and SIP Redirect responses need to be proxied for UASs that are not session agents; you can set the default behavior at the SIP Interface level.

### SIP maddr Resolution

Release S-C6.2.0 provides enhanced resolution of addresses found in SIP contact headers, or in the maddr (multicast address) parameter of SIP 3xx REDIRECT messages. Previous releases resolved these addresses as either a host address or as a session agent name. With Release 6.2.0 these addresses can also be resolved as session agent group (SAG) names.

Support for SAG-based resolution is provided by a new **sip-config** parameter, **sag-lookup-on-redirect**. By default, SAG lookup is disabled, providing compatibility with prior releases.

The following sample SIP REDIRECT and ACLI configuration fragment illustrate enhanced processing.

```
Status-Line: SIP/2.0 302 Moved
Message Header
Via: SIP/2.0/UDP
192.168.200.224:5060;branch=z9hG4bKa0fs40009o90sc8oo780.1
From: <sip:1111@192.168.1.222:6000>;tag=1
To: sut <sip:2223@192.168.1.224:5060>;tag=11
Call-ID: 1-28515@192.168.1.222
CSeq: 1 INVITE
Contact: <sip:1111@192.168.1.223;maddr=test.acmepacket.com>
Privacy: user;id;critical;session
P-Preferred-Identity: sipp <sip:sipp@192.168.200.222:5060>
P-Asserted-Identity: abc.com
Subject: abc
Proxy-Require: privacy,prack,abc
Content-Length: 0

session-group
    group-name                test.acmepacket.com
    description
    state                      enabled
    app-protocol              SIP
    strategy                   Hunt
    dest                       192.168.200.222
                              192.168.200.223
...
...
```

In this case, when the Oracle Communications Session Border Controller receives the 302, it resolves the information from maddr to a SAG name. In the above example, it will resolve to the configured SAG – test.acmepacket.com. The destinations configured in SAG test.acmepacket.com will be used to route the call.

SAG-based address resolution is based on the following set of processing rules.

1. When the Contact URI does not have an maddr parameter, and the hostname is not an IP Address, the Oracle Communications Session Border Controller will look for a SAG matching the hostname.
2. When the Contact URI has an maddr parameter that contains an IP address, the Oracle Communications Session Border Controller will not look for a SAG; it will use the IP Address as the target/next-hop.
3. When the Contact URI has an maddr parameter that contains a non-IP-address value, the Oracle Communications Session Border Controller will look for a SAG matching the maddr parameter value.

The above logic can be turned on by enabling sag-lookup-on-redirect in the sip-config object as shown below.

### SIP maddr Resolution Configuration

To configure the Oracle Communications Session Border Controller to perform SAG-based maddr resolution:

1. From superuser mode, use the following command sequence to access sip-config configuration mode. While in this mode, you configure SAG-based address resolution.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
```

```
ACMEPACKET(session-router) # sip-config
ACMEPACKET(sip-config) #
```

2. Use the **sag-lookup-on-redirect** parameter to enable SAG-based maddr resolution.
3. Use **done**, **exit**, and **verify-config** to complete SAG-based address resolution.

## Trust Mode

The Oracle Communications Session Border Controller supports the Calling Identity privacy requirements based on RFC 3323 and RFC 3325. The trust mode in the SIP interface determines whether the source and destination of a request is a trusted entity. With the implementation of this feature, the Oracle Communications Session Border Controller can understand and support the privacy headers and provide the capability for anonymous packets

The Oracle Communications Session Border Controller, which acts as a boundary device between the trusted platform and the untrusted Internet, understands the following headers:

- Privacy Header
- P-Asserted-Identity Header
- P-Preferred-Identity Header

Depending on the value of these headers and the mode in which the Oracle Communications Session Border Controller is being operated (B2BUA or the proxy), the appropriate actions are performed.

### About the Process

On receiving a message, the Oracle Communications Session Border Controller checks whether the message source is trusted or not. It checks the SIP interface's trust mode value and, if the source is a session agent, the session agent's trust me value. Depending on these values, the Oracle Communications Session Border Controller decides whether the request's or response's source is trusted. If it receives message from a trusted source and the message contains the P-Asserted-Identity header field, the Oracle Communications Session Border Controller passes this message to the outgoing side. The outgoing side then decides what needs to be done with this request or response.

If the request or the response is received from an untrusted source, the Privacy header value is id (privacy is requested), and the P-Asserted-Identity header field is included, the Oracle Communications Session Border Controller strips the Privacy and the P-Asserted-Identity headers and passes the request or the response to the outgoing side.

If the request or the response contains the P-Preferred-Identity header and the message source is untrusted, the Oracle Communications Session Border Controller strips the P-Preferred-Identity header from the request or the response and passes the message to the outgoing side.

If the source is trusted or privacy is not requested (the value of the Privacy Header is not id) and the request or the response contains the P-Preferred-Identity header, the Oracle Communications Session Border Controller performs the following actions:

- inserts the P-Asserted-Identity header field with the value taken from the P-Preferred-Identity header field
- deletes the P-Preferred-Identity header value
- passes this request or the response to the Outgoing side for the appropriate action, depending on the whether the destination is trusted or not

After the Oracle Communications Session Border Controller passes the request or the response to the outgoing side, it checks whether the destination is trusted by checking the SIP interface's trust mode value and the session agent's trust me value (if the destination is configured as session agent).

- The destination is trusted

The Oracle Communications Session Border Controller does nothing with the request or the response and passes it to the destination. If the P\_Asserted\_Identity headers are present, they are passed to the session agent (if the destination is configured as session agent).

- The destination is untrusted

The Oracle Communications Session Border Controller looks at the value of the Privacy header. If set to id, the Oracle Communications Session Border Controller removes all the P-Asserted-Identity headers (if present). It

## SIP Signaling Services

---

strips the Proxy-Require header if it is set to privacy. The Oracle Communications Session Border Controller also sets the From field of SIP header to Anonymous and strips the Privacy header.

If the Privacy header is set to none, the Oracle Communications Session Border Controller does not remove the P-Asserted-Identity header fields.

If there is no Privacy header field, the SD will not remove the P-Asserted-Identity headers.

To implement this feature, you need to configure the session agent's trust me parameter to enabled (if the message source is a session agent) and the SIP interface's trust mode to the appropriate value.

## Configurable Timers and Counters

SIP timers and counters can be set in the global SIP configuration, and two can be specific for individual SIP interfaces.

You can set the expiration times for SIP messages, and you can set a counter that restricts the number of contacts that the Oracle Communications Session Border Controller tries when it receives a REDIRECT. These are similar to two parameters in the global SIP configuration, trans-expire and invite-expire. You can also set a parameter that defines how many contacts/routes the Oracle Communications Session Border Controller will attempt on redirect.

## SIP Interface Configuration

To configure a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-interface) #
```

From this point, you can configure SIP interface parameters. To view all sip-interface parameters, enter a ? at the system prompt.

4. **state**—Enable or disable the SIP interface. The default is **enabled**. The valid values are:
  - enabled | disabled
5. **realm-id**—Enter the name of the realm to which the SIP interface is connected.
6. **sip-ports**—Access the **sip-ports** subelement.
7. **carriers**—Enter the list of carriers related to the SIP interface.

Entries in this field can be from 1 to 24 characters in length and can consist of any alphabetical character (Aa-Zz), numerical character (0-9), or punctuation mark (!"#\$%^&\*()+-=<>?'|{}[]@/\`'~,.\_:;) or any combination of alphabetical characters, numerical characters, or punctuation marks. For example, both 1-0288 and acme\_carrier are valid carrier field formats

8. **proxy-mode**—Enter an option for the proxy mode parameter. Valid values are:
  - **proxy**—Forward all SIP requests to selected targets.
  - **redirect**—Send a SIP 3xx redirect response with the selected target(s) in the Contact header.
  - **record-route**—Forward requests to selected target(s) and insert a Record-Route header with the Oracle Communications Session Border Controller's address. For stateless and transaction mode only.
9. **redirect-action**—Enter the value for the redirect action. Valid values are:
  - **proxy**—Send the SIP request back to the previous hop.
  - **recurse**—Recurse on the Contacts in the response.

The designated proxy action will apply to SIP 3xx responses received from non-session agents and to 3xx responses received from session agents without configured SIP Redirect message actions (for example, session agents without values for the redirect action field).

- 10. contact-mode**—Set the Contact header routing mode, which determines how the contact address from a private network is formatted.

For example, whether a maddr parameter equal to the Oracle Communications Session Border Controller's SIP proxy needs to be added to a URI present in a Contact header.

The default is **none**. The valid values are:

- **none**—The address portion of the header becomes the public address of that private realm.
- **maddr**—The address portion of the header will be set to the IP address of the Oracle Communications Session Border Controller's B2BUA.
- **strict**—The contents of the Request-URI is destroyed when a Record-Route header is present.
- **loose**—The Record-Route header is included in a Request, which means the destination of the request is separated from the set of proxies that need to be visited along the way.

- 11. nat-traversal**—Define the type of HNT enabled for SIP. The default is **none**. Valid values are:

- **none**—HNT function is disabled for SIP.
- **rport**—SIP HNT function only applies to endpoints that include the rport parameter in the Via header. HNT applies when the sent-by of the topmost VIA matches the Contact-URI host address, both of which must be different from the received Layer 3 address.
- **always**—SIP HNT applies to requests when the sent-by of the topmost VIA matches the Contact-URI host address, both of which must be different from the received Layer 3 address. (Even when the rport parameter is not present.)

- 12. nat-interval**—Set the expiration time in seconds for the Oracle Communications Session Border Controller's cached registration entry for an HNT endpoint. The default is **30**. The valid range is:

- Minimum—0
- Maximum—999999999

Oracle recommends setting the NAT interval to one-third of the NAT binding lifetime. A NAT binding lifetime is the network connection inactivity timeout. The value is configured (or hardwired) in the NAT device (firewall). This timer is used to cause the UA to send REGISTER messages frequently enough to retain the port binding in the NAT. Retaining the binding lets inbound requests to be sent through the NAT.

- 13. tcp-nat-interval**—Set the registration cache expiration time in seconds to use for endpoints behind a NAT device that register using TCP. On upgrade, the Oracle Communications Session Border Controller assigns this parameter the same value as the existing NAT interval. The default is **90**. The valid range is:

- Minimum—0
- Maximum—999999999

The Oracle Communications Session Border Controller uses the value you set for the TCP NAT interval as the expiration value passed back in SIP REGISTER (200 OK) responses to endpoints behind a NAT that register over TCP. The NAT interval value with which you are familiar from previous releases is used for endpoints behind a NAT that register over UDP. Requiring endpoints that register over TCP to send refresh requests as frequently as those registering over UDP puts unnecessary load on the Oracle Communications Session Border Controller. By adding a separate configuration for the TCP NAT interval, the load is reduced.

For upgrade and backward compatibility with Oracle Communications Session Border Controller releases prior to Release 4.1, when the tcpNatInterval is not present in the XML for a SIP interface configuration, the value of the NAT interval (natInterval) is used for the TCP NAT interval as well.

- 14. registration-caching**—Enable for use with all UAs, not just those that are behind NATs. The default is **disabled**. The valid values are:

- enabled | disabled

If enabled, the Oracle Communications Session Border Controller caches the Contact header in the UA's REGISTER request when it is addressed to one of the following:

- Oracle Communications Session Border Controller
- registrar domain value
- registrar host value

The Oracle Communications Session Border Controller then generates a Contact header with the Oracle Communications Session Border Controller's address as the host part of the URI and sends the REGISTER to the destination defined by the registrar host value.

Whether or not SIP HNT functionality is enabled affects the value of the user part of the URI sent in the Contact header:

- HNT enabled: the Oracle Communications Session Border Controller takes the user part of the URI in the From header of the request and appends a cookie to make the user unique. A cookie is information that the server stores on the client side of a client-server communication so that the information can be used in the future.
- HNT disabled: the user part of the Contact header is taken from the URI in the From header and no cookie is appended. This is the default behavior of the Oracle Communications Session Border Controller.

When the registrar receives a request that matches the address-of-record (the To header in the REGISTER message), it sends the matching request to the Oracle Communications Session Border Controller, which is the Contact address. Then, the Oracle Communications Session Border Controller forwards the request to the Contact-URI it cached from the original REGISTER message.

**15. min-reg-expire**—Set the time in seconds for the SIP interface. The value you enter here sets the minimum registration expiration time in seconds for HNT registration caching. The default is **300**. The valid range is:

- Minimum—0
- Maximum—999999999

This value defines the minimum expiration value the Oracle Communications Session Border Controller places in each REGISTER message it sends to the real registrar. In HNT, the Oracle Communications Session Border Controller caches the registration after receiving a response from the real registrar and sets the expiration time to the NAT interval value.

Some UAs might change the registration expiration value they use in subsequent requests to the value specified in this field. This change causes the Oracle Communications Session Border Controller to send frequent registrations on to the real registrar.

**16. registration-interval**—Set the Oracle Communications Session Border Controller's cached registration entry interval for a non-HNT endpoint. Enter the expiration time in seconds that you want the Oracle Communications Session Border Controller to use in the REGISTER response message sent back to the UA. The UA then refreshes its registration by sending another REGISTER message before that time expires. The default is **3600**. The valid range is:

- Minimum—0

A registration interval of zero causes the Oracle Communications Session Border Controller to pass back the expiration time set by and returned in the registration response from the registrar.

- Maximum—999999999

If the expiration time you set is less than the expiration time set by and returned from the real registrar, the Oracle Communications Session Border Controller responds to the refresh request directly rather than forwarding it to the registrar.

Although the registration interval applies to non-HNT registration cache entries, and the loosely related NAT interval applies to HNT registration cache entries, you can use the two in combination. Using a combination of the two means you can implement HNT and non-HNT architectures on the same Oracle Communications Session Border Controller. You can then define a longer interval time in the registration interval field to reduce the network traffic and load caused by excess REGISTER messages because there is no NAT binding to maintain.

**17. route-to-registrar**—Enable routing to the registrar to send all requests that match a cached registration to the destination defined for the registrar host; used when the Request-URI matches the registrar host value or the registrar domain value, not the Oracle Communications Session Border Controller’s address. Because the registrar host is the real registrar, it should send the requests back to the Oracle Communications Session Border Controller with the Oracle Communications Session Border Controller’s address in the Request-URI. The default is **disabled**. The valid values are:

- enabled | disabled

For example, you should enable routing to the registrar if your network uses a N Oracle Communications Session Border Controller and needs requests to go through its service proxy, which is defined in the registrar host field.

**18. teluri-scheme**—Enable to convert SIP URIs to tel (resources identified by telephone numbers) URIs.

If enabled, the requests generated on this SIP interface by the Oracle Communications Session Border Controller will have a tel URI scheme instead of the SIP URI scheme. Only the Request, From, and To URIs are changed to the tel scheme. After the dialog is established, the URIs are not changed. The default is **disabled**. The valid values are:

- enabled | disabled

**19. uri-fqdn-domain**—Change the host part of the URIs to the FQDN value set here. If set to enabled, and used with an FQDN domain/host, the requests generated by the Oracle Communications Session Border Controller on this SIP interface will have the host part of the URI set to this FQDN value. Only the Request, To, and From URIs are changed. After the dialog is established, the URIs are not changed.

**20. trust-mode**—Set the trust mode for the SIP interface, which is checked by the Oracle Communications Session Border Controller when it receives a message to determine whether the message source is trusted. The default is **all**. Available options are:

- **all**—Trust all SIP elements (sources and destinations) in the realm(s), except untrusted session agents. Untrusted session agents are those that have the **trust-me** parameter set to **disabled**.
- **agents-only**—Trust only trusted session agents. Trusted session agents are those that have the **trust-me** parameter set to **enabled**.
- **realm-prefix**—Trust only trusted session agents, and source and destination IP addresses that match the IP interface’s realm (or subrealm) address prefix. Only realms with non-zero address prefixes are considered.
- **registered**—Trust only trusted session agents and registered endpoints. Registered endpoints are those with an entry in the Oracle Communications Session Border Controller’s registration cache.
- **none**—Trust nothing.

Session agents must have one or more of the following:

- global realm
- same realm as the SIP interface
- realm that is a subrealm of the SIP interface’s realm

**21. trans-expire**—Set the TTL expiration timer in seconds for SIP transactions. This timer controls the following timers specified in RFC 3261:

- Timer B—SIP INVITE transaction timeout
- Timer F—non-INVITE transaction timeout
- Timer H—Wait time for ACK receipt
- Timer TEE—Used to transmit final responses before receiving an ACK

The default is **0**. If you leave this parameter set to the default, then the Oracle Communications Session Border Controller uses the timer value from the global SIP configuration. The valid range is:

- Minimum—0
- Maximum—999999999

**22. invite-expire**—Set the TTL expiration timer in seconds for a SIP client/server transaction after receiving a provisional response.

You set this timer for the client and the sever by configuring it on the SIP interface corresponding to the core or access side.

The default is **0**. If you leave this parameter set to the default, then the Oracle Communications Session Border Controller uses the timer value from the global SIP configuration. The valid range is:

- Minimum—**0**
- Maximum—**999999999**

**23. max-redirect-contacts**—Set the maximum number of contacts or routes for the Oracle Communications Session Border Controller to attempt in when it receives a SIP Redirect (3xx Response). The default is **0**. If you leave this parameter set to the default, then the Oracle Communications Session Border Controller will exercise no restrictions on the number of contacts or routes. The valid range is:

- Minimum—**0**
- Maximum—**10**

**24. response-map**—Enter the name of the SIP response map configuration that you want to apply to this SIP interfaces for outgoing responses. This parameter is blank by default.

**25. local-response-map**—Enter the name of the SIP response map configuration that you want to apply to this SIP interfaces for locally-generated SIP responses. This parameter is blank by default.

**26. options**—Optional.

### Configuring SIP Ports

To configure SIP ports:

1. From sip-interface, type **sip-ports** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(sip-interface) # sip-ports
ACMEPACKET(sip-port) #
```

2. **address**—Enter the IP address of the host associated with the sip-port entry on which to listen. For example:

```
192.168.11.101
```

3. **port**—Enter the port number you want to use for this sip-port. The default is **5060**. The valid range is:

- Minimum—**1025**
- Maximum—**65535**

4. **transport-protocol**—Indicate the transport protocol you want to associate with the SIP port. The default is **UDP**. The valid values are:

- **TCP**—Provides a reliable stream delivery and virtual connection service to applications through the use of sequenced acknowledgment with the retransmission of packets when necessary.
- **UDP**—Provides a simple message service for transaction-oriented services. Each UDP header carries both a source port identifier and destination port identifier, allowing high-level protocols to target specific applications and services among hosts.
- **TLS**—See the Security chapter for more information about configuring TLS.

5. **allow-anonymous**—Define the allow anonymous criteria for accepting and processing a SIP request from another SIP element.

The anonymous connection mode criteria includes admission control based on whether an endpoint has successfully registered. Requests from an existing SIP dialog are always accepted and processed. The default is **all**.

The following table lists the available options.

- **all**—All requests from any SIP element are allowed.
- **agents-only**—Only requests from configured session agents are allowed. The session agent must fit one of the following criteria:



- Have a global realm.
- Have the same realm as the SIP interface
- Be a sub-realm of the SIP interface's realm.

When an agent that is not configured on the system sends an INVITE to a SIP interface, the Oracle Communications Session Border Controller:

- Refuses the connection in the case of TCP.
- Responds with a 403 Forbidden in the case of UDP.
- **realm-prefix**—The source IP address of the request must fall within the realm's address prefix or a SIP interface sub-realm. A sub-realm is a realm that falls within a realm-group tree. The sub-realm is a child (or grandchild, and so on) of the SIP interface realm.

Only realms with non-zero address prefixes are considered. Requests from session agents (as described in the **agents-only** option) are also allowed.

- **registered**—Only requests from user agents that have an entry in the registration cache (regular or HNT) are allowed; with the exception of a REGISTER request. A REGISTER request is allowed from any user agent.

The registration cache entry is only added if the REGISTER is successful. Requests from configured session agents (as described in the **agents-only** option) are also allowed.

- **register-prefix**—Only requests from user agents that have an entry in the Registration Cache (regular or HNT) are allowed; with the exception of a REGISTER request. A REGISTER request is allowed only when the source IP address of the request falls within the realm address-prefix or a SIP interface sub-realm. Only realms with non-zero address prefixes are considered.

The Registration Cache entry is only added if the REGISTER is successful. Requests from configured session agents (as described in the **agents-only** option) are also allowed.

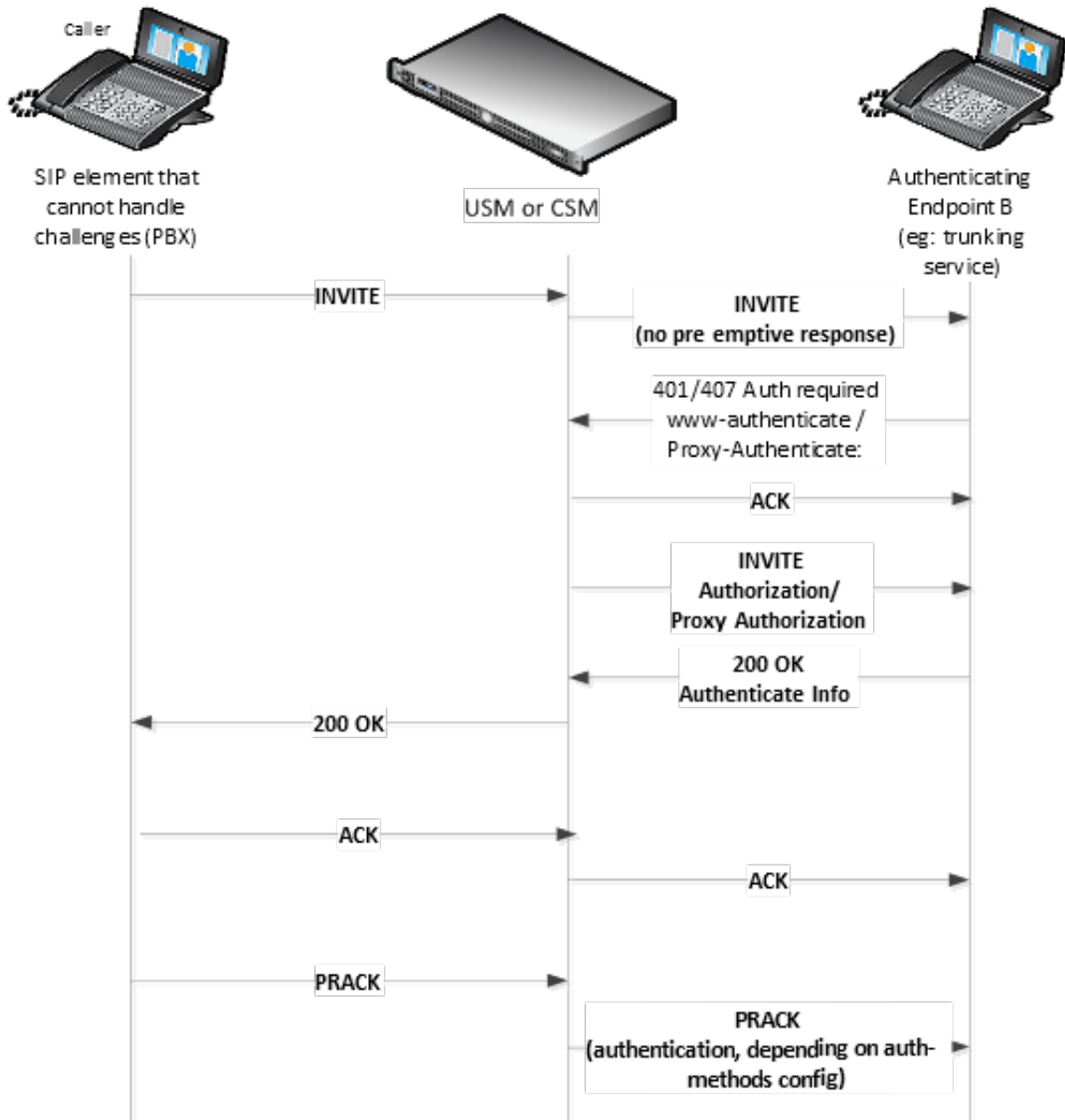
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## Digest Authentication with SIP

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Digest authentication for Session Initiation Protocol (SIP) is a type of security feature on the Oracle Communications Session Border Controller that provides a minimum level of security for basic Transport Control Protocol (TCP) and User Datagram Protocol (UDP) connections. Digest authentication verifies that both parties on a connection (host and endpoint client) know a shared secret (a password). This verification can be done without sending the password in the clear.

Digest authentication is disabled by default on the Oracle Communications Session Border Controller. When digest authentication is enabled, the Oracle Communications Session Border Controller (host) responds to authentication challenges from SIP trunking Service Providers (endpoint client). The Oracle Communications Session Border Controller performs authentication for each IP-PBX initiating the call. However, the authentication challenge process takes place between the host and the client only since the IP-PBX cannot handle authentication challenges. The following illustration shows the digest authentication process.



The digest authentication scheme is based on a simple challenge-response paradigm. A valid response contains a checksum (by default, the MD5 checksum) of the “username” and password. In this way, the password is never sent in the clear.

By default, the Oracle Communications Session Border Controller uses cached credentials for all requests within the same dialog, once the authentication session is established with a 200OK from the authenticating SIP element. If the in-dialog-methods attribute contains a value, it specifies the requests that have challenge-responses inserted within a dialog.

In digest authentication with SIP, the following can happen:

- More than one authenticating SIP element (IP-PBX) may be the destination of requests.
- More than one authentication challenge can occur in a SIP message. This can occur when there are additional authenticating SIP elements behind the first authenticating SIP element.

- The Oracle Communications Session Border Controller distinguishes whether the IP-PBX is capable of handling the challenge. If Digest Authentication is disabled (no auth-attributes configured) on the Session Agent, the challenge is passed back to the IP-PBX.
  - Note:** If there are multiple challenges in the request, and if the Oracle Communications Session Border Controller has only some of the cached credentials configured, the Oracle Communications Session Border Controller adds challenge-responses for the requests it can handle, and does not pass the challenge back to the IP-PBX.

## Challenge-Responses in Requests not in the Dialog

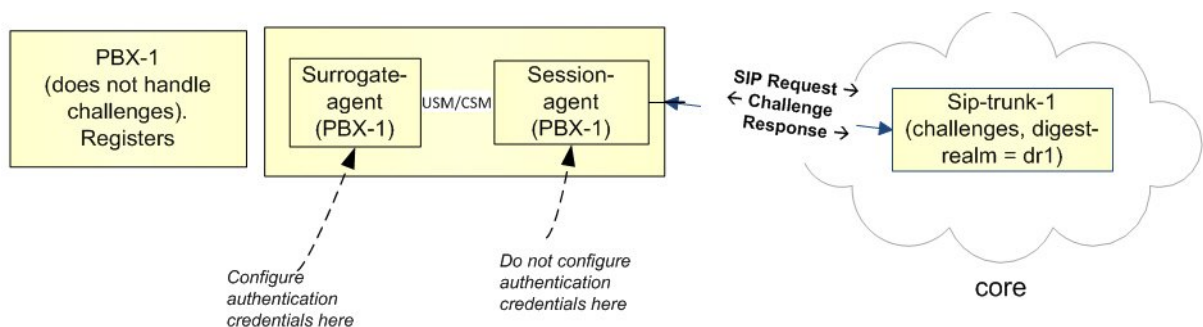
A digest authentication session starts from the client response to a www-authenticate/proxy-authenticate challenge and lasts until the client receives another challenge in the protection space defined by the auth-realm. Credentials are not cached across dialogs; however, if a User Agent (UA) is configured with the auth-realm of its outbound proxy, when one exists, the UA may cache credentials for that auth-realm across dialogs.

- Note:** Existing Oracle Communications Session Border Controller behavior with surrogate-agents is that they cache credentials from REGISTER for INVITE sessions only if the Oracle Communications Session Border Controller is considered a UA sending to its outbound proxy.

## Surrogate Agents and the Oracle Communications Session Border Controller

In the case where a surrogate-agent is configured for the IP-PBX, you do not have to configure digest authentication attributes in the session-agent object for the same IP-PBX. The surrogate-agent authentication configuration takes precedence over the session-agent authentication configuration and so it is ignored.

The following illustration shows an example of a surrogate-agent with a session-agent in the network.



## Configuring Digest Authentication

In the Oracle Communications Session Border Controller CLI, you can access the Digest Authentication object at the path session-router->session-agent->**auth-attribute**. If enabled, the Digest Authentication process uses the attributes and values listed in this table.

- Note:** If enabling Digest Authentication, all attributes listed below are required except for the in-dialog-methods attribute which is optional.

The following table lists the digest authentication object

```
ACMEPACKET (auth-attribute) # show
auth-attribute
    auth-realm          realm01
    username            user
    password            *****
    in-dialog-methods  ACK INVITE SUBSCRIBE
```

To configure digest authentication on the Oracle Communications Session Border Controller:

1. In Superuser mode, type configure terminal and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type session-router and press Enter to access the session router-related objects.

```
ACMEPACKET(configure)# session-router  
ACMEPACKET(session-router)#
```

3. Type session-agent and press Enter to access the session agent-related attributes.

```
ACMEPACKET(session-router)# session-agent  
ACMEPACKET(session-agent)#
```

4. Type auth-attribute and press Enter to access the digest authentication-related attributes.

```
ACMEPACKET(session-agent)# auth-attribute  
ACMEPACKET(auth-attribute)#
```

5. `auth-realm` — Enter the name (realm ID) of the host realm initiating the authentication challenge. This value defines the protected space in which the digest authentication is performed. Valid value is an alpha-numeric character string. Default is blank.

```
ACMEPACKET(auth-attribute)# auth-realm realm01
```

6. `username` — Enter the username of the client. Valid value is an alpha-numeric character string. Default is blank.

```
ACMEPACKET(auth-attribute)# username user
```

7. `password` — Enter the password associated with the username of the client. This is required for all LOGIN attempts. Password displays while typing but is saved in clear-text (i.e., \*\*\*\*\*). Valid value is an alpha-numeric character string. Default is blank.

```
ACMEPACKET(auth-attribute)# password *****
```

8. `in-dialog-methods` — Enter the in-dialog request method(s) that digest authentication uses from the cached credentials. Specify request methods in a list form separated by a space enclosed in parentheses. Valid values are:

- INVITE | BYE | ACK | CANCEL | OPTIONS | SUBSCRIBE | PRACK | NOTIFY | UPDATE | REFER

```
ACMEPACKET(auth-attribute)# in-dialog-methods (ack invite subscribe)
```



**Note:** The methods not in this list are still resubmitted if a 401/407 response is received by the Oracle Communications Session Border Controller.

If you do not specify any in-dialog-method value(s), digest authentication does not add challenge-responses to in-dialog requests within a dialog.

This attribute setting applies to in-dialog requests only.

## Additional Notes

The following are additional notes that describe the digest authentication process:

- The Oracle Communications Session Border Controller always challenges the first LOGIN request, and initial authentication begins with that request. The recalculated authorization key — the credentials — are then included in every subsequent request.
- If the Oracle Communications Session Border Controller does not receive any communication from the client within the expiration period, the Oracle Communications Session Border Controller logs the client out and tears down the transport connection. Faced with interface loss, the Oracle Communications Session Border Controller default behavior is to flush all warrant information from the target database. This response necessitates that the client first login/re-register with the Oracle Communications Session Border Controller, and then repopulate the empty database using a series of ADD requests. This behavior ensures that client and Oracle Communications Session Border Controller target databases are synchronized.

Alternatively, when faced with interface loss, the Oracle Communications Session Border Controller can retain all warrant information within the target database. This response necessitates only that the client first login/re-register with the Oracle Communications Session Border Controller. After successful registration the client should, but is

not required to, use a series of GET, ADD, and DELETE requests to ensure that the Oracle Communications Session Border Controller and client target databases are synchronized.

- The Oracle Communications Session Border Controller ignores the Authentication-Info header that comes in the 200OK response after digest authentication is complete. The Oracle Communications Session Border Controller receives a 401/407 response from the client. However, some surrogate-agents may process the Authentication-Info header in a single challenge.

### Digest Authentication and High Availability


The Oracle Communications Session Border Controller supports digest authentication in high availability (HA) environments. The session-agent configuration, which includes the digest authentication parameters on the primary Oracle Communications Session Border Controller, are replicated on the HA Oracle Communications Session Border Controller. However, cached credentials on the primary device are not replicated on the HA device.

## Recurse 305 Only Redirect Action

The Oracle Communications Session Border Controller has a SIP feature called redirect action. This is a feature that allows the Oracle Communications Session Border Controller, acting as a SIP Proxy or a Session Agent, to redirect SIP messages after receiving a SIP redirect (3xx) response. Previously, for the ACLI objects of sip-interface and session-agent on the Oracle Communications Session Border Controller, you could set the redirect-action parameter to **proxy** or **recurse**. In Release 6.3 you can additionally set a value of **recurse-305-only** for the redirect-action parameter.

### Redirect Action Process

When the redirect-action parameter is set to proxy, the Oracle Communications Session Border Controller sends SIP Redirect responses back to the previous hop (back to the User Agent Client (UAC)) when the User Agent Server (UAS) is not a session agent. The URI in the Contact of the response is changed from the URI that was in the original request.

 **Note:** If the target of the request is a session agent, the session agent's redirect action supercedes that of the SIP interface.

When the redirect-action parameter is set to recurse, if the Oracle Communications Session Border Controller receives a SIP redirect (3xx) response on the SIP interface, it automatically redirects all requests to the Contact URI specified in the 3xx response. The responses contain the same Contact URI that was in the original request sent to the UAS.

For example, if UAC X sends an INVITE to the Oracle Communications Session Border Controller set up as a SIP proxy, the Oracle Communications Session Border Controller forwards the INVITE to UAS Y (Y is not a session agent). Y then responds to the Oracle Communications Session Border Controller with a 3xx response (redirection message) with the same URI that was in the original request. This indicates to the Oracle Communications Session Border Controller that if it receives any future requests directed toward Y, that it should automatically redirect the request directly to Y. The Oracle Communications Session Border Controller then recurses, or repeatedly sends subsequent incoming messages to the Contact URI specified in the Header of the 3xx responses.

When the redirect-action parameter is set to recurse-305-only, if the Oracle Communications Session Border Controller receives a 305 SIP redirect response (Use Proxy) on the SIP interface, it automatically redirects all requests to the Contact URI specified in the 305 response. All other 3xx responses are sent back to the previous hop.

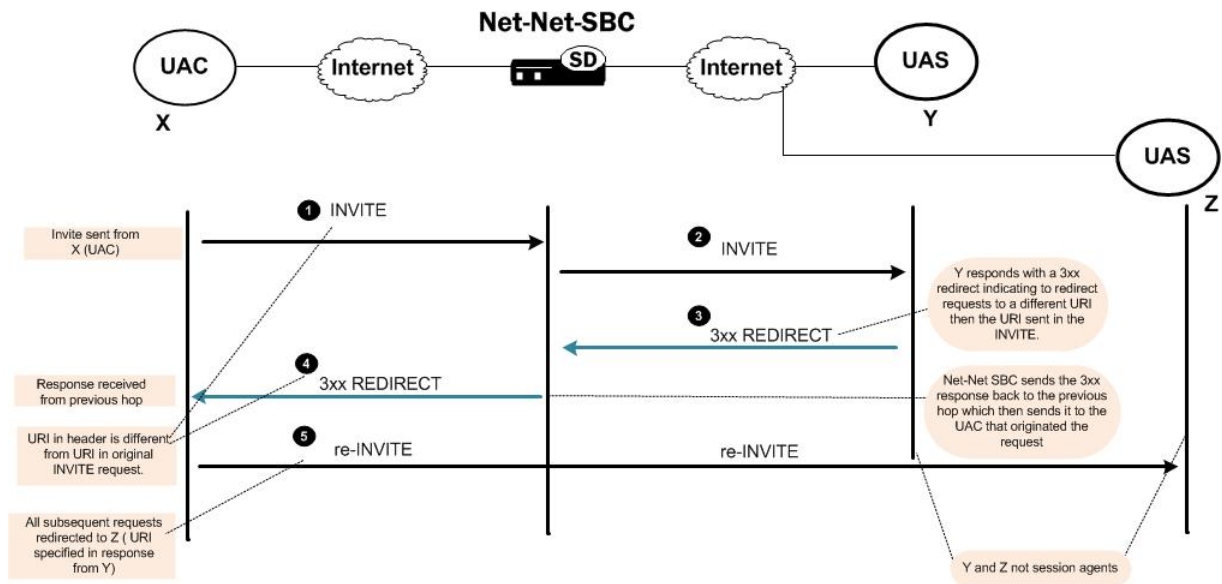
When the UAS is a session agent, the Oracle Communications Session Border Controller can send the SIP redirect response back to the UAC using the value in the session agent's redirect action field. If there are too many UASs to define as individual session agents, or if the UASs are Hosted NAT Traversal (HNT) endpoints, and SIP redirect responses need to be proxied for UASs that are not session agents, you can set the behavior at the SIP interface level.

## Redirect-Action Set to Proxy

The following occurs if you set the **redirect-action** parameter to proxy on the Oracle Communications Session Border Controller:

1. X (UAC) sends an INVITE to the Oracle Communications Session Border Controller.
2. The Oracle Communications Session Border Controller forwards the INVITE to Y (UAS).
3. Y sends the 3xx REDIRECT response to the Oracle Communications Session Border Controller with a different URI in the message header.
4. The Oracle Communications Session Border Controller forwards the 3xx REDIRECT response to the previous hop. X receives the 3xx REDIRECT response from the previous hop.
5. X redirects all subsequent requests to the URI in the message header received from Y.

The following illustration shows an example of a dialog between X, Y, Z, and the Oracle Communications Session Border Controller during a redirect-action session set to proxy.

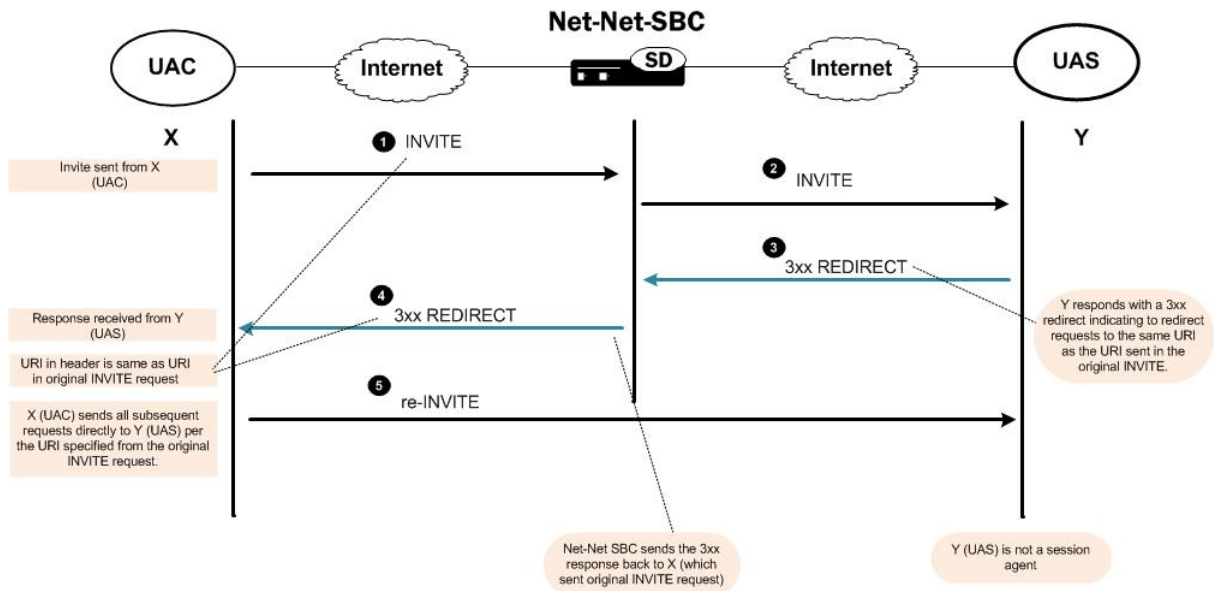


## Redirect-Action Set to Recurse

The following occurs if you set the **redirect-action** parameter to recurse on the Oracle Communications Session Border Controller:

1. X (UAC) sends an INVITE to the Oracle Communications Session Border Controller.
2. The Oracle Communications Session Border Controller forwards the INVITE to Y (UAS).
3. Y sends the 3xx REDIRECT response to the Oracle Communications Session Border Controller with the same URI as the URI sent in the original request.
4. The Oracle Communications Session Border Controller forwards the 3xx REDIRECT response to X (UAC).
5. X (UAC) sends all subsequent requests directly to Y (UAS) per the URI specified from the original INVITE request.

The following illustration shows an example of a dialog between X, Y, and the Oracle Communications Session Border Controller during a redirect-action session set to recurse.

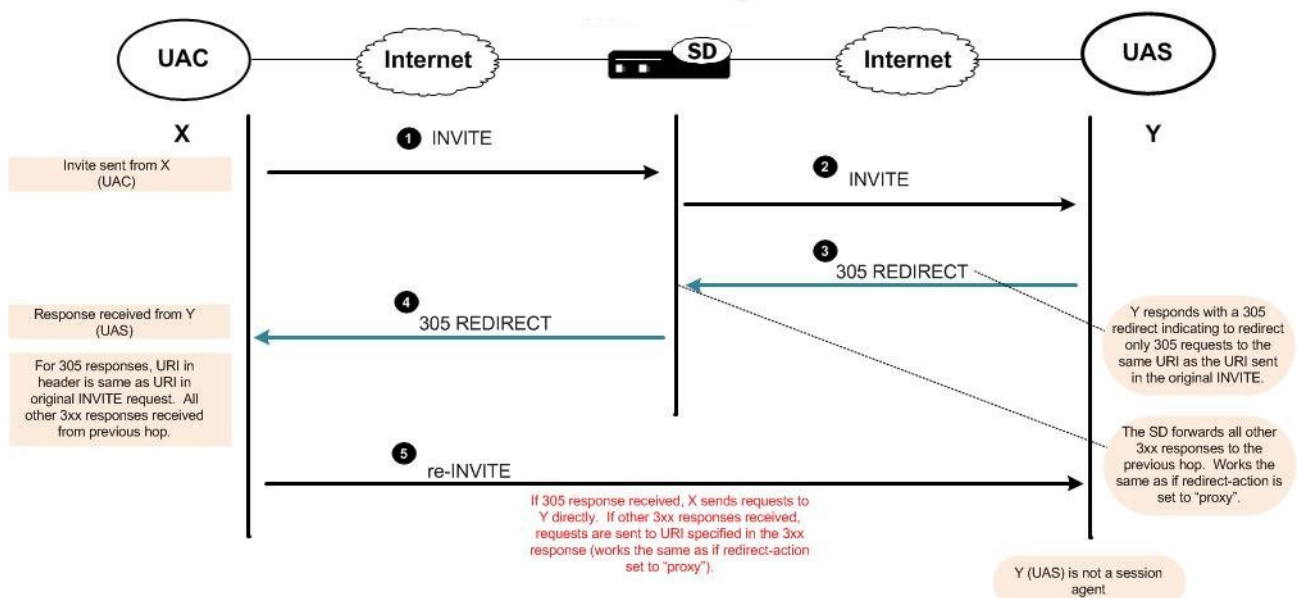


### Redirect-Action Set to Recurse-305-Only

The following occurs if you set the **redirect-action** parameter to **recurse-305-only** on the Oracle Communications Session Border Controller:

1. X (UAC) sends an INVITE to the Oracle Communications Session Border Controller.
2. The Oracle Communications Session Border Controller forwards the INVITE to Y (UAS).
3. Y sends a 305 REDIRECT response to the Oracle Communications Session Border Controller with the same URI as the URI sent in the original request.
4. The Oracle Communications Session Border Controller forwards the 305 REDIRECT response to X (UAC).
5. If 305 response received, X sends requests to Y directly. If other 3xx responses received, requests are sent to URI specified in the 3xx response (works the same as if **redirect-action** set to **proxy**).

The following illustration shows an example of a dialog between X, Y, and the Oracle Communications Session Border Controller during a redirect-action session set to **recurse-305-only**.





### Redirect Configuration for SIP Interface

You can configure the Oracle Communications Session Border Controller to redirect requests from a UAC to a UAS using the URI in 305 responses only. You can use the ACLI at the paths **session-router > sip-interface** or **session-router > session-agent**.

To configure the redirect-action feature on the SIP interface on the Oracle Communications Session Border Controller:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-interface** and press Enter to access the SIP interface-related configurations. The system prompt changes to let you know that you can begin configuring individual parameters for this object.

```
ACMEPACKET(session-router)# sip-interface  
ACMEPACKET(sip-interface)#
```

4. Enter **redirect-action** followed by the following value:

- recurse-305-only

```
ACMEPACKET(sip-interface)# redirect-action recurse-305-only
```

When the Oracle Communications Session Border Controller receives a 305 SIP redirect response (Use Proxy) on the SIP interface, it automatically redirects all requests to the Contact URI specified in the 305 response. All 3xx responses other than 305 responses are sent back to the previous hop.

To disable this feature, enter **redirect-action** and press Enter without entering a value.

```
ACMEPACKET(sip-interface)# redirect-action
```

### Redirect Configuration for Session Agent

To configure the redirect-action feature for a session agent on the Oracle Communications Session Border Controller:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-agent** and press Enter to access the SIP session agent-related configurations. The system prompt changes to let you know that you can begin configuring individual parameters for this object.

```
ACMEPACKET(session-router)# session-agent  
ACMEPACKET(session-agent)#
```

4. Enter **redirect-action** followed by the following value:

- recurse-305-only

```
ACMEPACKET(session-agent)# redirect-action recurse-305-only
```

When the Oracle Communications Session Border Controller receives a 305 SIP redirect response (Use Proxy) on the SIP interface, it automatically redirects all requests to the Contact URI specified in the 305 response. All 3xx responses other than 305 responses are sent back to the previous hop.

To disable this feature, enter **redirect-action** and press Enter without entering a value.

```
ACMEPACKET(sip-interface)# redirect-action
```



## Embedded Routes in Redirect Responses

When the Oracle Communications Session Border Controller recurses as the result of a redirect (3xx) response, the server might need to specify one or more intermediate hops. These hops are reflected in the Contact header for the 3xx response using embedded route headers and look like this:

```
Contact: <sip:touser@server.example.com?Route=%3Cproxy.example.com%Blr%3E>
```

The Contact header shows that the request should be sent to server.example.com using proxy.example.com.

You can configure your Oracle Communications Session Border Controller to specify that embedded headers in 3xx Contact headers are to be included in new requests such that they are tied to a session agent representing the new target (server.example.com). This behavior requires you to set the **request-uri-headers** parameter.

However, you can also use the **use-redirect-route** in global SIP configuration's options parameter so that the embedded Route header is used as the next hop to receive the new request.

When you configure this new option, the Oracle Communications Session Border Controller constructs a new request using the redirect Contact, and the SIP URI from the Contact becomes the Request-URI. Then, the system inserts the embedded routes as Route headers in the, using the same order in which they appeared in the redirect Contact. Afterward, the Oracle Communications Session Border Controller determines the next hop in the same way it does with any other request. If the first route is a loose route (i.e., it has the lr URI parameter), then the Oracle Communications Session Border Controller sends a request to host indicated in the first route. Otherwise, strict routing applies, and the Oracle Communications Session Border Controller sends the request to the host indicated in the Request-URI.

1. In Superuser mode, type configure terminal and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure)# session-router  
ACMEPACKET (session-router)#
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET (session-router)# sip-config  
ACMEPACKET (sip-config)#
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the ACLI **select** command) the configuration that you want to edit.

4. **options**—Set the options parameter by typing options, a Space, and then the option name.

```
ACMEPACKET (sip-config)# options use-redirect-route
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

## SIP PRACK Interworking


When you configure your Oracle Communications Session Border Controller with PRACK interworking for SIP, you enable it to interwork between endpoints that support RFC 3262, *Reliability of Provisional Responses in the Session Initiation Protocol*, and those that do not.

As its title indicates, RFC 3262 defines a reliable provisional response extension for SIP INVITEs, which is the 100rel extension tag. While some endpoints do not support the RFC, other SIP implementations require compliance with it. A session setup between two such endpoints fails. However, you can configure your Oracle Communications Session Border Controller to supply the provisional response on behalf of endpoints that do not support it—and thereby enable sessions between those endpoints and the ones requiring RFC 3262 compliance.

## SIP Signaling Services

You need to configure PRACK interworking for a SIP interface associated with the endpoints that need RFC 3262 support. To enable the feature, you set the **100rel-interworking** option. The Oracle Communications Session Border Controller applies PRACK interworking for either the UAC or the UAS. The Oracle Communications Session Border Controller checks to see whether or not it needs to apply PRACK interworking when an INVITE arrives at the ingress or egress SIP interface with the option enabled. First, it checks the Require header for the 100rel tag; if not found there, it checks the Supported header.

Since there is a slight difference in the application of this feature between the UAC and UAS, this section explains both.

 **Note:** If SDP is included in a PRACK request sent to a SIP interface where PRACK interworking is enabled, it will not be responded to, nor will any SDP be included in the locally-generated 200 OK to that PRACK.

### UAC-Side PRACK Interworking

The Oracle Communications Session Border Controller applies PRACK interworking on the UAC side when:

- A SIP INVITE does not contain a 100rel tag in a Require or Supported header
- The ingress SIP interface is enabled with the **100rel-interworking** option
- The UAS fails to send reliable provisional responses

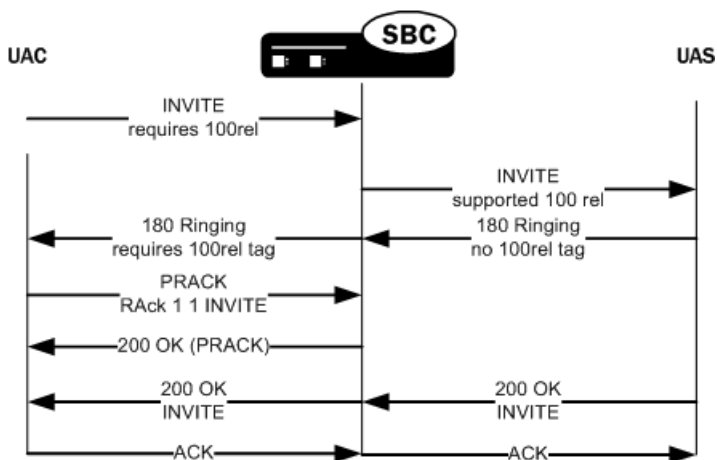
When it is to forward a non-reliable response to a UAC that requires RFC 3262 support, the Oracle Communications Session Border Controller converts the non-reliable response to a reliable one by adding the 100rel tag to the Require header and adding an Rseq header to the response. Further, the Oracle Communications Session Border Controller adds a Require header (complete with the 100rel tag) if there is not one already in the response, and then also adds Rseq header.

Note that the Oracle Communications Session Border Controller sets the value of the Rseq header as 1 for the first provisional response, and then increments it by 1 for each subsequent provisional response. It also adds the PRACK method to the Allow header when that header appears.

The Oracle Communications Session Border Controller retransmits the converted reliable provisional response in accordance with RFC 3262, until it receives a PRACK request. For the initial timeout for retransmission, the Oracle Communications Session Border Controller uses the value you set in the **init-timer** parameter in the global SIP configuration. It stops retransmitting when either it receives a transmission, or when the ingress SIP interface's transmit timer elapses.

If it never receives a PRACK, the Oracle Communications Session Border Controller does not generate an error response to the INVITE, relying instead on the downstream UAS to produce a final response.

The call flow for this application looks like this:



### UAS-Side PRACK Interworking

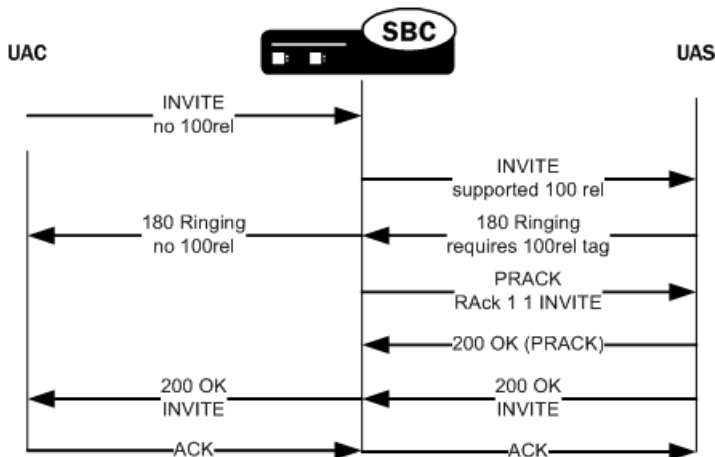
The Oracle Communications Session Border Controller applies PRACK interworking on the UAS side when:

- A SIP INVITE contains the 100rel tag in a Require or Supported header
- The egress SIP interface is enabled with the **100rel-interworking** option
- The UAS does send reliable provisional responses

When the UAC does not support RFC 3262, the Oracle Communications Session Border Controller generates a PRACK request to acknowledge the response. It also converts the response to non-reliable by removing the 100 rel tag from the Require header and removing the RSeq header from the response.

In the case of the UAS, the Oracle Communications Session Border Controller matches the PRACK to a converted reliable provisional response using the PRACK's RACK header. If it finds a matching response, the Oracle Communications Session Border Controller generates a 200 OK to the PRACK. And if it finds no match, then it generates a 481 Call Leg/Transaction Does Not Exist response. The Oracle Communications Session Border Controller generates a 400 Bad Request response if either the RACK is not in the PRACK request or it is not formatted properly.

The call flow for this application looks like this:



## PRACK Interworking Configuration

You enable PRACK interworking for ingress and egress SIP interfaces. Be sure you know on what side, ingress or egress, you need this feature applied.

To configure PRACK interworking for a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter. If you are editing an existing configuration, select the one on which you want to enable this feature.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. **options**—Set the options parameter by typing **options**, a Space, the option name **100rel-interworking** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(sip-interface)# options +100rel-interworking
```

**If you type options** and then the option value for either of these entries **without the plus sign**, you will overwrite any previously configured options. In order to append the new option to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

## Global SIP Timers

---

This section explains how to configure SIP retransmission and expiration timers.



**Note:** you can also set timers and counters per SIP interface.

### Overview

SIP timers define the transaction expiration timers, retransmission intervals when UDP is used as a transport, and the lifetime of dynamic TCP connections. The retransmission and expiration timers correspond to the timers defined in RFC 3261.

- **init timer:** is the initial request retransmission interval. It corresponds to Timer T1 in RFC 3261.

This timer is used when sending requests over UDP. If the response is not received within this interval, the request is retransmitted. The retransmission interval is doubled after each retransmission.

- **max timer:** is the maximum retransmission interval for non-INVITE requests. It corresponds to Timer T2 in RFC 3261.

The retransmission interval is doubled after each retransmission. If the resulting retransmission interval exceeds the max timer, it is set to the max timer value.

- **trans expire:** is the transaction expiration timer. This value is used for timers B, D, F, H and J as defined in RFC 3261.
- **invite expire:** defines the transaction expiration time for an INVITE transaction after a provisional response has been received. This corresponds to timer C in RFC 3261.

If a final response is not received within this time, the INVITE is cancelled. In accordance with RFC 3261, the timer is reset to the invite expire value when any additional provisional responses are received.

- **Inactive dynamic conn timer** defines the idle time of a dynamic TCP connection before the connection is torn down. Idle is defined as not transporting any traffic. There is no timer in RFC 3261 corresponding to this function.

### Timers Configuration

To configure timers:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-config  
ACMEPACKET(sip-config)#
```

4. **init-timer**—Enter the initial timeout value in milliseconds for a response to an INVITE request, and it applies to any SIP request in UDP. In RFC 3261, this value is also referred to as `TIMER_T1`. The default is **500**. The valid range is:

- Minimum—0
- Maximum—999999999

5. **max-timer**—Enter the maximum transmission timeout (T2) for SIP in milliseconds.

When sending SIP over UDP, a re-transmission timer is used. If the timer expires and the message is re-transmitted, the re-transmission timer is then set to twice the previous value (but will not exceed the maximum

timer value). Using the default values of 500 milliseconds and 4000 milliseconds, the re-transmission timer is 0.5, then 1, 2, and finally 4. The incrementing continues until the transmission expire timer activates. The default is **4000**. The valid range is:

- Minimum—0
  - Maximum—999999999
6. **trans-expire**—Enter the transaction expire timeout value (Timer B) in seconds to set the time for SIP transactions to live. The same value is used for Timers D, F, H and J. The default is **32**. The valid range is:
- Minimum—0
  - Maximum—999999999
7. **invite-expire**—Enter the invite expire timeout value (Timer C) in seconds to indicate the time for SIP client transaction will live after receiving a provisional response. The default is **180**. The valid range is:
- Minimum—0
  - Maximum—999999999
8. **inactive-dynamic-conn**—Enter the inactive dynamic connection value in seconds to set the time limit for inactive dynamic connections.

If the connection between the SIP proxy and a session agent is dynamic (for example, through dTCP), and the connection has been idle for the amount of time specified here, the SIP proxy breaks the connection. Idle is defined as not transporting any traffic. The default value is **32**. The valid range is:

- Minimum—0
- Maximum—999999999



**Note:** Setting this parameter to 0 disables this parameter.

The following example shows SIP config timer values for a peering network. Some parameters are omitted for brevity.

```

sip-config
    state                enabled
    operation-mode       dialog
dialog-transparency    disabled
home-realm-id          acme
    egress-realm-id
nat-mode                Public
registrar-domain
registrar-host
registrar-port         0
init-timer              500
max-timer               4000
trans-expire            32
invite-expire           180
inactive-dynamic-conn   32

```

## SIP Timers Discreet Configuration

Previous releases controlled various SIP timers with a single ACLI command, **trans-expire**, available in both sip-config and sip-interface modes. When executed in sip-config mode, the command essentially established a global default transaction expiration timer value. Executed at the sip-interface level, the command established a local, interface-specific value that overrode the global default.

Specific timers controlled by **trans-expire** are as follows:

Timer B, the INVITE transaction timeout timer, defined in Section 17.1.1.2 and Appendix A of RFC 3261, *SIP: Session Initiation Protocol*.

## SIP Signaling Services

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Timer D, the Wait-Time for response retransmits timer, defined in Section 17.1.1.2 and Appendix A of RFC 3261, *SIP: Session Initiation Protocol*.

Timer F, the non-INVITE transaction timeout timer, defined in Section 17.1.2.2 and Appendix A of RFC 3261, *SIP: Session Initiation Protocol*.

Timer H, the Wait-Time for ACK receipt timer, defined in Section 17.2.1 and Appendix A of RFC 3261, *SIP: Session Initiation Protocol*.

Timer J, the Wait-Time for non-INVITE requests timer, defined in Section 17.2.2 and Appendix A of RFC 3261, *SIP: Session Initiation Protocol*.

A new ACLI command (**initial-inv-trans-expire**) that enables user control over SIP Timer B for initial INVITE transactions. Other timers, namely B for non-initial INVITES, D, F, H, and J remain under the control of **trans-expire**.

Use **initial-inv-trans-expire** in the sip-config configuration mode, to establish a global, default transaction timeout value (expressed in seconds) used exclusively for initial INVITE transactions.

```
ACMEPACKET(sip-config) # initial-inv-trans-expire 4  
ACMEPACKET(sip-config) #
```

Allowable values are integers within the range 0 (the default) through 999999999. The default value, 0, indicates that a dedicated INVITE Timer B is not enabled. Non-default integer values enable a dedicated Timer B and set the timer value.

The default value retains compatibility with previous operational behavior in that Timers B, D, F, H, and J all remain subject to the single timer value set by **trans-expire**. However, when **initial-inv-trans-expire** is set to a supported non-zero value, SIP Timer B as it applies to initial INVITES, assumes that value rather than the value assigned by **trans-expire**. This functionality is available in both sip-config and in sip-interface objects.

If a dedicated Timer B is enabled at the sip-config level, you can use **initial-inv-trans-expire** in the sip-interface configuration mode, to establish a local interface-specific Timer B timeout value that overrides the global default value.

```
ACMEPACKET(sip-interface) # initial-inv-trans-expire 8  
ACMEPACKET(sip-interface) #
```

## Session Timer Support

---

The Oracle Communications Session Border Controller partially supports RFC4028 by establishing session timers without participating in the session timer negotiation.

When a 2xx response to a Session Refresh Request arrives, the Oracle Communications Session Border Controller will start a new timer or refresh the existing timer using the value of the Session-Expires header. When the session timer expires, the Oracle Communications Session Border Controller will send a BYE to both the upstream and downstream endpoints.

When accounting is configured, the Oracle Communications Session Border Controller will also send a RADIUS STOP record with Acct-Terminate-Cause=Session-Timeout.

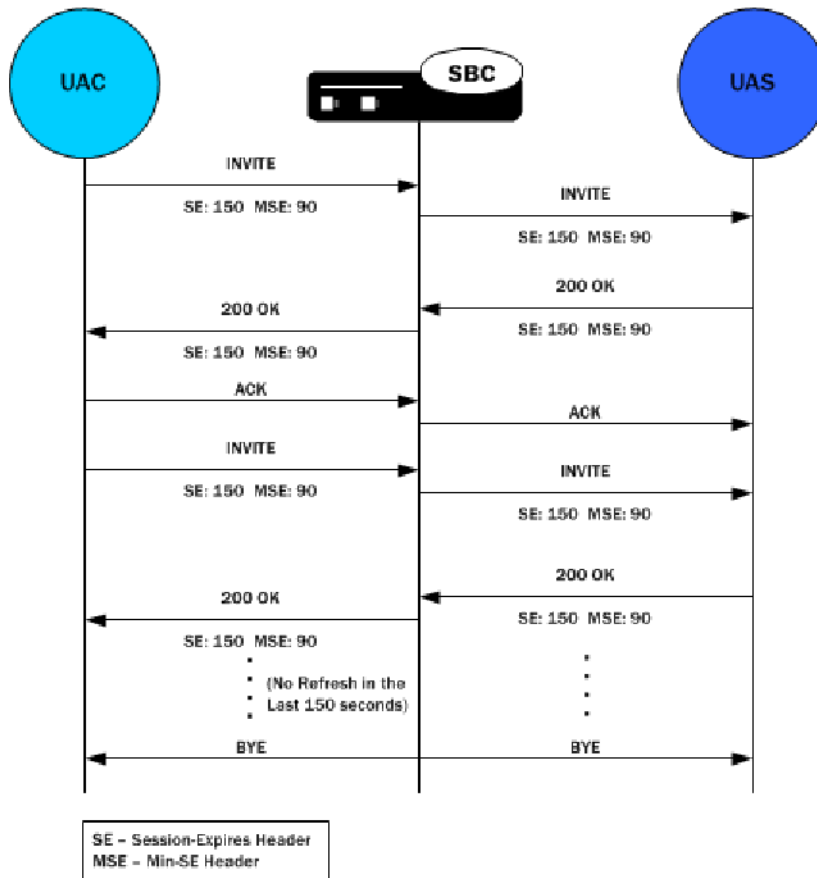
## Call Flow Example

The UAS obtains the value from the Session-Expires header field in a 2xx response to a session refresh request that it sends.

Proxies and UACs obtain this value from the Session-Expires header field in a 2xx response to a session refresh request that they receive.

Once the session timer runs out, the Oracle Communications Session Border Controller sends a BYE to both the UAC and the UAS to clear the session.

Enable this feature by adding the **session-timer-support** option to the sip config.



## SIP Per-User CAC

The Oracle Communications Session Border Controller's call admission control (CAC) supports an enhanced degree of granularity for SIP sessions.

Without this feature enabled, the Oracle Communications Session Border Controller performs call admission control (CAC) based on:

- Bandwidth limits configured in realms and nested realms
- Number of media flows available through the steering pool per realm
- Number of inbound sessions configured for a SIP session agent
- Number of total sessions (inbound and outbound) per SIP session agent
- Use of the Oracle Communications Session Border Controller's support for common open policy service (COPS), allowing the Oracle Communications Session Border Controller to perform CAC based on the policies hosted in an external policy server

These methods provide a basic level of call admission control in order to ensure that a SIP session agent's capacity is not exceeded. You can also ensure that signaling and media bandwidth capacities are not exceeded for physical trunks and peers.

With this feature enabled, the Oracle Communications Session Border Controller changes behavior so that it will only allow the configured number of calls or total bandwidth to and from each user in a particular realm. The overall realm bandwidth and steering pool limits still apply, and as before, the Oracle Communications Session Border Controller still rejects users who might be within their CAC limitations if accepting them with exceed the bandwidth limitations for parent or child realms and steering pools.

For SIP sessions, the Oracle Communications Session Border Controller now keeps track of the amount of bandwidth a user consumes and the number of active sessions per address of record (AoR) or per IP address, depending on the

## SIP Signaling Services

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CAC mode you select (either aor or ip). When an endpoint registers with the Oracle Communications Session Border Controller, the Oracle Communications Session Border Controller allots it a total amount of bandwidth and total number of sessions.

This section describes the details of how SIP per user CAC works.

You should note that the functionality this section describes only works if you enable registration caching on your Oracle Communications Session Border Controller.

For SIP sessions, the Oracle Communications Session Border Controller now keeps track of the amount of bandwidth a user consumes and the number of active sessions per address of record (AoR) or per IP address, depending on the CAC mode you select (either aor or ip). When an endpoint registers with the Oracle Communications Session Border Controller, the Oracle Communications Session Border Controller allots it a total amount of bandwidth and total number of sessions.

### Per User CAC Modes

There are three modes that you can set for this feature, and each has an impact on how the other two per-user-CAC parameters are implemented:

- none—No per user CAC is performed for users in the realm.
- aor—The Oracle Communications Session Border Controller performs per user CAC according to the AoR and the contact associated with that AoR for users in the realm.
- ip—The Oracle Communications Session Border Controller performs per user CAC according to the IP address and all endpoints that are sending REGISTER messages from the IP address for users in the realm.

### Per User CAC Sessions

You can set the number of CAC for sessions per user in the realm configuration. Depending on the CAC mode you set, the sessions are shared between contacts for the same AoR or the endpoints behind the same IP address.

When it receives an INVITE, the Oracle Communications Session Border Controller determines the registration entry for the calling endpoint and the registration for the called endpoint. It then decides if session can be established between the two. If it can, the Oracle Communications Session Border Controller establishes the session and changes the active session count for the calling and called endpoints. The count is returned to its original value once the session is terminated.

### Per User CAC Bandwidth

You can set the per user CAC bandwidth in realm configuration, too, and it is handled much the same way that the sessions are handled. That is, depending on the CAC mode you set, the bandwidth is shared between contacts for the AoR or the endpoints behind the same IP address. All endpoints must be registered with the Oracle Communications Session Border Controller.

When it receives a Request with SDP, the Oracle Communications Session Border Controller checks to see if there is enough bandwidth for the calling endpoint and for the called endpoint. The Oracle Communications Session Border Controller assumes that the bandwidth usage is symmetric, and it uses the maximum bandwidth configured for the codec that it finds in the Request. In the event that there are multiple streams, the Oracle Communications Session Border Controller determines the total bandwidth required for all of the streams. If the required bandwidth exceeds what is available for either endpoint, the Oracle Communications Session Border Controller rejects the call (with a 503 error response). If the amount of available bandwidth is sufficient, then the used bandwidth value is increased for both the registered endpoints: calling and called. Any mid-session requests for changes in bandwidth, such as those caused by modifications in codec use, are handled the same way.

The Oracle Communications Session Border Controller also keeps track of the bandwidth usage on a global level. When the call terminates, the bandwidth it was consuming is returned to the pool of available bandwidth.



## Notes on HA Nodes

This feature has been implemented so that a newly active system is able to perform SIP per user CAC. The standby Oracle Communications Session Border Controller is updated with the appropriate parameters as part of the SIP session update.

## SIP per User CAC Configuration

Note that you must enable registration caching for this feature to work.

To configure SIP per user CAC:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

4. Select the realm where you want to want to add SIP per user CAC.

```
ACMEPACKET (realm-config) # select
```

5. **user-cac-mode**—Set this parameter to the per user CAC mode that you want to use. The default value is **none**. The valid values are:

- **none**—No user CAC for users in this realm
- **aor**—User CAC per AOR
- **ip**—User CAC per IP

6. **user-cac-sessions**—Enter the maximum number of sessions per user for dynamic flows to and from the user. The default is **0**. Leaving this parameter set to its means that there is unlimited sessions, meaning that the per user CAC feature is disabled in terms of the constraint on sessions. The valid range is:

7. Minimum—**0**

8. Maximum—**999999999**

9. **user-cac-bandwidth**—Enter the maximum bandwidth per user for dynamic flows to and from the user. The default is **0** and leaving this parameter set to the default means that there is unlimited bandwidth, meaning that the per user CAC feature is disabled in terms of the constraint on bandwidth. The valid range is:

- Minimum—**0**
- Maximum—**999999999**


## SIP Per-Realm CAC

Building on the Oracle Communications Session Border Controller's pre-existing call admission control methods, CAC can be performed based on how many minutes are being used by SIP or H.323 calls per-realm for a calendar month.

In the realm configuration, you can now set a value representing the maximum number of minutes to use for SIP and H.323 session using that realm. Although the value you configure is in minutes, the Oracle Communications Session Border Controller performs CAC based on this value to the second. When you use this feature for configurations with nested realms, the parent realm will have the total minutes for all its child realms (i.e., at least the sum of minutes configured for the child realms).

The Oracle Communications Session Border Controller calculates the number of minutes used when a call completes, and counts both call legs for a call that uses the same realm for ingress and egress. The total time attributed to a call is the amount of time between connection (SIP 200 OK) and disconnect (SIP BYE), regardless of whether media is released or not; there is no pause for calls being placed on hold.

If the number of minutes is exhausted, the Oracle Communications Session Border Controller rejects calls with a SIP 503 Service Unavailable message (including additional information “monthly minutes exceeded”). In the event that the limit is reached mid-call, the Oracle Communications Session Border Controller continues with the call that pushed the realm over its threshold but does not accept new calls. When the limit is exceeded, the Oracle Communications Session Border Controller issues an alarm and sends out a trap including the name of the realm; a trap is also sent when the alarm condition clears.

 **Note:** The Oracle Communications Session Border Controller does not reject GETS/NSEP calls based on monthly minutes CAC.

You can change the value for minutes-based CAC in a realm configuration at any time, though revising the value downward might cause limits to be reached. This value resets to zero (0) at the beginning of every month, and is checkpointed across both system in an HA node. Because this data changes so rapidly, however, the value will not persist across and HA node if both systems undergo simultaneous failure or reboot.

You can use the ACLI **show monthly minutes <realm-id>** command (where **<realm-id>** is the realm identifier of the specific realm for which you want data) to see how many minutes are configured for a realm, how many of those are still available, and how many calls have been rejected due to exceeding the limit.

## SIP per Realm CAC Configuration

This section shows you how to configure minutes-based CAC for realms and how to display minutes-based CAC data for a specific realm.

### Enabling Realm-Based CAC

Note that setting the new monthly-minutes parameters to zero (0), or leaving it set to its default of 0, disables this feature.

To configure minutes-based CAC:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. Select the realm where you want to want to add SIP per user CAC.

```
ACMEPACKET(realm-config)# select
```

5. **monthly-minutes**—Enter the number of minutes allowed during a calendar month in this realm for SIP and H.323 calls. By default, this parameter is set to zero (0), which disabled monthly minutes-based CAC. You can enter a value as high as 71582788.

6. Save and activate your configuration.

### Viewing Realm-Based CAC Data

Use the ACLI **show monthly-minutes** command to see the following information:

- How many minutes are configured for a realm

- How many of those are still available
- How many calls have been rejected due to exceeding the limit

To view information about SIP per user CAC using the IP address mode:

In either User or Superuser mode, type **show monthly-minutes <realm-id>**, a Space, and the IP address for which you want to view data. Then press Enter. The **<realm-id>** is the realm identifier for the realm identifier of the specific realm for which you want data

```
ACMEPACKET# show monthly-minutes private_realm
```

## SIP Options Tag Handling

This section explains how to configure SIP options on a global or per-realm level and how to specify whether the feature treatment applies to traffic inbound to or outbound from a realm, or both.

SIP extensions that require specific behavior by UAs or proxies are identified by option tags. Option tags are unique identifiers used to designate new options (for example, extensions) in SIP. These option tags appear in the Require, Proxy-Require, and Supported headers of SIP messages.

Option tags are compatibility mechanisms for extensions and are used in header fields such as Require, Supported, Proxy-Require, and Unsupported in support of SIP.

The option tag itself is a string that is associated with a particular SIP option (i.e., an extension). It identifies this option to SIP endpoints.

### Overview

The SIP specification (RFC 3261) requires that the Oracle Communications Session Border Controller B2BUA reject any request that contains a Require header with an option tag the Oracle Communications Session Border Controller does not support. However, many of these extensions operate transparently through the Oracle Communications Session Border Controller's B2BUA. You can configure how SIP defines the Oracle Communications Session Border Controllers B2BUA treatment of specific option tags.

Also, there might be certain extensions that an endpoint indicates support for by including the option tag in a Supported header. If you do not want a given extension used in your network, the you can configure SIP option tag handling to remove the undesired option tag from the Supported header. You can also specify how option tags in Proxy-Require headers are to be treated.

### Configuration Overview

You configure the SIP feature element to define option tag names and their treatment by the Oracle Communications Session Border Controller when the option tag appears in a Supported header, a Require header, and a Proxy-Require header. If an option tag is encountered that is not configured as a SIP feature, the default treatments apply. You only need to configure option tag handling in the SIP feature element when non-default treatment is required.

You can specify whether a SIP feature should be applied to a specific realm or globally across realms. You can also specify the treatment for an option based on whether it appears in an inbound or outbound packet. Inbound packets are those that are coming from a realm to the Oracle Communications Session Border Controller and outbound packets are those which are going from the Oracle Communications Session Border Controller to the realm.

The following tables lists the SIP option tag parameters you need to configure.

Parameter	Description
name	SIP feature tag name
realm	Realm name with which the feature will be associated. To make the feature global, leave the field empty.
support mode inbound	Action for tag in Supported header in an inbound packet.

Parameter	Description
require mode inbound	Action for tag in Require header in an inbound packet
proxy require mode inbound	Action for tag in Proxy-Require header in an inbound packet
support mode outbound	Action for tag in Supported header in an outbound packet
require mode outbound	Action for tag in Require header in an outbound packet
proxy require mode outbound	Action for tag in Proxy-Require header in an outbound packet

### SIP Option Tag Handling Configuration

To configure SIP option tag handling:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-feature** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-feature
ACMEPACKET(sip-feature)#
```

From this point, you can configure SIP option tags parameters. To view all sip-feature parameters, enter a ? at the system prompt.

4. **name**—Enter a name for the option tag that will appear in the Require, Supported, or Proxy-Require headers of inbound and outbound SIP messages.

You must enter a unique value.



**Note:** Valid option tags are registered with the IANA Protocol Number Assignment Services under Session Initiation Protocol Parameters. Because option tags are not registered until the SIP extension is published as a RFC, there might be implementations based on Internet-Drafts or proprietary implementations that use unregistered option tags.

5. **realm**—Enter the name of the realm with which this option tag will be associated. If you want to apply it globally across realms, leave this parameter blank.
6. **support-mode-inbound**—Optional. Indicate the support mode to define how the option tag is treated when encountered in an inbound SIP message's Supported header. The default value is **pass**. Valid values are:
  - **pass**—Indicates the B2BUA should include the tag in the corresponding outgoing message.
  - **strip**—Indicates the tag should not be included in the outgoing message. Use strip if you do not want the extension used.
7. **require-mode-inbound**—Optional. Indicate the require mode to define how the option tag is treated when it is encountered in an inbound SIP message's Require header. The default value is **reject**. The valid values are:
  - **pass**—Indicates the B2BUA should include the tag in the corresponding outgoing message.
  - **reject**—Indicates the B2BUA should reject the request with a 420 (Bad Extension) response. The option tag is included in an Unsupported header in the reject response.
8. **require-mode-outbound**—Optional. Indicate the require proxy mode to define how the option tag is treated when encountered in an incoming SIP message's Proxy-Require header. The default is reject. The valid values are:
  - **pass**—Indicates the B2BUA should include the tag in the corresponding outgoing message.
  - **reject**—Indicates the B2BUA should reject the request with a 420 (Bad Extension) response. The option tag is included in an Unsupported header in the reject response.

9. **support-mode-outbound**—Optional. Indicate the support mode to define how the option tag is treated when encountered in an outbound SIP message's Supported header. The default value is **pass**. Valid values are:
- **pass**—Indicates the B2BUA should include the tag.
  - **strip**—Indicates the tag should not be included in the outgoing message. Use **strip** if you do not want the extension used.
10. **require-mode-outbound**—Optional. Indicate the require mode to define how the option tag is treated when it is encountered in an outbound SIP message's Require header. The default value is **reject**. Valid values are:
- **pass**—Indicates the B2BUA should include the tag.
  - **reject**—Indicates the B2BUA should reject the request with a 420 (Bad Extension) response. The option tag is included in an Unsupported header in the reject response.
11. **require-mode-outbound**—Optional. Indicate the require proxy mode to define how the option tag is treated when encountered in an outgoing SIP message's Proxy-Require header. The default value is **reject**. The valid values are:
- **pass**—Indicates the B2BUA should include the tag.
  - **reject**—Indicates the B2BUA should reject the request with a 420 (Bad Extension) response. The option tag is included in an Unsupported header in the reject response.

The following example shows SIP option tag handling configured for non-default treatment of option tags.

```

sip-feature
  name                newfeature
  realm               peer-1
  support-mode-inbound Strip
  require-mode-inbound Reject
  proxy-require-mode-inbound Pass
  support-mode-outbound Pass
  require-mode-outbound Reject
  proxy-require-mode-outbound Reject
  last-modified-date  2004-12-08 03:55:05

```

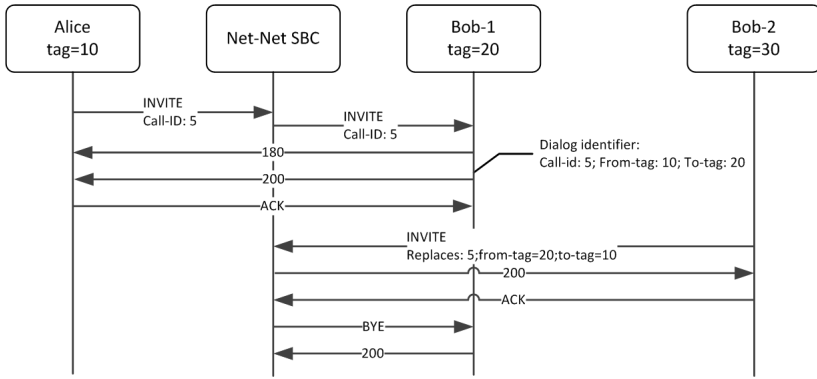
## Replaces Header Support

The Oracle Communications Session Border Controller supports the Replaces: header in SIP messages according to RFC 3891. The header, included within SIP INVITE messages, provides a mechanism to replace an existing early or established dialog with a different dialog which can be used for services such as call parking, attended call transfer and various conferencing features.

The Replaces: header indicates the dialog it wishes to replace by containing the corresponding dialog identifier. The identifier includes the triplet of the from tag, to tag, and call id. The orientation of endpoint-created tags, as from-tag and to-tag will match each of the two dialogs for a standard call. Thus the Replaces: header from an endpoint that indicates it wants to assume the dialog between the Oracle Communications Session Border Controller and Bob-1 appears as:

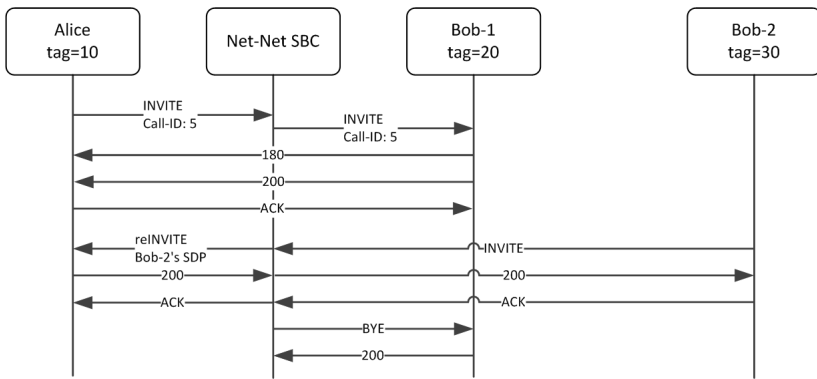
```
Replaces:5555;from-tag=20;to-tag=10
```

The Oracle Communications Session Border Controller validates that dialog identifier by matching an existing dialog and tries to install the new endpoint and remove the old endpoint by gracefully ending that dialog with a BYE. The replaces INVITE must come from an endpoint in the same realm as the endpoint it is replacing. If the UA sending the Replaces header is in a different realm as the original call leg (or indicates such architecture via a malformed Replaces: header), the Oracle Communications Session Border Controller replies to the Replaces: endpoint with a 481 Missing Dialog. Refer to the following diagram for the standard case.



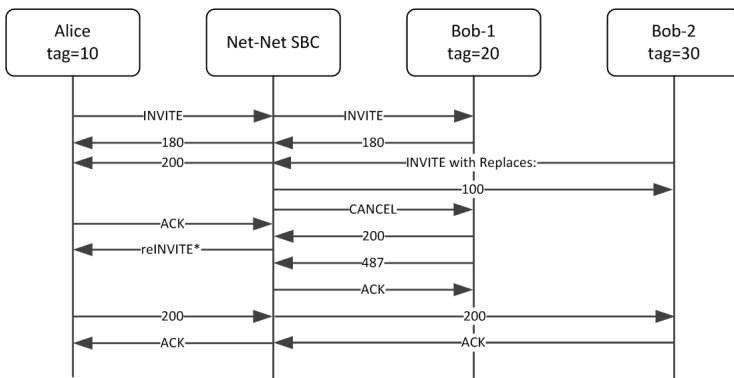
### New SDP Parameters in INVITE with Replaces

When an INVITE with Replaces: header is received, the media parameters in the new SDP are compared against the SDP of the dialog to be replaced. If any portion of the SDPs (excluding the session-origin line) is different, then the media must be renegotiated. The Oracle Communications Session Border Controller sends a re-INVITE with the new SDP to the dialog opposite of the one being replaced as shown below. If the re-INVITE fails for any reason, then the original dialogs will remain.



### Early Dialog Replacement

An INVITE with Replaces: header can replace an early dialog. That is, a dialog where the final 2xx class response to INVITE request has not arrived yet. The Oracle Communications Session Border Controller completes the originating side of the call with a 200 OK. The original dialog with the terminator is cancelled. SDP from the new terminator can be renegotiated if it changes.



The SDP from the original 183 response is used for the 200 response back to the originator if present to complete the early transaction. If reliable provisional messages are used, then no SDP is included in the 200 response.

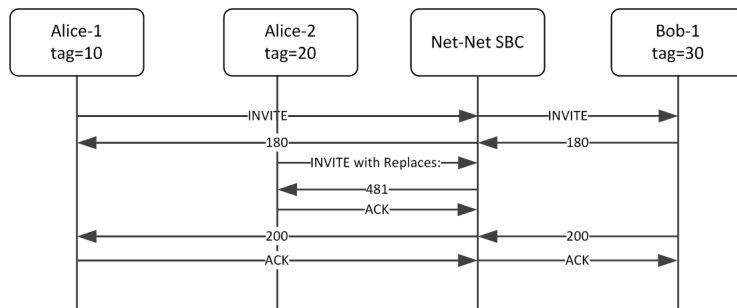
If no SDP is present in any of the provisional messages, then the Oracle Communications Session Border Controller constructs it from the original offer and modifying the IP port information for each c= and m= line with information

from the INVITE with Replaces: header. If there are more m= lines in the original offer than ports from the INVITE with Replaces: header, then the extra ports are disabled with port value of 0. If no SDP was offered in the original INVITE, then the SDP from the INVITE with Replaces: header is used as the offer in the 200 OK.

If the SDP media parameters were compatible between the replaced and replacing SDPs, then media does not need to be renegotiated and no re-INVITE is created. If the re-INVITE fails, the original dialogs are torn down using a BYE for the original server dialog.

## INVITE with Replaces in Early Dialog Server Side

The Oracle Communications Session Border Controller does not support replacing an early server dialog. It replies with a 481 (Dialog/Transaction does not exist) response to the endpoint requesting the replace.



## Replace Header Configuration

Replaces: header support is configured in the session-agent, realm, or sip-interface via the sip-profile configuration element. sip-profiles are defined once and attached to a chosen interface, realm or session-agent.

The replace-dialogs parameter is set to either **enabled** or **disabled**. In addition, you may set this parameter to **inherit** which uses the next lower order of precedence object. If there are no sip-profiles referenced in the higher ordered object, or if all the replace-dialogs parameters are set to **inherit**, then the feature is disabled.

To configure Replaces: header support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-profile** and press Enter. **If you are adding this feature to an existing configuration, then remember you must select the configuration you want to edit.**

```
ACMEPACKET(session-router)# sip-profile
ACMEPACKET(sip-profile)#
```

4. **replace-dialogs**—Set this parameter to **enabled** to enable Replaces: header support. A replaces parameter is also inserted in the Supported: header as sent into the realm where this sip profile is applied. You may also set this parameter to **inherit** for the element which this sip-profile is applied to, to inherit the value from the next-lower element.
5. Type **done** to save your work and continue.

## Debugging

### show sipd status

Includes Replaced Dialogs counts to show successfully replaced dialogs:

## SIP Signaling Services

```
# show sipd status
17:41:38-142
SIP Status          -- Period -- ----- Lifetime -----
Active             High   Total      Total  PerMax   High
Replaced Dialogs   -      -          1      1        1
```


### show sipd errors

Includes Replace Dialog Fails to show failed dialog replacements. This counter is incremented only when the dialog replacement attempt actually occurred but failed to successfully complete.

```
# show sipd errors
17:58:04-181
SIP Errors/Events   ----- Lifetime -----
Recent             Total  PerMax
Replace Dialog Fails  0      0      0
```

## SIP Options

This section explains how you can configure a limited list of specialized SIP features and/or parameters called options. The options described here were developed to meet specific needs not addressed by the standard SIP configuration parameters. Not all users have a need for these options.

 **Note:** Oracle recommends checking with your Oracle representative before applying any of these options.

### Overview

You can configure options for the SIP configuration and SIP interface. Both elements include a parameter (options) that you use to configure the options.

### Global SIP Options

The following table lists the SIP options supported by the Oracle Communications Session Border Controller (SBC).

Option	Description
add-error-to-tag=no	If present (even when set to no), suppresses the addition of an Acme tag on 3xx-6xx responses.
add-prov-to-tag=no	Prevents the SBC from adding a tag parameter to the To header (to-tag) to non-100 provisional responses to INVITE requests. Used when a provisional (101-199) response is received from the UAS on a client transaction without a to-tag. By default, the SBC adds the tag cookie in the response (as though it had a tag) sent back to the UAC for the associated server transaction. When you include this option in the SIP configuration, and the response from the UAS does not have a to-tag, the response forwarded to the UAC will not have a to-tag.
add-reg-expires	Causes an Expires header to always be included in a REGISTER response with the registration caching and HNT traversal functions of the SBC. Use for endpoints that do not understand the Expires parameter in the Contact header.
add-ruri-user=<methods>	Causes a userinfo portion to be added to a Request-URI when one is not present. Used to support the OKI phone, which registers a Contact of just an IP-Address but rejects initial INVITEs if the Request_URI does not have a userinfo part.



Option	Description
	<p>&lt;methods&gt; is a comma-separated list of methods to which the option should apply. If more than one method is listed, the list must be enclosed in quotes. This option only applies to out-of-dialog requests (no tag parameter in the To header). However, if ACK is listed, it will apply to all ACK requests because an ACK is always supposed to have a to-tag.</p>
allow-notify-no-contact	<p>Prevents the SBC from rejecting NOTIFYs with a 400 Bad Request response. NOTIFY requests without Contact header are allowed to pass through the SBC instead.</p>
call-id-host=<host>	<p>Causes the SBC to include a host part (ID@host) in the Call-ID it generated.</p> <p>&lt;host&gt; is the hostname (or IP address) that is to appear in the host part of the Call-ID. If not specified, the SIP port address is used.</p>
contact-endpoint=<param-name>	<p>Defines a URL parameter to report the real Contact address of an endpoint in a REGISTER message forwarded to a registrar, when the SBC is caching registration. (plain or HNT).</p> <p>If &lt;param-name&gt; is not specified, the default value endpoint is used. This parameter is added as a URL parameter in the Contact on the REGISTER message.</p> <p>In order for the registration cache to work properly, the softswitch/registrar is expected to include the endpoint parameter in the Request-URI of a SIP request it forwards to the address-of-record.</p>
contact-firewall=<param-name>	<p>Defines a URL parameter to report the NAT between the SBC and the real Contact address of an endpoint in a REGISTRAR message forwarded to a registrar when the SBC is doing registration caching for NHT.</p> <p>If &lt;param-name&gt; is not specified, the default value firewall is used.</p> <p>This parameter will be added as a URL parameter in the Contact on the REGISTER message.</p> <p>In order for the registration cache to work properly, the softswitch/registrar is expected to include the endpoint parameter in the Request-URI of any SIP request it forwards for the address-of-record.</p>
disable-privacy	<p>Prevents the change of the P-Preferred-Identity to P-Asserted-Identity and lets the P-Preferred-Identity go through unchanged.</p>
drain-sendonly	<p>Causes the SBC to examine the SDP attributes and change sendonly mode to sendrecv. This causes the endpoint receiving the SDP to send RTP, which is required for HNT traversal endpoints to work with media servers. The SBC sets up the flow so that RTP coming from the endpoint are dropped to prevent the UA that sent the sendonly SDP from receiving packets.</p> <p>See the option video-sbc-session also.</p>
encode-contact=<prefix>	<p>Causes the SBC to encode Contact addresses into the userinfo part of the URI. It applies only to Contact address that usually get the maddr parameter. Use when the SBC needs requests sent to the URI in the Contact sent instead to the SBC. The host part of the URI will have the SBC's address.</p>

## SIP Signaling Services

Option	Description
	<p>The &lt;prefix&gt; serves as a place between the original userinfo and the encoded address. If a &lt;prefix&gt; is specified, a default of +SD is used. Without this option, the SBC adds a maddr parameter.</p>
fix-to-header	<p>For requests that have the SBC address in both the Request-URI and the To-URI, it sets the hostport of the To-URI to a local policy's next hop target on out-of-dialog requests (no to-tag).</p> <p>This is the default IWF behavior, even without this option configured.</p>
forward-reg-callid-change	<p>Addresses the case when an endpoint reboots and performs a third party registration before its old registration expires. During this re-registration, the contact header is the same as it was pre-reregistration. As a consequence of the reboot, the SIP Call-ID changes.</p> <p>In this situation, the SBC does not forward the REGISTER to the registrar, because it believes the endpoint is already registered, based on a previous registration from the same Contact: header URI.</p> <p>To remedy this problem, the SBC now keeps track of the Call-ID in its registration cache. A new option in the SIP interface configuration element forces the SBC to forward a REGISTER message to the registrar when the Call-ID header changes in a REGISTER message received from a reregistering UAC.</p>
global-contact	<p>Addresses interoperability in the Dialog and Presence event packages that are used in hosted PBX and IP Centrex offerings. This option enables persistent URIs in the Contact headers inserted into outgoing SIP messages.</p> <p>If this option is not used, URIs placed in the Contact header of outgoing messages are only valid within the context of the dialog to which the message is associated.</p>
ignore-register-service-route-oos	<p>Prohibits a Register message from using a service route if that service route is an out-of-service session agent.</p>
load-limit=<cpu percentage>	<p>Defines the CPU usage percentage at which the SBC should start rejecting calls. Default value is 90%.</p>
lp-sa-match=<match strategy>	<p>Changes the ways local policies and session agents match; accounts for realm in matching process. Strategy choices are: all, realm, sub-realm, interface, and network.</p>
max-register-forward=<value>	<p>Defines a limit (as assigned in the value field) of REGISTERs to be forwarded to the registrar.</p> <p>During each second, the sipd counts how many REGISTERs have been sent to the registrar. It checks the threshold when it receives a REGISTER from the UA and determines that less than half the real registration lifetime is left. If the number of REGISTERs forwarded (new and updates) in the current second exceeds the configured threshold, it will respond to the UA from the cache.</p>
max-register-refresh=<value>	<p>Defines the desired limit of REGISTER refreshes from all the UAs. Each second of time, sipd counts the number of REGISTER/200-OK responses sent back. When the threshold is exceeded, it increments the expire time (based on NAT interval) by one second and resets the count.</p>

Option	Description
	By default no threshold is applied. The recommended value is somewhat dependent on the SBC hardware used, but 300 can be used as an initial value.
max-routes=<number of routes>	<p>Restricts the number of routes through which the sipd will iterate from a local policy lookup. For example, setting this option to 1 causes the SBC to only try the first, best, route. Setting this option to 0, or omitting it, lets the SBC use all of the routes available to it (with the priority scheme for route matching).</p> <p>When you test a policy using the test-policy CLI command, this option is not recognized and all options that match the criteria are displayed.</p>
max-udp-length=<maximum length>	<p>Setting this option to zero (0) forces sipd to send fragmented UDP packets. Using this option, you override the default value of the maximum UDP datagram size (1500 bytes; sipd requires the use of SIP/TCP at 1300 bytes).</p> <p>You can set the global SIP configuration's max-udp-length=x option for global use in your SIP configuration, or you can override it on a per-interface basis by configuring this option in a SIP interface configuration.</p>
media-release=<header-name>[;<header-param>]	<p>Enables the multi-system media release feature that encodes IP address and port information for the media streams described by SDP. It lets another SBC decode the data to restore the original SDP, which allows the media to flow directly between endpoints in the same network (that is serviced by multiple SBCs).</p> <p>The media release information can appear in the following places:</p> <ul style="list-style-type: none"> <li>• SIP header P-Media-Release: &lt;encoded-media-interface-information&gt;</li> <li>• Header parameter on a SIP header Contact: &lt;sip:1234@abc.com&gt; ; acme-media=&lt;encoded-media-interface-information&gt;</li> <li>• SDP attribute in the message body a=acme-media: &lt;encoded-media-interface-information&gt;</li> </ul> <p>Option includes the following:</p> <ul style="list-style-type: none"> <li>• &lt;header-name&gt; is SIP header in which to put the information or the special value sdp, which indicates the information should be put into the SDP.</li> <li>• &lt;header-param&gt; is the header parameter name in which to put the information or in the case of the special header name value sdp, it is the SDP attribute name in which to put the information.</li> </ul> <p>They identify to where the encoded information is passed. If you do not specify a header, P-Media-Release is used.</p>
no-contact-endpoint-port	<p>Enables the SBC to add a URL parameter (defined as an argument to the contact-endpoint option) to the Contact headers of REGISTER messages that it forwards to the registrar when it performs registration caching. The value of the contact-endpoint URL parameter is the real address of the endpoint; and if the endpoint is behind a NAT, this includes the IP</p>

Option	Description
	<p>address and a port number. However, not all network entities can parse that port number, which is included unconditionally. This feature allows you to configure the exclusion of the port number.</p> <p>Despite the fact that you set this parameter in the global SIP configuration, it is applied only to SIP interfaces. However, you can set a contact-endpoint option in the realm configuration, on which this new parameter has no effect.</p>
refer-to-uri-prefix=<prefix>	<p>Defines a prefix to be matched against the userinfo part of Contact headers (config=), of which the SBC should create a B2BUA map. This ensures that outgoing messages include the correct userinfo value. This option is used to enable add-on conferencing.</p>
reg-cache-mode=<mode>	<p>Affects how the userinfo part of Contact address is constructed with registration caching. &lt;mode&gt; values are:</p> <ul style="list-style-type: none"> <li>• none: userinfo from the received (post NAT) Contact is retained</li> <li>• from: userinfo from the From header is copied to the userinfo of the forwarded Contact header</li> <li>• append: append the UA's Contact address into a cookie appended to the userinfo from the original Contact userinfo. For HNT, the NAT/firewall address is used.</li> <li>• append-from: takes userinfo from the From header and appends the encrypted address to the userinfo from the original Contact userinfo. For HNT, the NAT/firewall address is used.</li> </ul> <p>The from mode is used with softswitches that do not use the cookies used by the SBC. It also helps limit the number of bytes in the userinfo; which might create duplicate contacts. For example, if the SBC address is 1.2.3.4, both 1234@5.6.7.8 and 1234@4.3.2.1 will result in a SBC contact of 1234@5.6.7.8.</p>
reg-contact-user-random	<p>Support the SIP random registered-contact feature. Gives the SBC the ability to support endpoints that randomly change their contact usernames every time they re-register. Only applicable to operators who need to support the Japan TTC standard JJ-90.22 in specific applications.</p> <p>Applies to cases when an endpoint re-registers with a different contact username, but with the same hostname/IP address and the same address of record (AoR). Without this feature enabled, the SBC forwards every re-registration to the registrar with the new contact information without it being considered a registration refresh. The SBC forwards it to the Registrar using the same sd-contact as the previous registration.</p> <p>When you set this option, the SBC does treat such a re-registration as a registration refresh when it is received prior to the half-life time for the specific contact. The SBC also uses the new contact username for the Request-URI in requests it sends to the UA, and verifies that the UA uses the correct one when that SBC is set to allow-anonymous registered mode.</p> <p>NOTE: The registration cache mode is set using the option reg-cache-mode, but regardless of how you configure it, the registration cache mode will be set to contact when SIP random registered-contact feature is enabled.</p>

Option	Description
register-grace-timer	Makes the grace time for the SIP Registration configurable. You can configure the grace timer in seconds.
reinvite-trying=[yes]	Causes the SBC to send a 100 Trying for re-INVITEs, which is normally suppressed. If you enter the option name but omit the value yes, the option is still active.
reject-interval=<value>	<p>Acts as a multiplier to increase the value presented to the UAC in the Retry-After field. For example, if reject-interval=5 (reject interval is set to 10); at a 90% rejection rate the SBC sends Retry-After: 45.</p> <p>When rejecting calls because of CPU load limiting, the SBC adds a Retry-After parameter to the error response (typically 503 Service Unavailable). By default the SBC sets the Retry-After value to be 1/10th of the current rejection rate.</p>
reject-register=[no   refresh]	Allows REGISTER messages through even during load limiting. By default, REGISTER messages are subject to load limiting.
response-for-not-found=<response code>	Change the 404 Not Found generated by the SBC to a different response code.
route-register-no-service-route	<p>Controls how a UA is registered. Option can have three values:</p> <ul style="list-style-type: none"> <li>• route-register-no-service-route—This option prevents the use of the Service-Route procedure to route the Re-Register requests after the UA has initially registered.</li> <li>• route-register-no-service-route=all—Prevents the use of the Service-Route procedure to route the Re-Register requests for all messages, after the UA has initially registered.</li> <li>• route-register-no-service-route=refresh—Prevents the use of the Service-Route procedure to route the Re-Register requests for all refresh-register messages, but not de-register messages, after the UA has initially registered.</li> </ul> <p>Addition idle argument ensures that, when enabled, the SBC follows the previously defined rules for idle calls, where idle means not engaged in any INVITE-based sessions.</p> <p>Sample syntax: route-register-no-service-route=refresh;idle</p>
sdp-insert-sendrecv	When a call is initiated, the SDP communicates between call offerer and call answerer to determine a route for the media. Devices can be configured to only send media (“a=sendonly”), to only receive media (“a=recvonly”), or to do both (“a=sendrecv”). Some devices, do not disclose this information. With this option configured, when either the offerer or answerer does not disclose its directional attribute, the SBC automatically inserts a sendrecv direction attribute to the media session.
set-inv-exp-at-100-resp	Set Timer C when a 100 Trying response is received (instead of waiting until 1xx (> 100) is received). If the SBC does not receive a 100 Trying response within Timer B, the call should be dropped because there is a problem communicating with the next hop.
strip-domain-suffix-route	Causes sipd to strip any Router headers from the inbound messages coming to the external address of a SIP NAT; if the message contains a FQDN that matches the configured domain suffix for that SIP NAT.

## SIP Signaling Services

Option	Description
video-sbc-session	Use with drain-sendonly for conference floor support. When configured with drain-sendonly and when the SBC receives an SDP, the SBC proxies the m=control and its related a= and c= unchanged. Although media streams are allocated for this m line, an actual flow is not set up.  SDP received with the following:  m=video  a=sendonly  is sent out as the following:  m=video  a=sendonly  a=X-SBC-Session
session-timer-support	This option enables the SBC to start the session timer for session refreshes coming from the UAC. The SBC determines whether or not a session is active based on session refreshes or responses. It terminates the session when no session refreshes occur within the session timer interval.
session-timer-support	Enables RFC4028 session timer support.
inmanip-before-validate	Enables SIP Header Pre-processing for HMR.
proccess-implicit-tel-URI	Correctly appends coodie in REGISTER message when user=phone does not exist.
offerless-bw-media	Reserves appropriate bandwidth for an INVITE with no SDP.

## SIP Interface Options

The following table lists the SIP interface options supported by the Oracle Communications Session Border Controller.

Option	Description
100rel-interworking	Enables RFC 3262, <i>Reliability of Provisional Responses in the Session Initiation Protocol</i> support.
contact-endpoint=<endpoint name>	The Oracle Communications Session Border Controller inserts the endpoint IP address and port into the Contact headers as messages egress using that SIP interface. The inserted data is the same as the information received in the Request or Response being forwarded.  If the endpoint name is not specified, the default value endpoint is used.
contact-firewall=<firewall name>	The Oracle Communications Session Border Controller inserts the firewall IP address and port into the Contact headers as messages egress using that SIP interface. The inserted data is the same as the information received in the Request or Response being forwarded.  If the endpoint name is not specified, the default value firewall is used.
contact-vlan=<VLAN/realm name>	The Oracle Communications Session Border Controller inserts the realm and VLAN ID into the Contact headers as messages egress using that

Option	Description
	<p>SIP interface. The inserted data is the same as the information received in the Request or Response being forwarded.</p> <p>If the endpoint name is not specified, the default value vln is used.</p>
dropResponse	<p>The Oracle Communications Session Border Controller drops responses by specified status codes. The option value can contain one or more status codes separated by semicolons. Error ranges can also be entered. If any of the response codes matches then a response is not sent. If the dropResponse option is set in both the sip-interface and the session-agent elements, the session-agent setting takes precedence.</p>
max-udp-length=<maximum length>	<p>Sets the largest UDP packers that the Oracle Communications Session Border Controller will pass. Packets exceeding this length trigger the establishment of an outgoing TCP session to deliver the packet; this margin is defined in RFC 3261. The system default for the maximum UDP packet length is 1500.</p> <p>You can set the global SIP configuration's max-udp-length=x option for global use in your SIP configuration, or you can override it on a per-interface basis by configuring this option in a SIP interface configuration.</p>
response-for-not-found=<response code>	<p>Change the 404 Not Found generated by the SBC to a different response code.</p>
strip-route-headers	<p>Causes the Oracle Communications Session Border Controller to disregard and strip all route headers for requests received on a SIP interface.</p>
upd-fallback	<p>When a request needs to be sent out on the SIP interface for which you have configured this option, the Oracle Communications Session Border Controller first tries to send it over TCP. If the SIP endpoint does not support TCP, however, then the Oracle Communications Session Border Controller falls back to UDP and tries the request again.</p>
via-header-transparency	<p>Enables the Oracle Communications Session Border Controller to insert its Via header on top of the top-most Via header received from user equipment (UE). It then forwards it on to the IP Multimedia Subsystem (IMS) core with the original Via header now located as the bottom-most Via header.</p> <p>The Oracle Communications Session Border Controller still replaces the Contact and other header addresses with its own, and does not pass on the core's Via headers in outbound requests.</p>
use-redirect-route	<p>Use Route parameter in Contact header as next-hop as received in a 3xx response.</p>
reg-via-proxy	<p>Enables your Oracle Communications Session Border Controller to support endpoints that register using an intervening proxy.</p>
lmsd-interworking	<p>Enables 3GPP2 LMSD Interworking.</p>
suppress-reinvite	<p>Enables reINVITE suppression.</p>



## SIP Session Agent Options

The following table lists the SIP session agent options supported by the Oracle Communications Session Border Controller.

Option	Description
dropResponse	The Oracle Communications Session Border Controller drops responses by specified status codes. The option value can contain one or more status codes separated by semicolons. Error ranges can also be entered. If any of the response codes matches then a response is not sent. If the dropResponse option is set in both the sip-interface and the session-agent elements, the session-agent setting takes precedence.
trans-timeouts=<value>	Defines the number of consecutive non-ping transaction timeouts that will cause a session agent to be out of service. When the session agent is configured, i.e. when the PING options are defined, the value is 10. If not defined, the default value is 5. A Value of 0 prevents the session agent from going out of service because of a non-ping transaction timeout.
via-origin=<parameter-name>	Causes a parameter to be included in the top Via header of requests sent to the session agent. The parameter indicates the source IP address of the corresponding request received by the Oracle Communications Session Border Controller.  <parameter-name> defines the name of the parameter. If not specified, the default value origin is used.
refer-reinvite	Enables SIP REFER with Replaces.

## SIP Realm Options

The following table lists the SIP session agent options supported by the Oracle Communications Session Border Controller.

Option	Description
number-normalization	Applies to the SIP To URI. (Currently the Oracle Communications Session Border Controller supports number normalization on From and To addresses for both inbound and outbound call legs.) Number normalization includes add, delete, and replace string functions that result in consistent number formats.  Number normalization occurs on ingress traffic, prior to the generation of accounting records or local policy lookups. (also applies for H.323 to SIP calls.)
refer-reinvite	Enables SIP REFER with Replaces.

## SIP Realm Options Configuration

To configure options:

Labels enclosed in <> indicate that a value for the option is to be substituted for the label. For example, <value>. In order to change a portion of an options field entry, you must re-type the entire field entry.

1. Navigate to the options parameter in the SIP configuration or SIP interface elements.
2. Enter the following:

```
options Space <option name>="<value>"
```



For example, if you want to configure the refer-to-uri-prefix option (the add-on conferencing feature):

Type options, followed by a Space.

Type refer-to-uri-prefix, followed by an equal sign (=).

Type the opening quotation mark (") followed by conf, another equal sign and the closing quotation mark.

Press Enter.

For example:

```
options refer-to-uri-prefix=conf=
```

If the feature value itself is a comma-separated list, it must be enclosed in quotation marks.

## Configuring Multiple Options

You can enter a list of options for this field:

1. Type **options** followed by a space.
2. Within quotation marks, enter the feature names and values of the parameters you need. Separate each one with a comma.
3. Close the quotation marks.
4. Press Enter.

For example:

```
ACMEPACKET(sip-config)# options "refer-to-uri-prefix="conf=", encode-  
contact="+SD", add-ruri-user=INVITE,ACK"
```

## Adding an Entry

Enter the new entry with a preceding plus (+) sign. For example:

```
options +response-for-not-found
```

This format allows previously configured options field values to remain intact without requiring re-entry of the entire field value.

## SIP Security

This section provides an overview of Oracle Communications Session Border Controller's security capability. Oracle Communications Session Border Controller security is designed to provide security for VoIP and other multi-media services. It includes access control, DoS attack, and overload protection, which help secure service and protect the network infrastructure (including the Oracle Communications Session Border Controller). In addition, Oracle Communications Session Border Controller security lets legitimate users to still place calls during attack conditions, protecting the service itself.

Oracle Communications Session Border Controller security includes the Net-SAFE framework's numerous features and architecture designs. Net-SAFE is a requirements framework for the components required to provide protection for the Session Border Controller (SBC), the service provider's infrastructure equipment (proxies, gateways, call agents, application servers, and so on), and the service itself.

## Denial of Service Protection

The Oracle Communications Session Border Controller Denial of Service (DoS) protection functionality protects softswitches and gateways with overload protection, dynamic and static access control, and trusted device classification and separation at Layers 3-5. The Oracle Communications Session Border Controller itself is protected from signaling and media overload, but more importantly the feature allows legitimate, trusted devices to continue receiving service even during an attack. DoS protection prevents the Oracle Communications Session Border Controller host processor from being overwhelmed by a targeted DoS attack from the following:

## SIP Signaling Services

---

- IP packets from an untrusted source as defined by provisioned or dynamic ACLs
- IP packets for unsupported or disabled protocols
- Nonconforming/malformed (garbage) packets to signaling ports
- Volume-based attack (flood) of valid or invalid call requests, signaling messages, and so on.
- Overload of valid or invalid call requests from legitimate, trusted sources

### Levels of DoS Protection

The multi-level Oracle Communications Session Border Controller Denial of Service protection consists of the following strategies:

- Fast path filtering/access control: involves access control for signaling packets destined for the Oracle Communications Session Border Controller host processor as well as media (RTP) packets. The SBC accomplishes media filtering using the existing dynamic pinhole firewall capabilities. Fast path filtering packets destined for the host processor require the configuration and management of a trusted list and a deny list for each Oracle Communications Session Border Controller realm (although the actual devices can be dynamically trusted or denied by the Oracle Communications Session Border Controller based on configuration). You do not have to provision every endpoint/device on the Oracle Communications Session Border Controller, but instead retain the default values.
- Host path protection: includes flow classification, host path policing and unique signaling flow policing. Fast path filtering alone cannot protect the Oracle Communications Session Border Controller host processor from being overwhelmed by a malicious attack from a trusted source. The host path and individual signaling flows must be policed to ensure that a volume-based attack will not overwhelm the Oracle Communications Session Border Controller's normal call processing; and subsequently not overwhelm systems beyond it. The Oracle Communications Session Border Controller must classify each source based on its ability to pass certain criteria that is signaling- and application-dependent. At first each source is considered untrusted with the possibility of being promoted to fully trusted. The Oracle Communications Session Border Controller maintains two host paths, one for each class of traffic (trusted and untrusted), with different policing characteristics to ensure that fully trusted traffic always gets precedence.
- Host-based malicious source detection and isolation – dynamic deny list. Malicious sources can be automatically detected in real-time and denied in the fast path to block them from reaching the host processor.

## Configuration Overview

NAT table entries are used to filter out undesired IP addresses (deny list). After the packet from an endpoint is accepted through NAT filtering, policing is implemented in the Traffic Manager based on the sender's IP address. NAT table entries are used to distinguish signaling packets coming in from different sources for policing purposes.

You can configure deny rules based on the following:

- ingress realm
- source IP address
- transport protocol (TCP/UDP)
- application protocol (SIP)

You can configure guaranteed minimum bandwidth for trusted and untrusted signaling paths.

You can configure signaling path policing parameters for individual source addresses. Policing parameters include:

- peak data rate in bits per second
- average data rate in bits per second
- maximum burst size

## SIP Unauthorized Endpoint Call Routing

The Oracle Communications Session Border Controller (SBC) can route new dialog-creating SIP INVITEs from unauthorized endpoints to a session agent or session agent group; then rejection can occur based on the allow-anonymous setting for the SIP port. This type of provisional acceptance and subsequent rejection applies only to INVITEs; the SBC continues to reject all other requests, such as SUBSCRIBE.

You might enable this feature if you have a network in which unauthorized SIP endpoints continually try to register even if the Oracle Communications Session Border Controller has previously rejected them and never will accept them. For instance, the user account associated with the endpoint might have been removed or core registrars might be overloaded.

## SIP Unauthorized Endpoint Call Routing Configuration

You enable the routing of unauthorized endpoints to session agents and session agent groups that will reject them in the SIP interface configuration.

To enable SIP unauthorized endpoint call routing:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-interface) #
```

If you are adding this feature to an existing configuration, then you will need to select the configuration you want to edit.

4. **route-unauthorized-calls**—Enter the name (or IP address) of the session agent or session agent group to which you want calls from unauthorized endpoints routed. This parameter is blank by default, meaning the SIP unauthorized call routing feature is disabled.

Remember your settings in the **allow-anonymous** parameter in the SIP port configuration provide the basis for rejection.

5. Save and activate your configuration.

## SIP NAT Function

This section explains how to configure the optional SIP NAT function. You can configure the SIP NAT function if you need to translate IP address and UDP/TCP port information. The SIP NAT function also prevents private IP addresses in SIP message URIs from traveling through an untrusted network.

### Overview


The Oracle Communications Session Border Controller is an intermediary device that provides NAT functions between two or more realms. It translates IP addresses between untrusted and trusted networks using NAT. A trusted network is inside the NAT, and an untrusted network is outside the NAT. A NAT also lets a single IP address represent a group of computers.

For SIP, the SIP NAT function on the Oracle Communications Session Border Controller does the following:

- routes SIP packets between the Oracle Communications Session Border Controller's SIP proxy (B2BUA) and external networks (or realms), including the translation of IP address and UDP/TCP port information.
- prevents private IP addresses in SIP message URIs from traveling through the untrusted network. SIP NAT either translates the private address to one appropriate for an untrusted address or encrypts the private address into the URI.

Packets arriving on the external address (at port 5060) are forwarded to the Oracle Communications Session Border Controller's SIP proxy with the source address changed to the home address (at port 5060). When the Oracle Communications Session Border Controller's SIP proxy sends packets to the home address (at port 5060), they are

forwarded to the external proxy address (and external proxy port), with the source address changed to the external address (at port 5060).

-  **Note:** The SIP config's NAT mode parameter works in conjunction with the SIP NAT function configuration. It identifies the type of realm in which the SIP proxy is located (public or private) and affects whether IPv6 addresses in SIP messages are encoded.

The translation of URIs in the actual SIP message occurs as messages are received and sent from the Oracle Communications Session Border Controller's SIP proxy. For the messages being sent to the external network, the contents of the SIP message are examined after the translation to determine if the destination needs to be changed from the external proxy address to an address and port indicated by the SIP message. This process takes place so the request is sent to where the Request-URI or the Route header indicates, or so the response is sent to where the Via indicates.

### NAT Modes


The specific addresses used in translating URIs in the SIP message depend on whether the Oracle Communications Session Border Controller is performing NAT functions for a trusted or untrusted network. This condition is determined by the NAT mode value you enter when you configure the SIP config element. The NAT modes are:

- **untrusted**—The SIP proxy is associated with an address for an untrusted network (the address value you entered when you configured the SIP interface's SIP port parameter), and the home address in the SIP NAT is the address of the external realm/network. When the URI contains the external address, it is translated to the SIP NAT's home proxy address (or to the SIP port address if the home proxy address field is empty). When a URI contains the external proxy address, it is translated to the home address.

If the URI contains any other private address (matching the realm's address prefix, identified in the SIP NAT's realm ID), it is encrypted and the address is replaced with the home address value. If the URI contains a user part, a suffix consisting of the user NAT tag and the encrypted address is appended to the user part. For example, with a user NAT tag value of `-private-`, the private URI of `sip@123192.169.200.17:5060` will become the public URI of `sip:123-private-eolmhet2chbl3@172.16.0.15`.


If there is no user part, the host consists of the host NAT tag followed by the encrypted address and the domain suffix. A `maddr` parameter equal to the home address (or received in the case of a Via header) is added to the URI. For example, with a host NAT tag value of `PRIVATE-` and a domain suffix value of `private.com`, the private URI of `sip:192.168.200.17:5060` will become the public URI of `sip:PRIVATE-eolmhet2chbl3.private.com:5060;maddr=172.16.0.15`.

- **trusted**—The SIP proxy is on a trusted network (the address value you entered when you configured the SIP interface's SIP port parameter), and the SIP NAT's external address is the public address of the external realm/network. When the URI contains the home address value, it is translated to the value set for the external proxy address. When the URI contains the SIP proxy's address, it is translated to the external address. If the URI contains any other private address (matching the realm's address prefix, identified in the SIP NAT's realm ID), the private address is encrypted and the address is replaced with the external address.

-  **Note:** Do not use the home proxy address value with private NAT functioning.

### Adding a maddr Parameter to a URI

When you configure a SIP interface, you can configure the contact mode. The contact mode sets the contact header routing mode, which determines how the contact address from a trusted network is formatted. You set the contact mode to add a `maddr` parameter equal to the SIP proxy to the URI in the Contact header. For example, the URI from the prior example (`sip:192.168.200.17:5060`) becomes `sip:123-trusted-eolmhet2chbl3@172.16.0.15;maddr=172.16.0.12`.

-  **Note:** For SIP elements that do not support the `maddr` parameter, configure a Contact mode as `none`.

You might require this encryption to cause other SIP elements in the untrusted network to send requests directly to the SIP proxy. Otherwise, the requests are sent to the home address. However, responses sent by the SIP proxy will have

the SIP proxy's source address, rather than the home address. Some SIP elements might drop responses that come from a IP address different from the one to which the request is sent.

## About Headers

You can specify which SIP headers you want effected by the SIP NAT function. The URIs in these headers are translated and encrypted, the encryption occurs according to the rules of this SIP NAT function.

You can enter header values by using either the full header name or its corresponding abbreviation, if applicable. The following table lists the available headers and their corresponding abbreviations

Header	Abbreviation
Call-ID	i
Contact	m
From	f
Record-Route	none
Route	none
Ready-To	none
Replaces	none
Refer-To	r
To	t
Via	v

SIP sessions are terminated and re-originated as new sessions as they are routed through the Oracle Communications Session Border Controller. Among the actions performed, SIP headers are modified to prevent the transmission of IP address and route information.

## Replacing Headers

In the SIP signaling message, any Via headers are stripped out and a new one is constructed with the Oracle Communications Session Border Controller's IP address in the sent-by portion. If a Contact header is present, it is replaced with one that has the Oracle Communications Session Border Controller's IP address. All other headers are subject to NATing based on the following rules:

- The Request-URI is replaced with the next hop's IP or FQDN address.
- All other headers are replaced based on the two SIP NAT function SIP NAT function rules

## Mapping FQDNs

The Oracle Communications Session Border Controller maps FQDNs that appear in the certain headers of incoming SIP messages to the IP address that the 12:00 inserts in outgoing SIP contact headers. The mapped FQDNs are restored in the SIP headers in messages that are sent back to the originator.

This feature is useful to carriers that use IP addresses in the SIP From address to create trunk groups in a softswitch for routing purposes. When the carrier's peer uses FQDNs, the carrier is forced to create trunk groups for each possible FQDN that it might receive from a given peer. Similarly, this can apply to SIP Contact and P-Asserted-Identity headers.

## SIP NAT Function Cookies

Cookies are inserted to hide that information is coming from a realm external to the home realm. They are used when information needs to be placed into a given element of a SIP message that must also be seen in subsequent SIP messages within a flow. When forwarding a SIP message, the Oracle Communications Session Border Controller

(SBC) encodes various information in the outgoing message, which is passed from one side to another in SIP transactions.

SIP NAT function cookies let the SBC hide headers, IPv4 addresses, and SIP URIs. These cookies are included when certain conditions are present in SBC SIP transactions.

Oracle's SIP NAT function cookies can be used in the userinfo, host, URL parameter, and tel URL parameter portions of the SIP message.

### userinfo

The Oracle Communications Session Border Controller places a cookie in the userinfo portion of a SIP URI when a SIP header contains a SIP URI, and includes that header type in the list of headers to be hidden (encrypted) in the associated SIP NAT function. The cookie for the userinfo portion is the following:

```
[user nat tag][encrypted 13-byte host IP][encrypted 13 byte maddr IP (if present)]
```

where:

- [user nat tag] refers to the SIP NAT function's original user NAT tag field.
- [encrypted 13-byte host IP] refers to the host IP encryption.
- [encrypted 13 byte maddr IP (if present)] refers to the maddr IP encryption, if it exists.

With a user NAT tag of -acme, the following SIP-URI:

```
sip:6175551212@192.168.1.100
```

might be translated into:

```
sip:6175551212-acme-pfils7n2pstna@172.16.1.10
```



**Note:** Multiple additional cookies might be appended with each hop (for example, from the external proxy to the home proxy and back).

### host

When hiding IP addresses in a SIP message, the SIP NAT function generates the following cookie for a SIP-URI with no userinfo portion:

```
[host nat tag][encrypted 13-byte host IP][encrypted 13 byte maddr IP (if present)][domain suffix]
```

where:

- [host nat tag] refers to the SIP NAT function's host NAT tag.
- [encrypted 13-byte host IP] refers to the host IP encryption.
- [encrypted 13 byte maddr IP (if present)] refers to the maddr IP encryption, if it exists.
- [domain suffix] refers to the SIP NAT function's domain suffix field.

With a SIP NAT function's host tag of ACME- and a domain suffix of .acme.com, the following SIP header:

```
Via: SIP/2.0/UDP 192.168.1.100:5060
```

might be translated into the following:

```
Via: SIP/2.0/UDP ACME-pfils7n2pstna.acme.com
```

### URL Parameter

If the SIP NAT function's use url parameter field has a value of from-to or all, the SIP NAT function places all cookies generated to hide SIP URIs in a custom tag appended to the header. Setting the use url parameter field to:

- from-to only affects the behavior of the SIP NAT function's cookies in the From and To headers.
- all affects all SIP headers processed by the SIP NAT function

The cookie is the following:

[;url-parameter]=[host nat tag][encrypted 13-byte host IP][encrypted 13-byte maddr IP]

where:

- [;url-parameter] refers to the SIP NAT function's parameter name field.  
This cookie type is associated with the all and from-to field value options of the SIP NAT function's use url parameter field.
- [host nat tag] refers to the SIP NAT function's host NAT tag field.
- [encrypted 13-byte host IP] refers to the host IP encryption.
- [encrypted 13 byte maddr IP (if present)] refers to the maddr IP encryption, if it exists.

With a host NAT tag of ACME- and a parameter name of acme\_param, the following SIP-URI:

```
sip:6175551212@192.168.1.100
```

might be translated into the following:

```
sip:6175551212@172.16.1.10;acme_param=ACME-pfils7n2pstna.
```

### tel URL

The SIP NAT function cookie is used when devices in your network are strict about the context portion of SIP messages regarding the conversion of tel URLs. This cookie for the tel URL parameter portion of a SIP message is the following:

```
tel URL parameter-[13-byte host IP][13 byte optional maddr IP]domain suffix
```

where:

- tel URL parameter refers to the SIP NAT function's use url parameter.  
This cookie type is associated with the use url parameter's phone field value for the SIP NAT.
- [13-byte host IP] refers to the host IP encryption.
- [13 byte optional maddr IP] refers to the maddr IP encryption, if it exists.
- domain suffix refers to the SIP NAT function's domain suffix field.

## Configuration Overview

Configuring the SIP NAT function falls into two areas, the SIP NAT interface parameters and the SIP NAT policies.

### SIP NAT Interface

The following tables lists the SIP NAT function interface parameters you need to configure on the Oracle Communications Session Border Controller (SBC).

Parameter	Description
realm ID	Name of the external realm. The realm ID must be unique; no two SIP NATs can have the same realm ID.  This realm ID must also correspond to a valid realm identifier entered when you configured the realm.
external proxy address	IPv4 address of the SIP element (for example, a SIP proxy) in the external network with which the SBC communicates. Entries must follow the IP address format.
external proxy port	UDP/TCP port of the SIP element (for example, a SIP proxy) in the external network with which the SBC communicates.  Minimum value is 1025, and maximum value is 65535. Default is 5060.
external address	IPv4 address on the media interface in the external realm. Enter a value that ensures any packet with an external address value as its destination

## SIP Signaling Services

Parameter	Description
	<p>address is routed to the SBC through the media interface connected to or routable from the external realm. Entries must follow the IP address format.</p> <p>To specify whether the external realm referenced in this field is private or public, configure the SIP config's NAT mode.</p>
home address	<p>IPv4 address on the media interface in the home realm. Enter a value that ensures any packet with a home address value as its destination address must be routed to the SBC through the media interface connected to or routable from the home realm. Entries must follow the IP address format.</p> <p>The value entered in this field must be different from the IP address value of the home realm's network interface element.</p> <p>The home realm network interface is associated with this SIP NAT by its realm ID and the realm's identifier and network interface value you entered when you configured the realm. The realm's network interface identifier value corresponds to this SIP NAT's realm ID, the SIP config's home realm ID, and the media manager's home realm ID.</p>
home proxy address	<p>Sets the IP address for the home proxy (from the perspective of the external realm).</p> <p>By default, this field is empty.</p> <p>An empty home proxy address field value signifies that there is no home proxy, and the external address will translate to the address of the SBC's SIP proxy. Entries must follow the IP address format.</p>
home proxy port	<p>Sets the port number for the home realm proxy.</p> <p>Value can be set to zero (0). Minimum is 1025 and maximum is 65535. Default is 5060.</p>
route home proxy	<p>Whether to route all inbound requests for the SIP NAT to the home proxy.</p> <p>enabled adds route if Request-URI is not the SBC</p> <p>disabled does not route inbound requests to the home proxy</p> <p>forced always adds route</p>

### SIP NAT Function Policies

The following tables lists the SIP NAT function policy parameters you need to configure on the Oracle Communications Session Border Controller (SBC).

Parameter	Description
domain suffix	<p>Domain name suffix of the external realm. The domain name suffix refers to and must conform to the hostname part of a URI. In combination with the user NAT tag and host NAT tag values, this value is used to help the SBC identify an encoded URI that it needs to translate when moving between public and private realms.</p> <p>This suffix is appended to encoded hostnames that the SIP NAT function creates. For example, if the encoded hostname is ACME-abc123 and the domain-suffix value is .netnetsystem.com, the resulting FQDN will be ACME-abc123.netnetsystem.com.</p>



Parameter	Description
address prefix	Defines which IPv4 address prefixes from incoming messages require SIP-NAT encoding (regardless of the realm from which these messages came).
tunnel redirect	Controls whether Contact headers in a 3xx Response message received by the SBC are NATed when sent to the initiator of the SIP INVITE message.
use url parameter	Establishes whether SIP headers will use the URL parameter entered in the parameter name for encoded addresses that the SIP NAT function creates. Also, if SIP headers will be used, which type of headers will use the URL parameter. For example, all headers or just the From and To headers. Enumeration field.
parameter name	Indicates the name of the URL parameter when use url applies. This field value will be used in SIP NAT encoding addresses that have a use url parameter value of either from-to or all.
user NAT tag	Identifies the prefix used when an address is encoded into the username portion of user@host;name=xxxx; where name = parameter name.  The user NAT tag values can consist of any characters that are valid for the userinfo part of a URI. In combination with the domain suffix and host NAT tag field values, this value is used to help the SBC identify an encoded URI that it needs to translate when moving between public and private realms.
host NAT tag	Identifies the prefix used when encoding an address into the hostname part of the URI or into a URL parameter. The host NAT tag values refer to domain labels and can consist of any characters that are valid for the hostname part of a URI. In combination with the domain suffix and user NAT tag values, this value is used to help the SBC identify an encoded URI that it needs to translate when moving between public and private realms.
headers	Lists the SIP headers to be affected by the SBC SIP NAT function. The URIs in these headers will be translated and encrypted, and encryption will occur according to the rules of this SIP NAT.

## SIP NAT Function Configuration

To configure the SIP NAT function on an Oracle Communications Session Border Controller (SBC):

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-nat** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-nat
ACMEPACKET(sip-nat)#
```

4. **realm-ID**—Enter the name of the realm you want to identify as the external realm.

The name you use as the realm ID must be unique. No two SIP NAT functions can have the same realm ID. Also, this value must correspond to a valid identifier entry already configured for the realm.

5. **domain-suffix**—Enter the domain suffix to identify the domain name suffix of the external realm. The domain suffix must begin with a (.) dot.

The domain name suffix refers to and must conform to the hostname part of a URI. For example:

```
.netnatsystem.com
```

The domain suffix is appended to encoded hostnames that the SIP NAT function creates. For example, if the encoded hostname is ACME-abc123, the resulting FQDN is ACME-abc123.netnetsystem.com.

6. **external-proxy-address**—Enter the external proxy address to identify the IPv4 address of the SIP element (for example, a SIP proxy) in the external network with which the SBC communicates.

Enter the value in the IP address format. For example:

```
192.168.11.200
```

7. **external-proxy-port**—Enter the external proxy port value to identify the UDP/TCP port of the SIP element (for example, a SIP proxy) in the external network with which the SBC communicates. The default is **5060**. The valid range is:

- Minimum—1025
- Maximum—65535

8. **external-address**—Enter the external address, which is an IPv4 address on the media interface in the external realm.

Enter the value in the IP address format. For example:

```
192.168.11.101
```

This value must be such that any packet with an external address value as its destination address is routed to the SBC through the media interface connected to or routable from the external realm.

9. **home-address**—Enter the home address, which is an IPv4 address on the network interface in the home realm. This value must be such that any packet with a home address value as its destination address must be routed to the SBC through the media interface connected to or routable from the home realm.

Enter the value in the IP address format. For example:

```
127.0.0.10
```

The value entered in this field **must be different** from the IP address value of the home realm's network interface element.

The home realm network interface is associated with this SIP NAT by its realm ID and the realm's identifier and network interface value you entered when you configured the realm. The realm's network interface identifier value corresponds to this SIP NAT's realm ID, the SIP config's home realm ID, and the media manager's home realm ID.

10. **home-proxy-address**—Enter the home proxy address to set the IP address for the home proxy (from the perspective of the external realm).

By default, this field is empty. No home proxy address entry signifies there is no home proxy, and the external address will translate to the address of the SBC's SIP proxy.

Enter the value in the IP address format. For example:

```
127.1.0.10
```


11. **home-proxy-port**—Enter the home proxy port to set the port number for the home realm proxy. The default value is **0**. The valid range is:

- Minimum—0, 1025
- Maximum—65535

12. **route-home-proxy**—Optional. Enable or disable requests being routed from a given SIP-NAT to the home proxy. The default value is **disabled**. The valid values are:

- **enabled**—All inbound requests for a specific SIP NAT are routed to the home proxy
- **disabled**—All inbound requests are not routed through the home proxy.
- **forced**—The Request is forwarded to the home proxy without using a local policy.

13. **address-prefix**—Optional. Indicate the IPv4 address prefix from incoming messages that requires SIP NAT function encoding (regardless of the realm from which these messages came).

 **Note:** This value overrides the value set in the realm's address prefix field.

This field's format incorporates an IPv4 address and number of bits in the network portion of the address. For example, a Class C address has a 24-bit network part. The address prefix for 101.102.103.x would be represented as 10.102.103.0/24.

The default value is an asterisk (\*). When you enter this value or do not enter a value, the realm's address prefix value is used.

**14. tunnel-redirect**—Set to one of the following values to indicate whether certain headers in a 3xx Response message received by the SBC are NATed when sent to the initiator of the SIP INVITE message. The default is **disabled**. The valid values are:

- **enabled**—Certain headers in a 3xx Response message are NATed.
- **disabled**—Certain headers in a 3xx Response message are not NATed.

**15. use-url-parameter**—Establish whether SIP headers will use the URL parameter (configured in the next step) for encoded addresses created by the SIP NAT function. If SIP headers will be used, this value identifies which types of headers will use the URL parameter. The default value is **none**. The available values include:

- **none**—No headers will use the URL parameter for address encoding.

The following example illustrates the functionality of an SBC using a use url parameter value of none:

```
sip: 1234@1.2.3.4 is translated into sip: 1234-acme-xxxx@5.6.7.8
```

where -acme-xxxx is a cookie and xxxx is the encoded version of 1.2.3.4.

- **from-to**—From and To headers will use the URL parameter for address encoding

The following example illustrates the functionality of a SBC using a use url parameter value of none:

```
sip: 1234@1.2.3.4 is translated into sip: 1234@5.6.7.8; pn=acme-xxxx
```

where -acme-xxxx is a cookie and xxxx is the encoded version of 1.2.3.4.

- **all**—All headers will use the URL parameter for address encoding. Acme Packet recommends not using this values because other SIP elements or implementations (other than the Oracle Communications Session Border Controller) might not retain the URL parameter in subsequent SIP messages that they send to the Oracle Communications Session Border Controller.
- **phone**—

If this field is set to either from-to or all, the SBC puts the encoded address of the SIP NAT into a URL parameter instead of using the encoding name inside the userinfo part of the address.

**16. parameter-name**—If you have configured the **use-url-parameter** with the from-to or all value, you need to indicate the hostname prefix.

The parameter name value is used in SIP NAT encoding addresses that have the use url parameter values of from-to or all.

**17. user-NAT-tag**—Enter a value to identify the username prefix used for SIP URIs. The values you can use can include any characters valid for the userinfo part of a URI. This should be made unique for each realm and SIP NAT function.

The default value is **-acme-**.

In combination with the domain suffix and host NAT tag values, this value is used to help the SBC identify an encoded URI that it needs to translate when moving between public and private realms.

**18. host-NAT-tag**—Enter a value for the host NAT tag field to identify the hostname prefix used for SIP URIs. The value refers to domain labels and can include any characters valid for the hostname part of the URI. This should be made unique for each realm and SIP NAT function.

The default value is **ACME-**.

In combination with the domain suffix and user NAT tag values, this value is used to help the SBC identify an encoded URI that it needs to translate when moving between public and private realms.

## SIP Signaling Services

**19. headers**—List the SIP headers you want affected by the SIP NAT function. The URIs in these headers are translated and encrypted, and encryption occurs according to the SIP NAT function rules.

To enter the full default list, type `headers`, followed by a Space and `-d`, then press Enter.

You can also insert the following tags in SIP NAT headers if you want to replace FQDNs with next hop or SIP interface IP addresses:

- `fqdn-ip-tgt`: replaces the FQDN with the target address
- `fqdn-ip-ext`: replaces the FQDN with the SIP NAT external address

Enter the tag using the following format:

```
<header-name>=<tag>
```

For example:

```
To=fqdn-ip-tgt
```

The FQDN in a `To` header is replaced with the target IP address.

You can insert the following tags to apply NAT treatment to a `From` header in an INVITE when the gateway sends it into the home realm.

- `ip-ip-tgt`: replaces any IP address in the `From` header with the next hop target
- `ip-ip-ext`: replaces any IP address in the `From` header with the SBC's external address

To view all SIP NAT function parameters, enter a `?` at the system prompt. The following example shows SIP NAT configuration for peering network.

```
sip-nat
    realm-id                peer-1
    domain-suffix          .p1.acme.com
    ext-proxy-address      192.168.11.200
    ext-proxy-port        5060
    ext-address            192.168.11.101
    home-address           127.0.0.10
    home-proxy-address    127.1.0.10
    home-proxy-port       5060
    route-home-proxy      enabled
    address-prefix        *
    tunnel-redirect       disabled
    use-url-parameter     none
    parameter-name
    user-nat-tag           -p1-
    host-nat-tag          P1-
    headers                Call-ID Contact From Join Record-
Route                    Refer-To Replaces Reply-To Route
To Via                   f i m r t v
```

## SIP Realm Bridging

This section explains how to configure the internal routing among realms known as realm bridging. Realm bridging lets you cross-connect SIP interfaces. You can use one of the following two methods for bridging realms:

- **local policy bridging**: use this method to enable dynamic internal routing between realms if your SIP interfaces do not have the SIP NAT function applied.
- **SIP NAT bridging**: use this method if your SIP interfaces have the SIP NAT function applied.

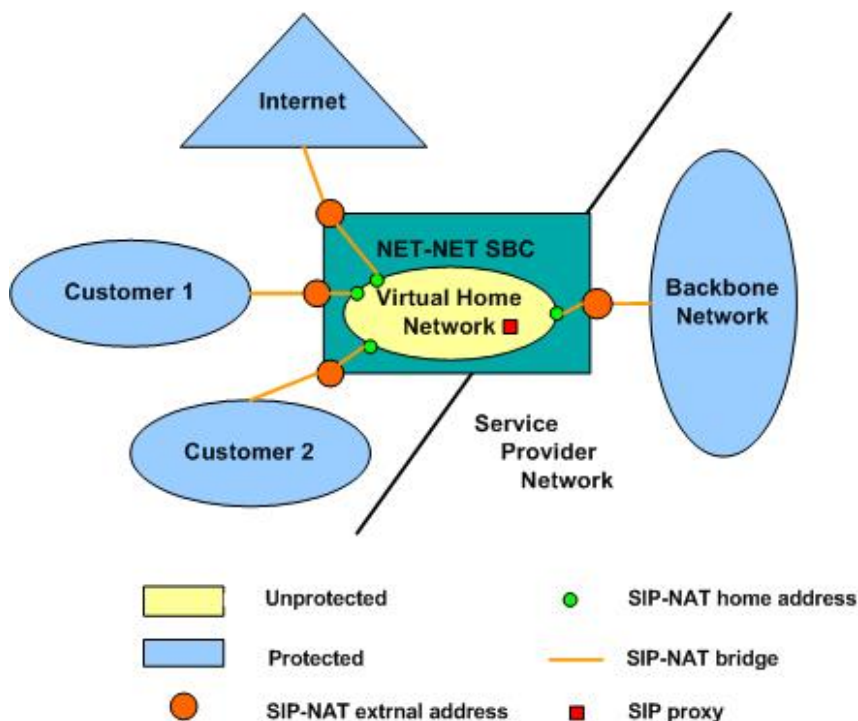
## About SIP NAT Bridging

Each SIP NAT has a presence in two realms, trusted and untrusted. The SIP NAT bridge is the conduit for packages in and out of the home realm. It creates a bridge between realms by providing address translations; removing all references to the original IP addressing from the packets sent to the destination network.

With the SIP NAT bridge, an untrusted (or public) home network can reside within the Oracle Communications Session Border Controller, while the other entities (the backbone network, the Internet, or customer networks) are all trusted (or private). One of the primary functions of the SIP NAT bridge is to protect networks from one another so that address bases can remain hidden. Using a SIP NAT bridge, no one network has direct access to the data of other networks.

Establishing a SIP NAT bridge lets you route every SIP Request message through the backbone. Without using this functionality, it would appear as though all messages/sessions were coming from the Oracle Communications Session Border Controller's SIP proxy (the SIP server that receives SIP requests and forwards them on behalf of the requestor).

The following diagram illustrates this unprotected (or public) and protected (or private) division.



## SIP NAT Bridge Configuration Scenarios

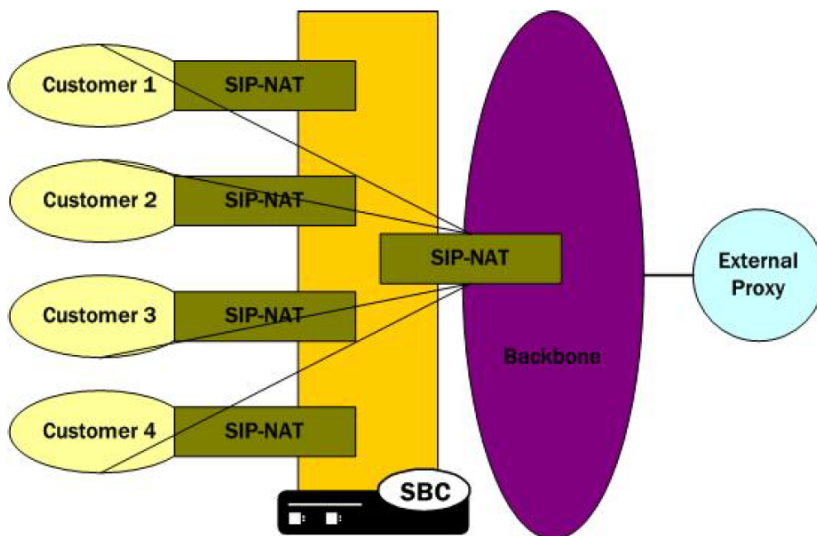
You can configure the SIP NAT bridge functionality in a many-to-one or a one-to-one relationship. For example, multiple customer SIP NATs can be tied to a single backbone SIP NAT, or a single customer SIP NAT can be tied to a single backbone SIP NAT.

You might need to use several SIP NATs on the customer side while using only one on the backbone side in a many-to-one relationship. Or you might configure one SIP NAT on the backbone side for every one that you configure on the customer side in a one-to-one relationship.

You can route all customer side SIP NAT requests to the corresponding backbone SIP NAT regardless of the Request URI. If a request arrives from the customer network with a Request URI that does not match the customer SIP NAT external address or the local policy that would route it to the backbone SIP NAT; the route home proxy value is used.

### Many to One Configuration

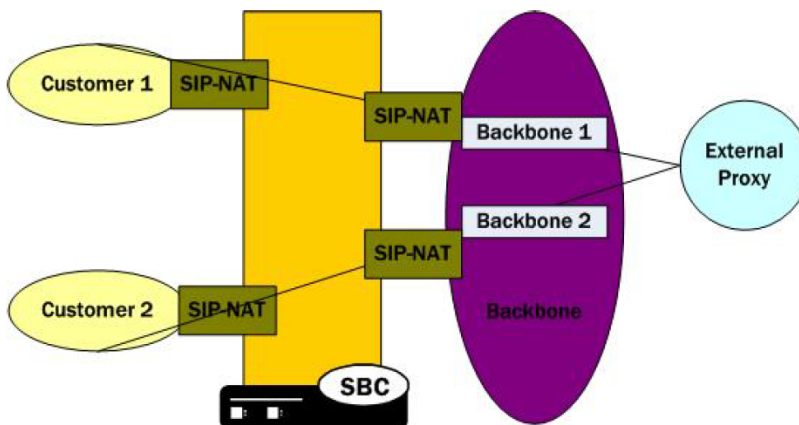
In the many-to-one scenario, multiple customer SIP NATs are tied to a single backbone SIP NAT. The following diagram illustrates the many-to-one SIP NAT bridge configuration.



### One-to-One Configuration

In the one-to-one scenario, a single customer SIP NAT is tied to a single backbone SIP NAT. On the backbone SIP NAT side, you configure the home proxy address to match the home address of the customer SIP NAT. On the customer side, you configure the home proxy address to match the home address of the backbone SIP NAT.

The following diagram illustrates the one-to-one SIP-NAT bridge configuration.



### SIP NAT Bridge Configuration

You create a bridge between SIP NATs by pointing them at one another. You point the SIP NATs at each other by configuring the home address and home proxy address to create the bridge. In addition, you can configure the route home proxy on the customer's side of a SIP NAT to force all requests to be routed to the corresponding backbone SIP NAT, regardless of the Request URI. You need to force requests when elements in the customer's network send requests with a Request URI that does not match the customer's SIP NAT external address. Or when the Request URI does not match a local policy element that would route the requests to the backbone SIP NAT.

You also need a home network to create a SIP NAT bridge. If you do not have a real home network, you need to create a virtual one. You also need to configure instances of the SIP NAT to create the SIP NAT bridge within your network.

## Creating a Virtual Home Network

A virtual home network is a home network that resides entirely within the Oracle Communications Session Border Controller, as does a real home network. The difference between the two is the real home network also has a physical connection to the Oracle Communications Session Border Controller.

The internal home realm/network is usually configured with addresses within the special loopback range (127.0.0.0/8) as described in RFC 3330. This applies to the SIP port addresses for the home realm's SIP interface, and all home addresses for SIP NATs. The address 127.0.0.1 should not be used because it conflicts with the default loopback interface setup by the system for inter-process communication.

To create a virtual home network:

1. Set the name and subport ID of the network interface associated with the home realm element to lo0:0.
2. To enable the SIP proxy to listen for messages on the virtual home realm, configure the home realm ID. It must correspond to the realm's identifier, in which you set the network interface subelement to point to the appropriate network interface element.

The following table lists the field values you need to set when you are using SIP NAT bridge functionality and you do not have a real home network.

Configuration Element		Sample Values
realm configuration	identifier	home
	network interfaces	lo0:0
	address prefix	127.0.0.0/8
SIP configuration	home realm ID	home
	SIP ports address	127.0.0.100

## Many-to-One Configuration

To configure many-to-one:

1. For the backbone SIP NAT, ensure the home proxy address field is blank.
2. For the customer side SIP NAT:

Set the home address to match the home address of the customer.

Set the home proxy address to match the backbone SIP NAT home address.

Set route home proxy to forced.

The following table lists the field values you need to set to create a many-to-one SIP NAT bridge.

SIP NAT Entity	Field	Sample Values
Backbone SIP NAT	home address	IPv4 address of the home realm. For example: 127.0.0.120
	home proxy address	IPv4 address of the home proxy from the perspective of the external realm. For a backbone SIP NAT, leave blank.
Customer SIP NAT	home address	127.0.0.120
	home proxy address	127.0.0.110
	route home proxy	forced

### One-to-One Configuration

In the one-to-one scenario, a single customer SIP NAT is tied to a single backbone SIP NAT. The home proxy address field value of the backbone SIP NAT must match the home address of the customer SIP NAT. On the customer side, the home address of the customer SIP NAT should be defined as the home address of the customer, the home proxy address field value should match the home address of the backbone SIP NAT, and route home proxy should be set to forced.

The following table lists the field values you need to set to create a one-to-one SIP NAT bridge.

SIP NAT Entity	Field	Sample Values
Backbone SIP NAT	home address	IPv4 address of the home realm. For example: 127.0.0.110
	home proxy address	IPv4 address of the home proxy from the perspective of the external realm. 127.0.0.120
Customer SIP NAT	home address	127.0.0.120
	home proxy address	127.0.0.110
	route home proxy	forced

### Shared Session Agent

Usually, the same set of servers (the external proxy) is used for all SIP NATs to the backbone network. In order to support redundant servers in the backbone of a SIP NAT bridge, the original egress realm as determined by the incoming Request URI needs to be retained after a local policy lookup.

When a request arrives at the Oracle Communications Session Border Controller, it determines the matching (target) session agent and, after the local policy is examined, sets the new outbound session agent to the one from the selected target.

If the target session agent's realm is set to \*, the Oracle Communications Session Border Controller retains the original session agent's realm ID. Because the target session agent does not have a realm ID defined, the original egress realm is retained.

### SIP Hosted NAT Traversal (HNT)

---

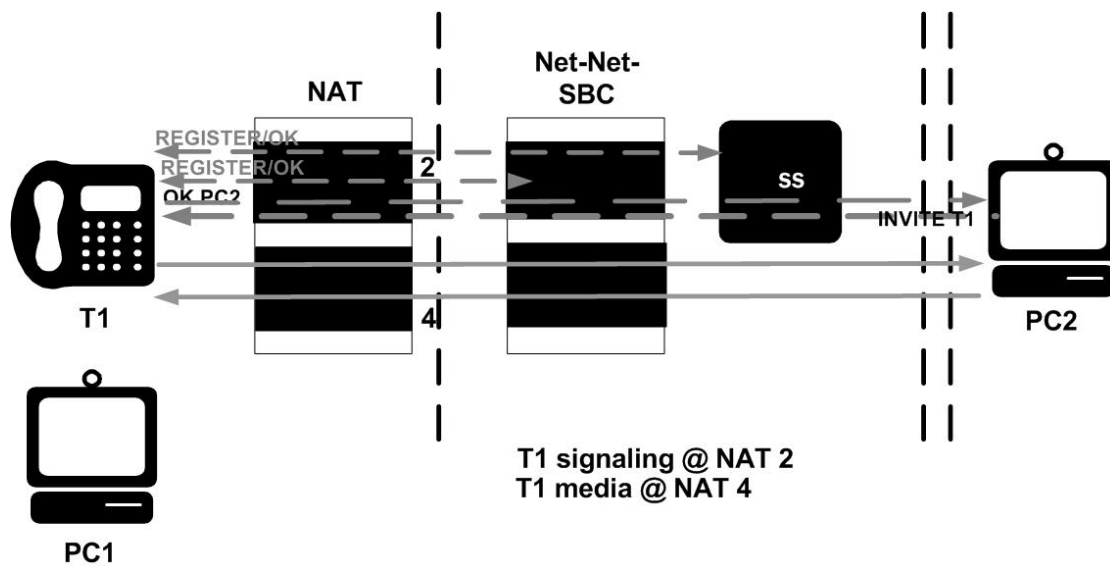
This section explains how to configure SIP Hosted Network Address Translation (HNT) traversal. SIP HNT lets endpoints behind a NAT/firewall device send and receive signaling and media using the Oracle Communications Session Border Controller as a relay.

#### About SIP HNT

SIP HNT is a technique the Oracle Communications Session Border Controller uses to provide persistent reachability for SIP UAs located in private Local Area Networks (LANs) behind Nat/firewall devices. It relies on frequent, persistent messaging to ensure that the binding on the intermediary NAT device is not torn down because of inactivity. HNT does not require support for the NAT in the SIP endpoint.

The following diagram illustrates SIP HNT traversal.z





The Oracle Communications Session Border Controller's HNT function allows endpoints located behind NATs to communicate; providing means to traverse NATs. The Oracle Communications Session Border Controller interacts with endpoints (using SIP) to allow persistent inbound and outbound signaling and media communications through these NATs.

The Oracle Communications Session Border Controller automatically detects when an intermediate NAT exists between the UA and the Oracle Communications Session Border Controller by comparing the Layer 3 IP address of a REGISTER message with the IP address indicated within the UA. The Oracle Communications Session Border Controller sends signaling responses to the address and port that the request came from, rather than the address and port indicated in the request. The Via header in the request message indicates where the response should be sent.

### Using HNT with Existing NAT Device

For network architectures in which premise devices and endpoints reside behind an existing NAT device, the Oracle Communications Session Border Controller's HNT function allows these premise NATs to be traversed without requiring an upgrade to the premise equipment, the deployment and management of additional premise-based hardware or software, or any NAT device configuration changes.

### Registering Endpoints

The Oracle Communications Session Border Controller uses periodic endpoint registration messages to dynamically establish and maintain bindings in the NAT. These bindings keep a signaling port (port that is opened on a firewall to allow traffic to pass through it is a pinhole) open in the NAT that allows the inbound signaled communications to pass through. Using the endpoint registrations, the Oracle Communications Session Border Controller then maps the Layer 3 (OSI network layer that deals with switching and routing technologies for data transmission between network devices) IPv4 address/port information from the NAT device to the Layer 5 (OSI session layer that deals with session and connection coordination between applications) entity (for example, user name or phone number) behind the NAT so that when an incoming signaling message is received, the Oracle Communications Session Border Controller sends it to the appropriate address and port on the NAT for the called party.

### Establishing Media Flows

During call setup, the ports for bidirectional media flows are established dynamically. Since the media flows also pass through the Oracle Communications Session Border Controller, it can identify the IPv4 address/port information on the NAT device used for the outgoing media coming from the user name/phone number. The Oracle Communications Session Border Controller then uses that same NAT's IPv4 address/port information to send incoming media to the correct user name/phone number behind the NAT device.

### Prerequisites

In order to achieve HNT, the endpoints involved must be capable of:

- symmetric signaling: sending and receiving SIP messages from the same transport address (IP address or User Datagram Protocol/Transmission Control Protocol (UDP/TCP) port)
- symmetric media: sending and receiving Real-Time Transport Protocol (RTP) messages from the same UDP port

These conditions are required to allow signaling and media packets back through the NAT (through the bound external address and port). These packets must come from the address and port to which the outbound packet that created the NAT binding was sent. The NAT sends these inbound packets to the source address and port of the original outbound packet.

When SIP HNT is used, the Oracle Communications Session Border Controller sends signaling responses to the address and port that the request came from rather than the address and port indicated in the request. The Via header in the request message indicates where the response should be sent.

### Keeping the NAT Binding Open

Additional measures are also required to keep the NAT binding open because most NAT bindings are discarded after approximately a minute of inactivity. The Oracle Communications Session Border Controller keeps the SIP NAT binding open by returning a short expiration time in REGISTER responses that forces the endpoint to send frequent REGISTER requests.

In order to keep the NAT binding open for SIP, the Oracle Communications Session Border Controller maintains the registration state. When an endpoint first registers, the Oracle Communications Session Border Controller forwards that REGISTER message on to the real registrar. You can define the real registrar using either of the following methods:

- Configure the SIP config registrar host and registrar port to indicate the real registrar.
- Map the SIP config registrar host and registrar port values to the SIP NAT home proxy address and home proxy port values. Then configure the SIP NAT's external proxy address and external proxy port values to correspond to the real registrar.



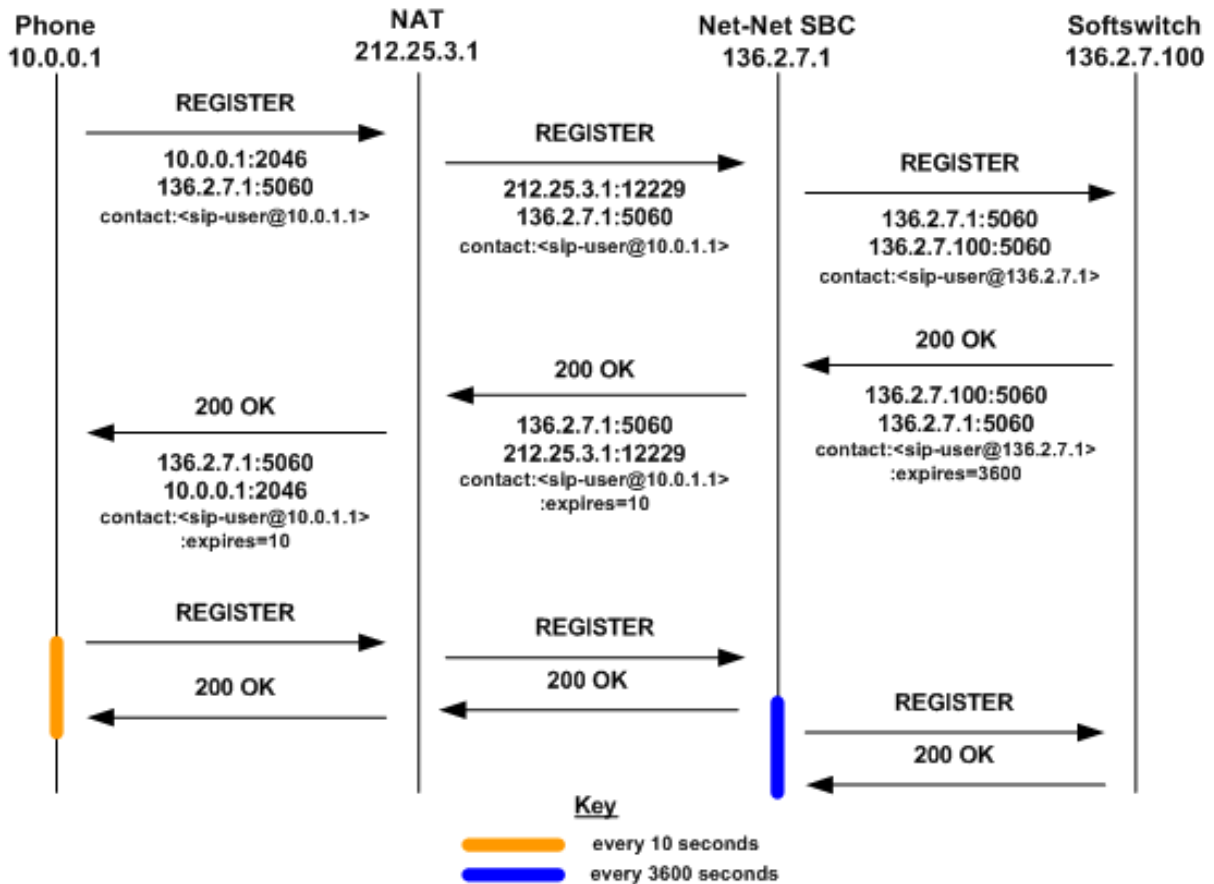
**Note:** A registrar can be located in a SIP NAT realm.

When a successful response is received, the Oracle Communications Session Border Controller caches the registration to memory. This cached registration lives for the length of time indicated by the expiration period defined in the REGISTER response message from the registrar. The response sent back to the endpoint has a shorter expiration time (defined by the SIP config's NAT interval) that causes the endpoint to send another REGISTER message within that interval. If the endpoint sends another REGISTER message before the cached registration expires, the Oracle Communications Session Border Controller responds directly to the endpoint. It does not forward the message to the real registrar.

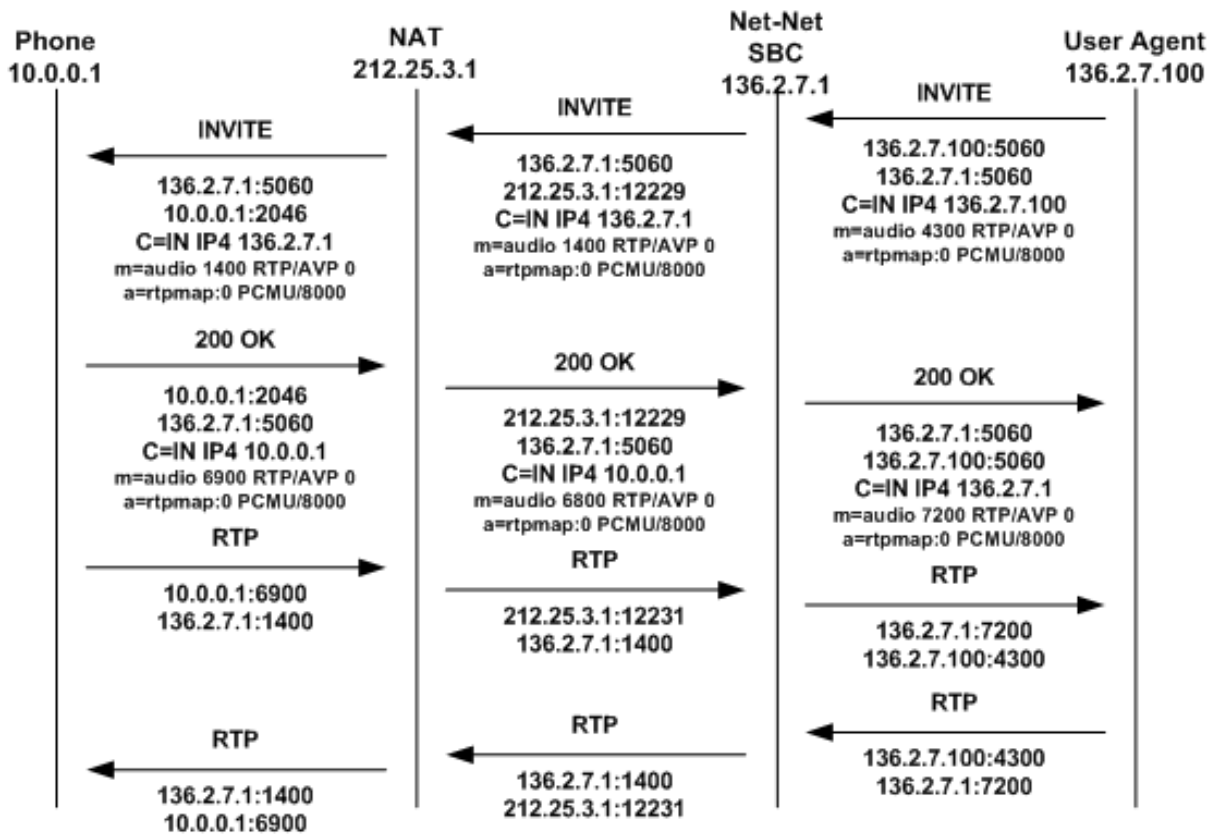
If the cached registration expires within the length of time indicated by the NAT interval, the REGISTER message is forwarded to the real registrar. If the Oracle Communications Session Border Controller does not receive another REGISTER message from the endpoint within the length of time indicated by the NAT interval, it discards the cached registration.

The Contact Uniform Resource Identifier (URI) in the REGISTER message sent to the registrar by the Oracle Communications Session Border Controller points at the Oracle Communications Session Border Controller so that the proxy associated with the real registrar sends inbound requests to the Oracle Communications Session Border Controller. This way, the inbound requests can be forwarded to the endpoint through the NAT binding.

The following example illustrates the SIP HNT registration call flow for the SIP HNT feature.



The following example illustrates the SIP HNT invitation call flow for the SIP HNT feature.



### Working with Multiple Domains

You can use a wildcard (\*) with the HNT feature to accommodate multiple domains and to allow the Oracle Communications Session Border Controller to cache all HNT endpoints. The wildcard functionality is enabled in the SIP config by entering an asterisk (\*) in the registrar domain and registrar host fields.

The wildcard allows the use of either a local policy or Domain Name Service (DNS) to resolve the domain name to the correct registrar. Either method can be used to route the Fully Qualified Domain Name (FQDN) when the you enter an asterisk (\*) for the register host. An FQDN consists of an unlimited number of domain labels (domain names), each separated by a dot (.). The FQDN can include the top level domain name (for example, acmepacket.com).

In the hostname acme-packet.domainlbl.example100.com, the syntax is as follows:

- acme-packet is a domain label
- domainlbl is a domain label
- example100 is a domain label
- com is the top label

The information configured in a local policy is used before DNS is used. If the next hop destination address (defined in the local policy's next hop field) is an IPv4 address, a DNS server is not needed. A DNS server is needed when the IPv4 address of the next hop destination address is a FQDN or cannot be determined from the Oracle Communications Session Border Controller's configuration. Even with a configured local policy, the next hop destination address might be an FQDN that requires a DNS lookup.

If the registrar host does not use the wildcard, the Oracle Communications Session Border Controller always uses the configured address. You can limit the number of endpoints that receive the HNT function. For example, you can use a non-wildcarded registrar domain field value (like acme.com) with a wildcarded registrar host field value.

### HNT Configuration Overview

To configure SIP HNT NAT traversal, you need to configure both the SIP interface and the SIP config.

#### SIP HNT Single Domain Example

The following example shows values entered for the SIP config and SIP interface elements to configure SIP HNT for a single domain and registrar.

- SIP config

Parameter	Sample Value
registrar domain	netnetsystem.com
registrar host	192.168.12.1
registrar port	5060

- SIP interface

Parameter	Sample Value
NAT traversal	always
NAT interval	60
minimum registration expire	200
registration caching	disabled
route to registrar	enabled

## SIP HNT Multiple Domain Example

The following example shows values entered for the SIP config and SIP interface elements to configure SIP HNT for a multiple domains and multiple registrars.

- SIP config

Parameter	Sample Value
registrar domain	*
registrar host	*
registrar port	0

- SIP interface

Parameter	Sample Value
NAT traversal	always
NAT interval	60
minimum registration expire	200
registration caching	disabled
route to registrar	enabled

## HNT Configuration

To configure a SIP interface on the Oracle Communications Session Border Controller (SBC):

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-interface) #
```

From this point, you can configure physical interface parameters. To view all SIP interface parameters, enter a ? at the system prompt.

4. **nat-traversal**—Define the type of HNT enabled for SIP. The default value is **none**. Available values include:

- **none**—Disables the HNT feature for SIP (default value)
- **rport**—SIP HNT function only applies to endpoints that include the rport parameter in the Via header and the sent-by of the topmost VIA matches the Contact-URI host address, both of which must be different from the received Layer 3 address.
- **always**—SIP HNT applies to requests when the sent-by of the topmost VIA matches the Contact-URI host address, both of which must be different from the received Layer 3 address. (Even when the rport parameter is not present.)

5. **nat-interval**—Set the expiration time in seconds for the SBC's cached registration entry for an HNT endpoint. The default value is **30**. The valid range is:

- Minimum—0
- Maximum—999999999

Acme Packet recommends setting the NAT interval to one-third of the NAT binding lifetime. A NAT binding lifetime is the network connection inactivity timeout. The value is configured (or hardwired) in the NAT device (firewall). This timer is used to prevent the NAT device from keeping an unused port open.

6. **registration-caching**—Enable for use with all UAs, not just those that are behind NATs. By default, this field is set to **disabled**. If enabled, the SBC caches the Contact header in the UA's REGISTER request when it is addressed to one of the following:

- SBC
- registrar domain value
- registrar host value

The SBC then generates a Contact header with the SBC's address as the host part of the URI and sends the REGISTER to the destination defined by the registrar host value.

Whether or not SIP HNT functionality is enabled affects the value of the user part of the URI sent in the Contact header:

- **enabled**—The SBC takes the user part of the URI in the From header of the request and appends a cookie to make the user unique. A cookie is information that the server stores on the client side of a client-server communication so that the information can be used in the future.
- **disabled**—The user part of the Contact header is taken from the URI in the From header and no cookie is appended. This is the default behavior of the Oracle Communications Session Border Controller.

When the registrar receives a request that matches the address-of-record (the To header in the REGISTER message), it sends the matching request to the SBC, which is the Contact address. Then, the v forwards the request to the Contact-URI it cached from the original REGISTER message.

7. **min-reg-expire**—Set the time in seconds for the SIP interface. The value you enter here sets the minimum registration expiration time in seconds for HNT registration caching. The default value is **300**. The valid range is:

- Minimum—1
- Maximum—999999999

This value defines the minimum expiration value the SBC places in each REGISTER message it sends to the real registrar. In HNT, the SBC caches the registration after receiving a response from the real registrar and sets the expiration time to the NAT interval value.

Some UAs might change the registration expiration value they use in subsequent requests to the value specified in this field. This change causes the SBC to send frequent registrations on to the real registrar.


8. **registration-interval**—Set the SBC's cached registration entry interval for a non-HNT endpoint. Enter the expiration time in seconds that you want the SBC to use in the REGISTER response message sent back to the UA. The UA then refreshes its registration by sending another REGISTER message before that time expires. The default value is **3600**. The valid range is:

- Minimum—1

A registration interval of zero causes the SBC to pass back the expiration time set by and returned in the registration response from the registrar.

- Maximum—999999999

If the expiration time you set is less than the expiration time set by and returned from the real registrar, the SBC responds to the refresh request directly rather than forwarding it to the registrar.

 **Note:** With registration caching, there is no NAT; therefore, a short registration interval causes the UA to send excess REGISTER messages.

Although the registration interval applies to non-HNT registration cache entries, and the loosely related NAT interval applies to HNT registration cache entries, you can use the two in combination. Using a combination of the two means you can implement HNT and non-HNT architectures on the same SBC. You can then define a longer interval time in the registration interval field to reduce the network traffic and load caused by excess REGISTER messages because there is no NAT binding to maintain.

9. **route-to-registrar**—Enable routing to the registrar to send all requests that match a cached registration to the destination defined for the registrar host; used when the Request-URI matches the registrar host value or the registrar domain value, not the SBC’s address. Because the registrar host is the real registrar, it should send the requests back to the SBC with the SBC’s address in the Request-URI. The default value is **disabled**. The valid values are:

- enabled | disabled

For example, you should enable routing to the registrar if your network uses a SBC and needs requests to go through its service proxy, which is defined in the registrar host field.

## Global SIP Configuration

To configure the SIP configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) #
```

From this point, you can configure SIP config parameters. To view all SIP config parameters, enter a ? at the system prompt.

4. **registrar-domain**—Optional. Define the domain to match against the host part of a URI to determine if a request is addressed to the registrar. If there is a match, the registration caching, NAT traversal, and route to registrar parameter values for the SIP interface are applied to the request. By default, this field remains empty. Available values are:

- an asterisk (\*) to specify the values apply to all requests.
- any alphanumeric character or any combination of alphanumeric characters. For example, acme1.com.

A hostname consists of any number of domain labels, separated by dots (.), and one top label. A top label is the last segment of the hostname. It must start with an alphabetical character. After the first character, a top label can consist of any number or combination of alphanumeric characters, including those separated by dashes. The dash must be preceded and followed by alphanumeric characters. A single alphabetical character is the minimum requirement for a hostname field (for example, c to indicate .com).

When the REGISTER message’s Request-URI has an FQDN, it is matched against the registrar domain’s value to determine if the message needs to be forwarded to the registrar port on the registrar host. The registrar domain’s value is also used when route to registrar is set to enabled, to determine if a request needs to be forwarded to the registrar.

Only the right-hand part of the domain name in the Request-URI needs to match the registrar domain value. For example, acme3.acmepacket.com matches acmepacket.com. However, the entire domain label within the domain name must match. For example, the domain label “acme3.acmepacket.com” would not match packet.com.

5. **registrar-host**—Define the address of the registrar for which requests for registration caching, NAT traversal, and router to registrar options apply. You can use a specific hostname, a IP address, or a wildcard (\*):

- an asterisk (\*) indicates normal routing (local policy, DNS resolution, and so on) is used to determine the registrar’s address.
- hostname: can consist of any alphanumeric character or any combination of alphanumeric characters (for example, acme1.com). The hostname can consist of any number of domain labels, separated by dots (.), and one top label. You can use the minimum field value of a single alphabetical character to indicate the top label value (for example, c to indicate .com).

## SIP Signaling Services

---

- IPv4 address: must follow the dotted notation format. Each of the four segments can contain a numerical value between zero (0) and 255. For example, 192.168.201.2. An example of a invalid segment value is 256.

By default, the registrar host field remains empty.

6. **registrar-port**—Set the SIP registrar port number. The SIP registrar server configured in this and the registrar host field is the real registrar. Or the values entered in those fields map to the home proxy address and home proxy port of the SIP NAT with external proxy address and external proxy port values that correspond to the real registrar. The default value is 0. The valid range is:

- Minimum—0, 1025
- Maximum—65535

The following example shows the values for a single domain and registrar configuration.

```
sip-config
  state                               enabled
  operation-mode                       dialog
dialog-transparency                   disabled
  home-realm-id                        acme
  egress-realm-id
  nat-mode                              Public
  registrar-domain
  registrar-host
  registrar-port                        0
  init-timer                            500
  max-timer                             4000
  trans-expire                          32
  invite-expire                         180
  inactive-dynamic-conn                 32
  red-sip-port                          1988
  red-max-trans                         10000
  red-sync-start-time                   5000
  red-sync-comp-time                    1000
  last-modified-date                   2005-03-19 12:41:28
```

## Endpoint-Initiated Keep-Alives

---

The Oracle Communications Session Border Controller provides three Keep-Alive algorithms designed to generate sufficient traffic flow to maintain NAT (Network Address Translation) bindings. Prior releases provided a single Keep-Alive algorithm that is described in the SIP Hosted NAT Traversal (HNT) section of the SIP Signaling Services chapter of the *ACLI Configuration Guide*. This method is SD-based and requires no explicit participation by the SIP endpoint. Instead the SD manipulates SIP registration requests and responses to spoof the endpoint — generating frequent and extraneous registration requests that generate a periodic traffic flow to maintain existing NAT bindings.

Unlike the current SD-centric method, the new methods require the active participation of the SIP endpoint. With these methods the SIP endpoint initiates a Keep-Alive negotiation with the SD that produces a periodic request/response message sequence which also generates sufficient traffic flow to maintain NAT bindings.

The new methods are based upon the following RFCs.

RFC 5626, *Managing Client-Initiated Connections in the Session Initiation Protocol (SIP)*, which, in Section 3.5.2, provides a framework for the implementation of an endpoint-initiated Keep-Alive on both TCP and UDP connections and provides two approaches to generating endpoint-initiated Keep-Alive traffic flows. One approach described in Section 4.4.1 of the RFC is restricted to connection-oriented transport protocols such as TCP. It defines an end-point initiated ping which requires an SD pong in response. A second approach described in Section 4.4.2 of the RFC is restricted to connectionless transport protocols such as UDP. It defines an endpoint-initiated STUN binding request which requires as SD-originated STUN binding response.

RFC 6223, *Indication of support for Keep-Alive*, was initially published as an internet draft (most recently as draft-ietf-sipcore-keep-12.txt). RFC 6223 defines a procedure that enables a SIP endpoint to signal its capability and willingness to send and receive periodic Keep-Alive messages to a device referred to by the RFC as an edge proxy, a



role performed by the SD. After receiving such a signal, the SD returns a response indicating its willingness to exchange Keep-Alives, and specifying the interval between Keep-Alive exchanges.

RFC 5389, *Session Traversal Utilities for NAT (STUN)*, defines STUN binding requests and responses in Section 6 of the RFC. Endpoints that support endpoint-initiated Keep-Alives over a UDP connection must be capable of constructing and transmitting a STUN binding request, and receiving and parsing a STUN binding response.

## Endpoint-Initiated Keep-Alive Negotiation

Endpoint-initiated Keep-Alive negotiation requires the presence of the keep parameter in the Via header of Registration requests and responses issued and received by a SIP endpoint. As defined in RFC 6223, keep is

A SIP Via header field parameter that a SIP entity can insert in the topmost Via header field that it adds to the request, to explicitly indicate willingness to send keep-alives towards its adjacent downstream SIP entity. A SIP entity can add a parameter value to the keep parameter in a response to explicitly indicate willingness to receive keep-alives from its adjacent upstream SIP entity.

While RFC 6223 allows the presence of the keep parameter in either REGISTER or INVITE requests and responses, the current implementation, is registration-based. keep parameters contained in methods other than REGISTER are ignored.

As shown in the following sample registration exchange, the SIP endpoint signals its willingness to exchange Keep-Alive messages by placing an unvalued keep parameter in the SIP Via header of a REGISTER request.

```
REGISTER sip:512@172.16.101.23:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.101.38:5070;branch=ddl;keep
From: "512" <sip:512@172.16.101.38:5070;transport=UDP>;tag=443322
To: "512" <sip:512@172.16.101.38:5070>
Call-ID:1-14400@172.16.101.38
CSeq: 1 REGISTER
Max-Forwards: 70
Contact: "512" <sip:512@172.16.101.38:5070>;expires:18000
User-Agent: ADTRAN_Total_Access_908e_(2nd_Gen)/A1.02.00.E
Content-Length: 0
```

The SD strips the keep parameter from the Via header of the REGISTER request, and forwards the request to the Registrar.

```
REGISTER sip:512@192.168.7.32:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.101.23:5060;branch=z9hG4bK3q43klh3dafekdepnbaqip04e1
From: "512" <sip:512@172.16.101.38:5070;transport=UDP>;tag=443322
To: "512" <sip:512@172.16.101.38:5070>
Call-ID: 1-14400@172.16.101.38
CSeq: 1 REGISTER
Max-Forwards: 69
Contact: "512" <sip:512@192.168.101.23:5060;transport=udp>;expires:18000
User-Agent: ADTRAN_Total_Access_908e_(2nd_Gen)/A1.02.00.E
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
```

The Registrar indicates successful registration with a 200 OK REGISTER response back to the SD.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.101.23:5060;branch=z9hG4bK3q43klh3dafekdepnbaqip04e1
From: "512" <sip:512@172.16.101.38:5070;transport=UDP>;tag=443322
To: "512" <sip:512@172.16.101.38:5070>;tag=b68b3d53a5a90225609112ff6c211bef.16a6
Call-ID: 1-14400@172.16.101.38
CSeq: 1 REGISTER
Contact: <sip:512@192.168.101.23:5060;transport=udp>;expires=3600
Server: OpenSER (1.3.0-notls (i386/linux))
Content-Length: 0
```

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The SD indicates its willingness to exchange Keep-Alives by assigning a value, which specifies the interval between Keep-Alives, to the keep parameter and re-inserting the parameter and its assigned value into the Via header of the REGISTER response.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.16.101.38:5070;branch=ddl;keep=20
From: "512" <sip:512@172.16.101.38:5070;transport=UDP>;tag=443322
To: "512" <sip:512@172.16.101.38:5070>;tag=b68b3d53a5a90225609112ff6c211bef.16a6
Call-ID: 1-14400@172.16.101.38
CSeq: 1 REGISTER
Contact: <sip:512@172.16.101.38:5070>;expires=3600
Server: OpenSER (1.3.0-notls (i386/linux))
Content-Length: 0
```

### Connection Oriented Keep-Alives

After the endpoint-initiated Keep-Alive exchange has been negotiated for a TCP connection, the SIP endpoint transmits a periodic ping at intervals specified by the value of the keep parameter. As shown below, the ping consists of carriage return (CR) and line feed (LF) characters.

```
CRLFCRLF
CR = %x0D
LF = %x0A
```

The SD responds with a pong as shown below.

```
CRLF
CR = %x0D
LF = %x0A
```

The request ping and the response pong are transmitted between SIP messages, and cannot be sent in the middle of SIP message. If the TCP connection is secured by TLS, the Keep-Alive requests and responses are transmitted within the TLS encrypted stream.

As specified in Section 4.4.1 of RFC 5626 the SD must respond to a ping within a 10-second interval. If the endpoint fails to receive a valid pong within that interval, it must treat the flow as failed.

### Connectionless Keep-Alives

After the endpoint-initiated Keep-Alive exchange has been negotiated for an unreliable UDP connection, the SIP endpoint, acting as a STUN client, transmits a periodic STUN binding request so that the interval between each request is randomly distributed between 80 and 100 percent of the value of the keep parameter. Assuming a parameter value of 20 seconds, for example, the SIP endpoint transmits a STUN binding request at random intervals between 16 and 20 seconds in length.

The STUN binding request has the following format.

```
0                                     1                                     2                                     3
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+
|0 0|          STUN Message Type          |          Message Length          |
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     Magic Cookie                                     |
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     Transaction ID (96 bits)                                     |
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+
```

The initial two bits of any STUN message are always 00

STUN Message Type

This 14-bit field contains the message class and method — for a STUN binding request, class is request and method is binding.

Message Length

This 16-bit field contains the length, in octets, of the Attribute section of the STUN binding request. Since the Attribute section is not present in a binding request, this field contains a value of 0.

Magic Cookie

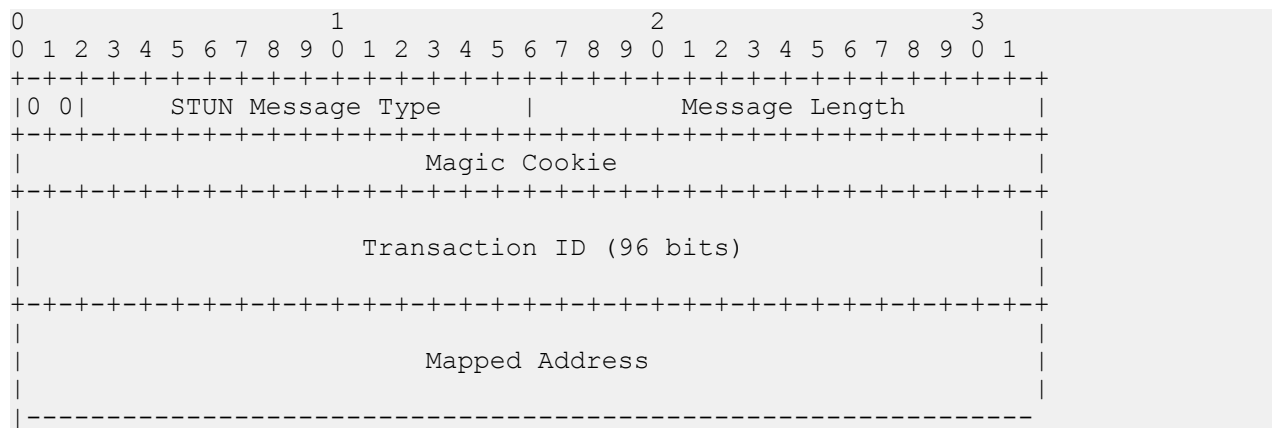
This 16-bit field always contains a value of 0x2112A442.

Transaction ID

This 96-bit field contains a random value that provides a transaction identifier. The value is generated by the STUN client and echoed in the STUN server response.

Upon receipt of a binding request, the SD, acting as a STUN server, transmits a STUN binding response. Receipt of the binding response by the endpoint confirms the UDP connection between the endpoint and the SD, and the viability of NAT bindings in the transmission path.

The STUN binding response has the following format.



The initial two bits of any STUN message are always 00.

STUN Message Type

This 14-bit field contains the message class and method — for a STUN binding response, class is response and method is binding.

Message Length

This 16-bit field contains the length, in octets, of the Attribute section of the STUN binding response. Since the binding response contains a single attribute, MAPPED-ADDRESS, the message length can be either 8 or 16 octets, depending on the IP address type.

Magic Cookie

This 16-bit field always contains a value of 0x2112A442.

Transaction ID

This 96-bit field contains a random value that provides a transaction identifier. The value is generated by the STUN client and echoed by the STUN server.

MAPPED-ADDRESS

This attribute contains the IP address and port number of the proximate NAT device, that is the address translator closest to the SD, that forwarded the STUN request toward the SD. Specific fields within the attribute identify the IP family (Version 4 or 6), the IP address, and port number. The attribute length is 8 octets if it contains an IPv4 address, or 16 octets if it contains an IPv6 address.

Once initiated, endpoint transmission of STUN binding requests and SD responses continue for the duration of the SIP Registration, 1 hour in the sample negotiation, or until the endpoint transmits a new REGISTER request. In the event of such a new request, the endpoint once again indicates its willingness to exchange STUN Keep-Alives with an unvalued keep parameter in the Via header. If endpoint-initiated Keep-Alive renegotiation is not successful, the endpoint must cease the transmission of Keep-Alive messages.

An endpoint failure to issue a timely STUN binding request is not fatal. In the absence of an expected request, the SD takes no action with regard to the UDP connection, or to current calls, nor does it remove entries from its registration cache. After reception of a tardy STUN request, the SD simply returns a STUN response.

In contrast, however, Section 4.4.2 of RFC 5626 states that an endpoint “considers the flow failed” if (1) it receives a malformed STUN bonding response from the SD, or (2) it fails to receive a timely binding response from the SD. If the endpoint receives a STUN binding response that contains a different MAP Address field from the previous response, It MUST treat this event as a failure on the flow.

### Endpoint-Initiated Keep-Alives Configuration

You use the **registrar-keep-alive** attribute, available in SIP Interface configuration mode, to enable endpoint-initiated Keep-Alives on a SIP interface.

1. In Superuser mode, use the following CLI command sequence to access SIP Interface configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

2. The **registrar-keep-alive** attribute enables endpoint-initiated Keep-Alive on the current SIP interface.

**none** — (the default) disables endpoint-initiated Keep-Alive processing

**always** — assuming that the endpoint has included the keep parameter in the Via REGISTER request header, forces the SD to place a keep parameter and an assigned in the Via REGISTER response — thus enabling endpoint-initiated Keep-Alive exchange with that endpoint

**bnat** — assuming that the endpoint has included the keep parameter in the Via REGISTER request header, forces the SD to place a keep parameter and an assigned value in Via REGISTER response only if the requesting endpoint is identified as being behind an intervening NAT device (based on comparing source IP packet addresses with IP addresses extracted from the SIP request).

```
ACMEPACKET(sip-interface)# registrar-keep-alive bnat
ACMEPACKET(sip-interface)#
```

3. If Keep-Alive is enabled on the current SIP interface (**registrar-keep-alive** is **always** or **bnat**), use the **nat-interval** attribute to specify the value of the keep parameter provided by the SD to SIP endpoints served by a UDP connection.

Allowable values are integers within the range 1 through 999999999 (seconds).

In the absence of an explicit assignment, this attribute defaults to a value of 30 seconds.

The SIP endpoint transmits periodic STUN binding requests so that the interval between each request is randomly distributed between 80 and 100 percent of the value of the **nat-interval** attribute.

Assuming the default value (30 seconds) the interval between STUN binding requests would vary from 24 to 30 seconds. This default value closely aligns to Section 4.4.2 of RFC 5626, which recommends that the time between each keep-alive request SHOULD be a random number between 24 and 29 seconds.

```
ACMEPACKET(sip-interface)# nat-interval 20
ACMEPACKET(sip-interface)#
```

4. If Keep-Alive is enabled on the current SIP interface (**registrar-keep-alive** is **always** or **bnat**), use the **tcp-nat-interval** attribute to specify the value of the keep parameter provided by the SD to SIP endpoints served by a TCP or TCP/TLS connection.

Allowable values are integers within the range 1 through 999999999 (seconds).

In the absence of an explicit assignment, this attribute defaults to a value of 90 seconds.

The SIP endpoint transmits periodic pings at intervals specified by this attribute.

```
ACMEPACKET(sip-interface) # tcp-nat-interval 120
ACMEPACKET(sip-interface) #
```

5. Use **done**, **exit**, and **verify-config** to complete this configuration.
6. If necessary, repeat Steps 1 through 5 to configure Keep-Alives on additional SIP interfaces.

## SD-originated Keep-Alive Negotiation

SD-originated Keep-Alives are useful on TCP connections with iOS (iPhone Operating System) clients, such as iPhones or iPads. iOS places inactive clients in a dormant state after a period of time. Once in the dormant state, clients cannot transmit data unless awakened by network traffic. SD-originated Keep-Alives are generated at a sufficient frequency to prevent iOS clients from entering the dormant state.

SD-originated Keep-Alive negotiation is similar to the earlier described negotiation, except for the parameter used. SD-originated Keep-Alive negotiation requires the presence of a well known proprietary parameter in the Via header of Registration requests and responses issued and received by a SIP endpoint. The proprietary parameter can be defined as follows:

A SIP Via header field parameter that a SIP endpoint can insert in the topmost Via header field that it adds to the request, to explicitly indicate willingness to receive Keep-Alives from its adjacent downstream SIP entity. This same SIP endpoint can assign a value to this proprietary parameter in the response to specify the keep-alive interval.

As shown in the following sample registration exchange, the SIP endpoint signals its willingness to receive Keep-Alive messages by placing an rkeep parameter in the SIP Via header of a REGISTER request; the assigned value of 20 specifies the interval between Keep-Alives.

```
REGISTER sip:512@172.16.101.23:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.101.38:5070;branch=ddl;rkeep=20
From: "512" <sip:512@172.16.101.38:5070;transport=UDP>;tag=443322
To: "512" <sip:512@172.16.101.38:5070>
Call-ID: 1-14400@172.16.101.38
CSeq: 1 REGISTER
Max-Forwards: 70
Contact: "512" <sip:512@172.16.101.38:5070>;expires:18000
User-Agent: ADTRAN_Total_Access_908e_(2nd_Gen)/A1.02.00.E
Content-Length: 0
```

The SD strips the rkeep parameter from the Via header of the REGISTER request, and forwards the request to the Registrar.

```
REGISTER sip:512@192.168.7.32:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.101.23:5060;branch=z9hG4bK3q43klh3dafekdepnbaqip04e1
From: "512" <sip:512@172.16.101.38:5070;transport=UDP>;tag=443322
To: "512" <sip:512@172.16.101.38:5070>
Call-ID: 1-14400@172.16.101.38
CSeq: 1 REGISTER
Max-Forwards: 69
Contact: "512" <sip:512@192.168.101.23:5060;transport=udp>;expires:18000
User-Agent: ADTRAN_Total_Access_908e_(2nd_Gen)/A1.02.00.E
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
```

The Registrar indicates successful registration with a 200 OK REGISTER response back to the SD. The expires parameter in the Contact header grants a registration period of 1 hour (3600 seconds).

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.101.23:5060;branch=z9hG4bK3q43klh3dafekdepnbaqip04e1
From: "512" <sip:512@172.16.101.38:5070;transport=UDP>;tag=443322
To: "512" <sip:512@172.16.101.38:5070>;tag=b68b3d53a5a90225609112ff6c211bef.16a6
```

## SIP Signaling Services

```
Call-ID: 1-14400@172.16.101.38
CSeq: 1 REGISTER
Contact: <sip:512@192.168.101.23:5060;transport=udp>;expires=3600
Server: OpenSER (1.3.0-notls (i386/linux))
Content-Length: 0
```

After inserting the rkeep parameter and assigned value in the Via header of the REGISTER response, the SD forwards the response to the endpoint. The presence of the rkeep parameter signals the SD's willingness to transmit Keep-Alive messages, and the parameter value specifies the exchange frequency in seconds.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.16.101.38:5070;branch=ddl;rkeep=20
From: "512" <sip:512@172.16.101.38:5070;transport=UDP>;tag=443322
To: "512" <sip:512@172.16.101.38:5070>;tag=b68b3d53a5a90225609112ff6c211bef.16a6
Call-ID: 1-14400@172.16.101.38
CSeq: 1 REGISTER
Contact: <sip:512@172.16.101.38:5070>;expires=3600
Server: OpenSER (1.3.0-notls (i386/linux))
Content-Length: 0
```

### SD-Originated Keep-Alive Format

After the Keep-Alive exchange has been negotiated for a TCP connection, the SD, transmits a periodic ping at a frequency specified by the proprietary parameter (rkeep in the example negotiation). As shown below, the ping consists of carriage return (CR) and line feed (LF) characters.

CRLF  
CRLF

CR = %x0D

LF = %x0A

pings are transmitted between SIP messages, and cannot be sent in the middle of SIP message. If the TCP connection is secured by TLS, pings are transmitted within the TLS encrypted stream.

### SD-Initiated Keep-Alives Configuration

Use the following procedure to configure SD-initiated Keep-Alives.


1. In Superuser mode, use the following CLI command sequence to access SIP Interface configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

2. The **options sbc-initiated-keep-alive-name** attribute enables SD-initiated Keep-Alive on the current SIP interface and specifies the name of the proprietary Via header parameter used during the SD-initiated Keep-Alive negotiation.

Prefix the command with a plus (+) sign to avoid over-writing existing options configuration.

The following example enables SD-initiated Keep-Alives and identifies the proprietary rkeep header as the header that contains negotiation information.

 **Note:** Endpoints require prior knowledge of this proprietary header name because the SD lacks the capability to relay configuration data to relevant endpoints.

```
ACMEPACKET(sip-interface)# options +sbc-initiated-keep-alive-name=rkeep
ACMEPACKET(sip-interface)#
```

3. The **option sbc-initiated-keep-alive-interval** attribute specifies the frequency of SD-initiated Keep-Alives.
4. If SD-initiated Keep-Alive is enabled on the current SIP interface (**options sbc-initiated-keep-alive-name** identifies a proprietary Via header parameter used for Keep-Alive negotiations), use the **options**

**sbc-initiated-keep-alive-interval** attribute to specify the default interval between SD-initiated pings.

Prefix the command with a plus (+) sign to avoid over-writing existing options configuration.

Allowable values are integers within the range 1 through 999999999 (seconds).

In the absence of an explicit assignment, this attribute defaults to a value of 30 seconds.

This default interval will be used only the proprietary parameter proposed by the endpoint lacks a value.

```
ACMEPACKET(sip-interface) # options +sbc-initiated-keep-alive-interval=45
```

```
ACMEPACKET(sip-interface)#
```

5. Use **done**, **exit**, and **verify-config** to complete this configuration.
6. If necessary, repeat Steps 1 through 5 to configure SD-initiated Keep-Alives on additional SIP interfaces.

## Statistics

The SD does not gather or maintain endpoint-initiated Keep-Alive statistical counts. However, malformed STUN binding requests and responses are included in the count of mal-formed SIP messages. Additionally, if DoS is enabled, STUN binding exchanges are included in endpoint message counts.

## SIP Registration Local Expiration

When you deploy multiple Oracle Communications Session Border Controllers (SBC) in series and they have registration caching and HNT configured, registration cache entries might expire prematurely in instances with several devices provisioned with the same address of record (AoR). Now you can configure a SIP interface option to prevent the premature expiration.

When you use registration caching and HNT, the SBC adjusts the expiration time it sends to user agents (UAs) in REGISTER responses based on the registration interval you configure. It can be the case that a SIP user has multiple registered contact endpoints at the UA to which a response is sent. If the URI in the Contact contains the UA's address and that UA included the Contact in the REGISTER request, then the Contact is seen as exclusively belonging to that UA. In the REGISTER response, this Contact (exclusive to the UA) includes the local expiration time, a time based on the SIP interface configuration's registration or NAT interval value. Additional Contacts (not exclusive to the UA) in the REGISTER response have the expiration time from the REGISTER response the registrar sent to the SBC.

It is this default behavior can cause registration cache entries to expire prematurely in the SBC nearest a registrar when multiple SBCs are deployed in series. Multiple registering UAs for a single SIP user, for example, might trigger the early expiration. The SIP you can configure an option per SIP interface that causes the SBC to send the local registration expiration time in all in the Expires parameter of all Contact headers included in REGISTER responses sent from the SIP interface.

## SIP Registration Local Expiration Configuration

You can configure this feature either for the global SIP configuration, or for an individual SIP interface.

To configure SIP registration local expiration for the global SIP configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

3. Type **sip-config** and press Enter. If you are editing an existing configuration, select the configuration so you can enable this feature.

## SIP Signaling Services

---

```
ACMEPACKET(session-router) # sip-config
ACMEPACKET(sip-config) #
```

4. **options**—Set the options parameter by typing **options**, a Space, the option name **reg-local-expires** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(sip-config) # options +reg-local-expires
```

If you type **options** and then the option value for either of these entries without the plus sign, you will overwrite any previously configured options. In order to append the new option to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

To configure SIP registration local expiration for an individual SIP interface:

6. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

7. Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

8. Type **sip-interface** and press Enter. If you are editing an existing configuration, select the one on which you want to enable this feature.

```
ACMEPACKET(session-router) # sip-interface
ACMEPACKET(sip-interface) #
```

9. **options**—Set the options parameter by typing **options**, a Space, the option name **reg-local-expires** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(sip-interface) # options +reg-local-expires
```

If you type **options** and then the option value for either of these entries without the plus sign, you will overwrite any previously configured options. In order to append the new option to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

10. Save and activate your configuration.

## Simultaneous TCP Connection and Registration Cache Deletion

---

You can configure the Oracle Communications Session Border Controller to automatically start a timer when a user deregisters or changes location by registering with a new contact address.

Not all devices tear down TCP connections associated with these old addresses when a user registers with a new contact address.

### Registration Cache Deletion Configuration

You can apply **suppress-reinvite** to the sip-interface facing the User Agents whose re-INVITES are to be responded to locally.

To enable Registration Cache Deletion:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```



3. Type **sip-interface** and press Enter. If you are adding this feature to a pre-existing configuration, you will need to select and edit it.

```
ACMEPACKET(session-router) # sip-interface
ACMEPACKET(sip-interface) #
```

4. **tcp-conn-dereg**—The delay (in seconds) that is used to determine when to terminate the TCP connection for a user that has been removed from the registration cache or changed location. This feature can be disabled by setting the value to zero.

```
ACMEPACKET(session-agent) # tcp-conn-dereg 5300
```

5. Save your work.

## SBC Incorrectly Appends Cookie in SIP REGISTER Message

The Oracle Communications Session Border Controller does not recognize a SIP URI containing tel-URI information if it doesn't also contain a "user=phone" parameter. This behavior adversely affects creation of the acme\_nat tag and placement of the cookie.

You can enable the option **proces-implicit-tel-URI** to recognize an implicit tel-URI and places the cookie in the correct location.

### process-implicit-tel-URI Configuration

To enable process-implicit-tel-URI

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET(session-router) # sip-config
ACMEPACKET(sip-config) #
```

4. **options**—Set the options parameter by typing **options**, a Space, the option-name **process-implicit-tel-URI** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(sip-config) # options +process-implicit-tel-URI
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the SIP interface configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

## SIP HNT Forced Unregistration

If you use HNT and experience the issue explained in this section, consider using the Oracle Communications Session Border Controller (SBC) forced unregistration feature. When this feature is enabled and a registration entry for an endpoint expires, the SBC notifies the soft switch to remove this binding using REGISTER message. In that REGISTER message, the expires header will be set to 0 and the expires parameter in the Contact header will also be set to 0.

The benefits of using forced unregistration include:

- Leveraging existing HNT configuration to provide near real-time information about the UA's status to the registrar/softswitch

## SIP Signaling Services

---

- Preserving resource utilization for the SBC and the softswitch by deleting a contact binding that is no longer valid or needed
- Preventing extra bindings from being generated at the softswitch (e.g., in instances when the UA or NAT restart)

This feature applies to:

- HNT endpoints with registration caching enabled by default, and when the **nat-traversal** parameter in the SIP interface configuration is set to always
- non-HNT endpoints with registration caching enabled, when the registration-interval parameter in the SIP interface configuration is used in the expires header sent to the UA in the 200 OK

## When to Use Forced Unregistration

For typical HNT use, it is common that the registration interval between the client UA and the Oracle Communications Session Border Controller (SBC) is between 60 and 120 seconds. This differs significantly from the re-registration interval between the SBC and the registrar, which varies from approximately 30 to 60 minutes.

If the UA fails to refresh its registration, the contact binding at the SBC is deleted after the registration expires. This expiration is determined by the expires= header in the 200 OK. The binding at the real registrar will remain intact. This creates a discrepancy between the real state of the UA and state of the softswitch. In the best case scenario, the contact binding expires at the softswitch after a few minutes.

For network management, this discrepancy can be problematic because the service provider would be unaware of the UA's status until the binding expires at the softswitch. This can take a considerable amount of time to happen.

In addition, the SBC encodes a cookie in the userinfo of the Contact header in the REGISTER message. This is a function of the source IPv4 address and port from which the request came, i.e., the ephemeral port in the NAT for DSL scenarios. Therefore, additional bindings that remain for long periods of time are created at the registrar if, for example, the:

- UA reboots
- Ethernet link between the UA and the DSL router is lost for over two minutes
- DSL crashes
- DSL/ATM layer between the DSL router

## Caution for Using Forced Unregistration

You should use caution when applying SIP HNT forced unregistration for the following reasons:

- It can have an impact on the performance of your Oracle Communications Session Border Controller and the registrar, especially when you have a large number of HNT endpoints in your configuration that become unavailable simultaneously.
- It is possible that the registrar might become vulnerable to overload in the case where the registrar must authenticate a large number of register messages generated when HNT endpoints are de-registered. It is possible that the cached registration credentials might become “stale” over time (e.g., the nonce value usually has a limited lifetime). Without proper credentials, the registrar will reject the de-registrations.

Given these concerns, we recommend that you consult with your Oracle systems engineer before adopting the use of forced unregistration.

## SIP HNT Forced Unregistration Configuration

To enable SIP HNT forced unregistration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the **session-router** path.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router) # sip-config
```

4. Use the ACLI **select** command so that you can work with the SIP configuration.

```
ACMEPACKET(sip-config) # select
```

5. **options**—Set the options parameter by typing **options**, a Space, the option name **force-unregistration**, and then press Enter.

```
ACMEPACKET(sip-config) # options +force-unregistration
```

**If you type options force-unregistration, you will overwrite any previously configured options. In order to append the new option to the sip-config's options list, you must prepend the new option with a plus sign as shown in the previous example.**

## Adaptive HNT

This section explains how to configure adaptive HNT. The adaptive HNT expires feature allows the Oracle Communications Session Border Controller to automatically determine the maximum SIP REGISTER message expires time interval in order to keep each individual NAT pinhole open when performing SIP HNT. This feature applies only to SIP over UDP.

### Overview

Without adaptive HNT, the Oracle Communications Session Border Controller keeps NAT pinholes open and port mapping cached by forcing the UAC to send frequent SIP REGISTER messages. It does so by setting the expires time to a short interval. Some NATs only need a message to be sent by the private client once every twenty minutes, while other NATs delete their cache/pinhole in thirty seconds if no messages appear. Given this large variation in time intervals, the Oracle Communications Session Border Controller's nat-interval (expire time) has been set to a low value in order to support as many NAT types as possible. However, CPU performance and scalability issues result from such a small refresh time, especially when there is a very large number of potential registered users.

When you use adaptive HNT, the Oracle Communications Session Border Controller waits for a time interval and then sends a SIP OPTIONS message to the UAC to see if it can still be reached. If the UAC can still be reached, the Oracle Communications Session Border Controller increases the timer and tries again. In case the pinhole closes because it has exceeded the NAT's cache time, the Oracle Communications Session Border Controller sets the expires time to be slightly longer than the time it tests using the OPTIONS method. This way, the UAC will send another REGISTER message shortly thereafter and impact on service will be minimal.

### Adaptive HNT Example

An example call flow using adaptive HNT involves a basic HNT user and a Oracle Communications Session Border Controller. It begins when the Oracle Communications Session Border Controller receives and forwards the 200 OK for the REGISTER message. Then the Oracle Communications Session Border Controller sends an expires timer for slightly longer than the time for which to test; in this example, it begins the test for the amount of time set for the minimum NAT interval. It adds ten seconds to this time when it sends the expires timer. This way, there is time for the OPTIONS message to be sent before the REGISTER message is received (which would refresh the NAT's cache). The Oracle Communications Session Border Controller also tries to keep the REGISTER time short enough so that even if the NAT pinhole closes, there is minimal time before the UAC creates a new NAT binding by sending another REGISTER. Because a ten second interval may be too long, you might want to set this value to a better-suited time.

The test succeeds with a minimum test-timer because the UAC responded to the OPTIONS message. So the test-timer value is increased by thirty seconds and tried again. The expires time in the REGISTER message will be increased to the test-timer value plus ten seconds. This time, the UAC does not respond to the OPTIONS message even though it was sent multiple times. Because the OPTIONS fails, when the Oracle Communications Session Border Controller

receives another REGISTER, it responds with the previously successful timer value (in this case, the minimum NAT interval).

However, if the OPTIONS request succeeds, then the Oracle Communications Session Border Controller persists with the test until it fails or until the maximum NAT timer value is reached. In this case, when the OPTIONS message fails, the Oracle Communications Session Border Controller uses the last successful test-timer value as the time for the expires header in the 200 OK for the REGISTER message.

### Synchronize A-HNT Successful Timer to Standby

Adaptive HNT enables the Oracle Communications Session Border Controller to determine, through testing, an optimum SIP REGISTER expires time interval that keeps the NAT pinhole open. For an HA node, this successful time value is determined through testing by the active system and then replicated to the standby. If there is a switchover during the active system's testing process, then it will restart for that endpoint.

### Adaptive NHT Configuration

You configure the SIP interface to set the state of this feature and to define the increments of time the Oracle Communications Session Border Controller uses to perform adaptive HNT. Remember that the Oracle Communications Session Border Controller uses the time you specify as the NAT interval, the supported time interval, as the basis on which to begin testing.

To configure adaptive HNT:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the **session-router** path.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-interface
```

4. **sip-dynamic-hnt**—Enable this parameter if you want to use adaptive HNT. The default value is **disabled**. The valid values are:
  - enabled | disabled
5. **max-nat-interval**—Set the amount of time in seconds that testing should not exceed. The Oracle Communications Session Border Controller will keep the expires interval at this value. The default value is **3600**. The valid range is:
  - Minimum—0
  - Maximum—999999999
6. **nat-int-increment**—Set the amount of time in seconds to use as the increment in value in the SIP expires header. The default value is **10**. The valid range is:
  - Minimum—0
  - Maximum—999999999
7. **nat-test-increment**—Set the amount of time in seconds that will be added to the test timer. The default value is **30**. The valid range is:
  - Minimum—0
  - Maximum—999999999

## SIP IP Address Hiding and NATing in XML

Adding to its topology hiding and NAT capabilities, the Oracle Communications Session Border Controller now performs those functions for pertinent IP addresses that are not part of the standard SIP message header format. Previously, such addresses were visible to the next hop in the SIP session path.


Note that this feature adds to the Oracle Communications Session Border Controller's pre-existing ability to perform this function for XML messages; this new support is specifically for the keyset-info message type.

For incoming SIP NOTIFY messages, the Oracle Communications Session Border Controller searches for the application/keyset-info+xml content type in the message. When it finds this content type, it searches further to detect the presence of <di:remote-uri> or <di:local-uri> XML tags and then NATs the IP addresses in the tags it finds. Specifically, the Oracle Communications Session Border Controller changes:

- The <di:remote-uri> IP address to be the egress SIP interface's IP address
- The <di:local-uri> IP address to be the IP address of the next hop to which the message is being sent

### Sample SIP NOTIFY with NATed XML

The following is a sample SIP NOTIFY message as it might arrive at the Oracle Communications Session Border Controller.

 **Note:** that it contains the <di:remote-uri> or <di:local-uri> XML tags on which the system will perform NAT; these lines appear in bold text.

```
NOTIFY sip:15615281021@10.152.128.253:5137;transport=udp SIP/2.0
To: 15615281021 <sip:15615281021@10.152.128.102:5080>;tag=5c93d019904036a
From: <sip:15615281021@10.152.128.102:5080>;tag=test_tag_0008347766
Call-ID: 3215a76a979d0c6
CSeq: 18 NOTIFY
Contact: <sip:15615281021@10.152.128.102:5080;maddr=10.152.128.102>
Via: SIP/2.0/UDP 10.152.128.102:5060;branch=z9hG4bK_brancha_0023415201
Event: keyset-info
Subscription-state: active;expires=2778
Accept: application/keyset-info+xml
Content-Type: application/keyset-info+xml
Content-Length: 599
Max-Forwards: 70
<?xml version="1.0"?>
<keyset-info xmlns="urn:ietf:params:xml:ns:keyset-info"
  version="16"
  entity="15615281021">
  <ki-data>
    <ki-state>"active"</ki-state>
    <ki-event>"unknown"</ki-event>
  </ki-data>
  <di:dialog id="dialog_id_201" call-
id="1395216611-1987932283256611-11-0884970552" local-
tag="test_tag_0008347790" direction="recipient">
    <di:state>trying</di:state>
    <di:duration>2778</di:duration>
    <di:local-uri>sip:15615281021@10.152.128.253:5137</di:local-uri>
    <di:remote-uri>sip:1004@10.152.128.102</di:remote-uri>
  </di:dialog>
</keyset-info>
```

Once the Oracle Communications Session Border Controller has completed the NAT process, the <di:remote-uri> and <di:local-uri> XML tags look like this

```
<di:local-uri>sip:15615281021@192.168.200.99:5137</di:local-uri>
<di:remote-uri>sip:1004@192.168.200.49</di:remote-uri>
```

because egress the SIP interface's IP address is 192.168.200.49 and the next hop's IP address is 192.168.200.99.

This feature does not require any configuration.

## SIP Server Redundancy

---

This section explains how to configure SIP server redundancy. SIP server redundancy involves detecting that an upstream/downstream SIP signaling entity has failed, and adapting route policies dynamically to remove it as a potential destination.

### Overview

You establish SIP server redundancy by creating session agents, which are virtual representations of the SIP signaling entities. These agents are then collected into a session agent group, which is a logical collection of two or more session agents that behaves as a single aggregate entity.

Rather than direct signaling messages to a single session agent (IP), the signaling message is directed to a session agent group (SAG). The group will have a set distribution pattern: hunt, round robin, proportionally distributed, and so on. Signaling is spread amongst the agents using this chosen pattern.

You direct the signaling message by configuring a route policy, known as a local policy, which determines where SIP REQUESTS should be routed and/or forwarded. The values in the To and From headers in the SIP REQUEST are matched with the content of the local policy within the constraints set by the session agent's previous hop value and SIP interface values such as the list of carriers.

To summarize, you need:

- two or more session agents
- a session group containing those session agents
- a local policy which directs traffic to the session agent group

### Configuration Overview

You make a session agent group a target by using a local policy to select the next hop from the members of a session agent group. You need to set the replace URI field of the configured local policy to enabled; which causes NAT rules such as realm prefixing to be overridden. The replace URI field allows you to indicate whether the local policy's value is used to replace the Request-URI in outgoing requests. This boolean field can be set to either enabled or disabled.

When the SIP NAT's route home proxy field is set to forced, it forces the Request to be forwarded to the home proxy without using a local policy. When this option is set to either disabled or enabled and the Request-URI matches the external address of the SIP NAT, the local policy is used.


However, the local policy only replaces the Request-URI when the original Request-URI matches the SBC's IP address or hostname. This behavior is in accordance with that described in RFC 3261. The original Request-URI will be the home proxy address value (the home address of the SIP NAT into the backbone) and not the Oracle Communications Session Border Controller (SBC) address.

Using strict routing, the Request-URI would be the next hop, but the message would also include a Route header with the original Request-URI. With loose routing, the Request-URI remains unchanged and the next hop value is added as the top Route header.

Sometimes the next hop field value must replace the Request-URI in the outgoing request, even if the original Request-URI is not the SBCC. To accomplish this, an option has been added to the local policy that causes the next hop value to be used as the Request-URI and prevents the addition of Route headers. This option is the replace uri value in the local policy.

The following table lists the policy attributes for the local policy:

Parameter	Description
next hop	IP address of your internal SIP proxy. This value corresponds to the IP address of the network interface associated with the SIP proxy.
realm	Number of the port associated with the SIP port.
replace uri	Stores the transport protocol used for sending an receiving signaling messages associated with the SIP port.
allow anonymous	Indicates whether this SIP port allows anonymous connections from session agents.

 **Note:** You should also define the ping method intervals for the session agents so that the SBC can detect when the agents are back in service after failure.

## SIP Server Redundancy Configuration

To enable replace URI:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **local-policy** and press Enter to access the system-level configuration elements. The system prompt changes.

```
ACMEPACKET(configure)# local-policy
ACMEPACKET(local-policy)#
```

3. Type **policy-attributes** and press Enter. The system prompt changes.

```
ACMEPACKET(local-policy)# policy-attributes
ACMEPACKET(local-policy-attributes)#
```

From this point, you can configure policy attributes for the local policy. To see all local policy attribute options, enter a **?** at the system prompt.

4. **action**—Set this parameter to **replace-uri**, which causes NAT rules such as realm prefixing to be overridden. The default value is **none**. Valid values are:

- none | replace-uri | redirect

The replace URI field allows you to indicate whether the local policy's value is used to replace the Request-URI in outgoing requests. This boolean field can be set to either enabled or disabled.

## Administratively Disabling a SIP Registrar

The Oracle Communications Session Border Controller's registration cache feature is commonly used to support authorization. It also allows the Oracle Communications Session Border Controller to respond directly to SIP REGISTER requests from endpoints rather than forwarding every REGISTER message to the Registrar(s). In the Oracle Communications Session Border Controller, Registrars are frequently configured as session agents, and an association between each endpoint and its Registrar is stored with the registration cache information.

In Release 4.0.1 and later, the **invalidate-registrations** parameter in the session agent configuration enables the Oracle Communications Session Border Controller to detect failed Registrar session agents and automatically forward subsequent REGISTER requests from endpoints to a new Registrar. You can now perform the same behavior manually through a new CLI command. When you use this command, the Oracle Communications Session Border Controller acts as though the registrations have expired.

For each SIP session agent, you can enable the manual trigger command, and then use the command from the main Superuser CLI prompt. The **reset session-agent** command provides a way for you to send a session agent

offline. Session agents can come back online once they send 200 OK messages the Oracle Communications Session Border Controller receives successfully.

Without using the manual trigger, session agents can go offline because of they do not respond to pings or because of excessive transaction timeouts. However, you might not want to use these more dynamic methods of taking session agents out of service (and subsequently invalidating any associated registrations). You can disable both of these mechanisms by setting the following parameters to 0:

- **ping-interval**—Frequency (amount of time in seconds) with which the Oracle Communications Session Border Controller pings the entity the session agent represents)
- **ttr-no-response**—Dictates when the SA (Session Agent) should be put back in service after the SA is taken OOS (Out Of Service) because it did not respond to the Oracle Communications Session Border Controller

However, you can still use the new SIP manual trigger even with these dynamic methods enabled; the trigger simply overrides the configuration to send the session agent offline.

### Considerations for Implicit Service Route Use

When implicit service route support is enabled for a SIP interface (in IMS applications), the Oracle Communications Session Border Controller stores the Service Route URIs from the Service-Route headers that are included in 200 OK responses to REGISTER messages. Subsequently, and even when a session agent is rendered invalid, re-REGISTER messages follow the route stored in the cache instead of using the one defined in the Oracle Communications Session Border Controller.

However, you might not want to use this behavior when you send session agents offline. If you instead want use the route defined in the Oracle Communications Session Border Controller, then you need to configure the SIP interface option called **route-register-no-service-route**.

### Manual Trigger Configuration

This section shows you how to enable the manual trigger for sending session agents out of service, and how to then use the trigger from the command line. This section also shows you how to verify that you have successfully put a session agent out of service.

To enable a SIP session agent to manually trigger it to go out of service:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # session-agent
ACMEPACKET (session-agent) #
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the CLI **select** command) the configuration you want to edit.

4. **invalidate-registrations**—Set this parameter to enabled if you want to use the manual trigger to send this session agent offline (and therefore invalidate the registrations associated with it). The default is disabled.
5. Save and activate your configuration.

To use the manual trigger that sends session agents offline:

6. Note the hostname value (typically the IP address of the endpoint) for the session agent you want to put out of service. You use this name as an argument in the CLI command to use the manual trigger.
7. At the Superuser prompt, type **reset session-agent**, a Space, and the hostname value for the session agent. The press Enter.



```
ACMEPACKET# reset session-agent 192.168.20.45
```

If you enter a session agent that does not exist, the system notifies you that it cannot carry out the reset.

## Manual Trigger Confirmation

To confirm that a session agent has been sent offline:

Use the **show sipd endpoint-ip** command to confirm the session agent state.

```
ACMEPACKET# show sipd endpoint-ip 1016
User <sip:1016@172.18.1.80>
  Contact exp=3582
    UA-Contact: <sip:1016@192.168.1.132:5060> UDP
      realm=access local=172.18.1.132:5060
UA=192.168.1.132:5060
  SD-Contact: <sip:1016-o3badgbbnjcq5@172.18.2.80:5060> realm=core
  Call-ID: 1-7944@192.168.1.132'
  SR=172.18.2.92
  SA=172.18.2.93
  Service-Route='<sip:test@s-cscf::5060;orig;lr>'
ACMEPACKET# reset session-agent 172.18.2.92
Accepted
Reset SA failover timer
ACMEPACKET# show sipd endpoint-ip 1016
User <sip:1016@172.18.1.80>
  Contact <invalidated> exp=3572
    UA-Contact: <sip:1016@192.168.1.132:5060> UDP
      realm=access local=172.18.1.80:5060
UA=192.168.1.132:5060
  SD-Contact: <sip:1016-o3badgbbnjcq5@172.18.2.80:5060> realm=core
  Call-ID: 1-7944@192.168.1.132'
  SR=172.18.2.92 (failed 2 seconds ago)
  SA=172.18.2.93
  Service-Route='<sip:test@s-cscf::5060;orig;lr>'
ACMEPACKET#
```

In the above ACLI example the first iteration of the **show sip endpoint-ip** command provides information for the in-service 172.18.2.92 session agent; the second command iteration displays information for the now out-of-service session agent.

## Surrogate Agent Refresh on Invalidate

Surrogate agent registrations normally only re-register when nearing their expiration time. When a registrar fails, the surrogate agent will wait until the expiration time to refresh the registration with an in-service registrar.

You can configure your Oracle Communications Session Border Controller to immediately refresh the surrogate agent registrar with an in-service registrar by enabling the existing parameter **invalidate-registrations**.

## Invalidate Registrations

An existing feature called **invalidate-registrations** located in the session agent keeps track of when surrogate agents go out of service. When REGISTER messages are received, registration entries that had out-of-service session agents since the last REGISTER will always allow the message through to the registrar (as opposed to responding directly from the cache).

The **invalidate-registrations** parameter in session agent configuration enables the Oracle Communications Session Border Controller to detect failed Registrar session agents.

If invalidate-registrations is enabled for the session agent, a response from a surrogate REGISTER that contains a service-route header that corresponds to a session-agent is installed to the registration cache entry.

## SIP Signaling Services

---

The surrogate-agents are scanned. Surrogate agents with registration entries matching the out-of-service registrar have their timer reset to initiate a refresh. For an immediate refresh, the registration entry will only be considered when the service-route session agent goes out-of-service. The service-route session agent takes precedence and any previous registrar session agent will not be considered for an immediate refresh of the surrogate-agent registration.

### Performance Impact

In cases with a large number of surrogate-agent registrations, there may be an impact to CPU usage when a session-agent goes out-of-service. All of the surrogate-agent registrations are scanned at that time. Refresh registrations are then sent out on timers.

## Media Inactivity Timer Configuration

To disable the media inactivity timer for calls placed on hold:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET (session-router) # session-agent
ACMEPACKET (session-agent) #
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the ACLI **select** command) the configuration you want to edit.

4. **invalidate-registration**—Set this parameter to **enabled** if you want to use the manual trigger to send this session agent offline (and therefore invalidate the registrations associated with it). The default is **disabled**.
5. Save and activate your configuration.

## SIP Distributed Media Release

---

This section explains how to configure distributed media release (DMR). SIP DMR lets you choose whether to include multi-system (multiple Oracle Communications Session Border Controllers) media release information in SIP signaling requests sent into a specific realm.

### Overview

The SIP DMR feature lets RTP/RTCP media be sent directly between SIP endpoints (for example, SIP phones or user agents) without going through a Oracle Communications Session Border Controller; even if the SIP signaling messages traverse multiple Oracle Communications Session Border Controllers. It encodes IPv4 address and port information for the media streams described by the media, for example SDP.

With SIP DMR, the media realm and IPv4 address and port information from the UA's SDP is encoded into SIP messages (either in the SIP header or in the SDP) as they enter the backbone network. The information is decoded by a Oracle Communications Session Border Controller from SIP messages that come from the backbone network. The decoded address and port information is put into the SDP sent the UAs in the access (private/customer) network.

This functionality lets the RTP/RTCP flow directly between the UAs in the access network without traversing the Oracle Communications Session Border Controllers and without passing into the backbone network. The media can then flow directly between the two SIP endpoints in the same network, if it is serviced by multiple Oracle Communications Session Border Controllers.

You can enable this feature on a per-realm basis and multiple realms can be supported.

## Endpoint Locations

You can configure the Oracle Communications Session Border Controller to release media when the source and destination of the call are in the same network, customer VPN, or customer LAN. In architectures that use DMR, the Oracle Communications Session Border Controller is only part of the media path for traffic that originates and terminates in different networks.

If configured to do so, the Oracle Communications Session Border Controller can release media:

- Between endpoints supported by a single Oracle Communications Session Border Controller
  - In the same network/VPN
  - In the same network behind the same NAT/firewall
- Between endpoints supported by multiple distributed Oracle Communications Session Border Controllers
  - In the same network/VPN

## Location of the Encoded Information

Encoded media release information can appear in three different places:

- SDP attribute

Media release data can be encoded into an SDP attribute in the SIP message body (for example, `media-release=sdp;acme-media`). The encoded data is placed into an `acme-media` attribute in the SDP:

```
a=acme-media:<encoded-media-interface-info>
```

- SIP header parameter

Media release data can be placed in a header parameter of a SIP header (for example, `media-release=Contact;acme-media`). The encoded data is placed into an `acme-media` parameter in the Contact header:

```
Contact: <sip:1234@abc.com>;acme-media=<encoded-media-interface-info>
```

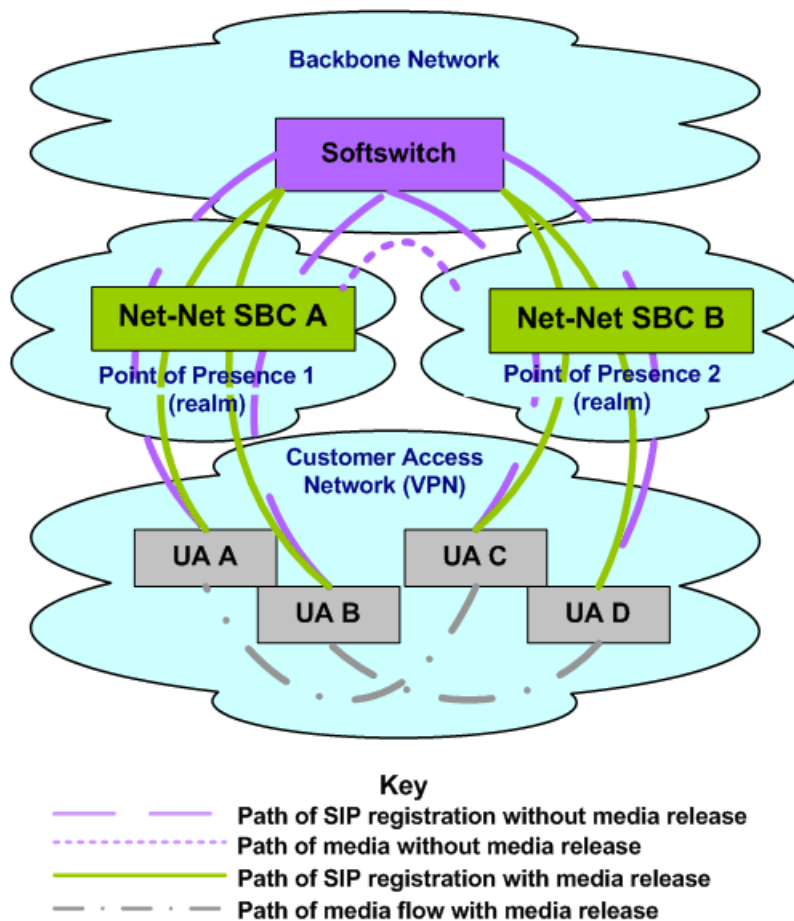
- SIP header

Media release data can appear in a SIP header (for example, `media-release=P-Media-Release`). The encoded data is placed into a P-Media-Release header:

```
P-Media-Release: <encoded-media-interface-info>
```

## Example Distributed Media Release

The following example shows the network diagram for DMR in a multiple-site VPN environment supported by multiple, distributed Oracle Communications Session Border Controllers.



As shown in the network diagram, UA A and UA B register with the softswitch through Oracle Communications Session Border Controller A while UA C and UA D register with the softswitch through Oracle Communications Session Border Controller B. Without DMR, the media for calls between UA A/UA B and UA C/UA D is steered through both Oracle Communications Session Border Controller A and Oracle Communications Session Border Controller B.

With SIP DMR, the media realm and IPv4 address and port information from the UA's Session Description Protocol (SDP) is encoded into SIP messages (either in the SIP header or in the SDP) as they enter the backbone (public/service provider) network. The information is decoded from SIP messages that come from the backbone network. The decoded address and port information is put into the SDP sent to the UAs in the access (private/customer) network. This functionality allows for the RTP/RTCP to flow directly between the UAs in the access network without traversing the Oracle Communications Session Border Controllers and without passing into the backbone network.

## Overview of SIP DMR Configuration

To configure SIP DMR:

1. Edit the SIP config element's option field.

The `media-release="<header-name>[;<header-param>]"` option defines how the SIP distributed media release feature encodes IPv4 address and port information. If the media-release parameter is configured in the options field but no header is specified, the parameter value of `P=Media-Release` will be used. This parameter is optional and is not configured by default.

2. Enable SIP DMR for the entire realm by setting the realm config element's `msm release` field to enabled.

The media IPv4 address and port information is encoded into outgoing SIP messages and decoded from incoming SIP messages for all of the realms (in each realm-config element) with which the SIP distributed media release will be used.



**Note:** You can also use the realm config element's `mm in network` field to release the media back to a connected network that has multiple realms. This field is not specific SIP distributed media release and it is not required for the SIP DMR to work. However, if this field is set to enabled and the ingress and egress realms are part of the same network interface, it lets the Oracle Communications Session Border Controller release the media.

## SIP DMR Configuration

To configure media release:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type `sip-config` and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

From this point, you can configure SIP config parameters. To view all SIP config parameters, enter a `?` at the system prompt.

4. Type `options` followed by a Space.
5. After the Space, type the media release information in the following format:

```
media-release="<header-name> [ ; <header-param> ] "
```

- `header-name` either refers to the SIP header in which to put the information or to the special `header-name` value of `sdp` to indicate the information should be put into the SDP.
- `parameter-name` refers to the header parameter name in which to put the information or, in the case of the special `header-name` value of `sdp`, to the SDP attribute name in which to put the information.

For example:

```
ACMEPACKET(sip-config)# options media-release=P-Media-Release
```

6. Press Enter.



**Note:** If the `media-release` parameter is configured in the options field, but no header is specified, then the parameter value of `P-Media-Release` will be used. `P-Media-Release` is a proprietary header and means that the media will be encoded in the SIP header with this name.

The following example shows where the encoded information (for example, SDP data) is passed.

```
media-release="P-Media-Release"
media-release="Contact;acme-media"
media-release="sdp;acme-media"
```

## Configuring the Realm

You need to set the each realm config element's `msm release` field to enabled for all the realms for which you want to use SIP DMR.

Although the `mm in network` field is not specific to the SIP distributed media release feature, it can be used to release the media back to a connected network that has multiple realms. This field does not need to be configured in order for the SIP distributed media release feature to work. However, if this field is set to enabled and the ingress and egress realms are part of the same network interface, it lets the Oracle Communications Session Border Controller release the media.

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `media-manager` and press Enter to access the media-related configurations.

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---

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config  
ACMEPACKET (realm-config) #
```

From this point, you can configure realm parameters. To view all realm configuration parameters, enter a ? at the system prompt.

4. **msm-release**—Enable DMR within this realm on this Oracle Communications Session Border Controller. The default value is **disabled**. The valid values are:
  - enabled | disabled
5. Repeat for each realm on which you want to enable DMR.

## Add-On Conferencing

---

This section explains how to configure the add-on conferencing functionality. It also includes a description of the SIP B2BUA functionality related to the SIP add-on conferencing. This description includes information about Contact header mapping and processing and Refer-to header processing.

### Overview

SIP add-on conferencing lets you:

- Use the Oracle Communications Session Border Controller's add-on conferencing feature for network architectures in which the conference initiator is located on a different network than that of the media server.
- Configure the Oracle Communications Session Border Controller to enable Contact header mapping for the Refer-To header.

### Caveats

The following caveats are associated with add-on conferencing:

- Contact header mapping is not replicated on the standby Oracle Communications Session Border Controller in an HA Oracle Communications Session Border Controller pair architecture.
- Upon switchover, any conferences in progress remain in progress, but no new parties can be invited to or join the conference.
- By default, the Oracle Communications Session Border Controller does not map SIP Contact headers for reasons of performance.

### Add-On Conferencing Scenario

The add-on conferencing scenario described in the following example applies to a network architecture involving the Oracle Communications Session Border Controller and a media server that is located on a different network from the other conference participants. In this scenario, the Oracle Communications Session Border Controller resides on a standalone network that connects two additional, separate networks.

Some network architectures have a media server on a different network from the one on which the phones reside. In this scenario, all requests and/or responses going from the phones (Phone A, Phone B, or Phone C) to Media Server D and vice versa are translated according to their corresponding SIP-NAT. All headers subjected to NAT are encoded and decoded properly as they traverse the Oracle Communications Session Border Controller, except for the Contact header. This exception occurs because the SIP process on the Oracle Communications Session Border Controller runs as a SIP B2BUA and not as a SIP proxy.

The SIP B2BUA re-originates the Contact headers of the User Agents (UAs) participating in SIP sessions with local Contact headers to make sure that they receive all future in-dialog requests. For an in-dialog request, the B2BUA can identify the dialog and find the Contact URI of the other leg of the call.

The Oracle Communications Session Border Controller add-on conferencing feature applies to situations when the Contact URI is used in another dialog. In such a case, the SIP B2BUA will not be able to find the correct dialog that retrieves the correct Contact URI of the other leg if it needs to replace the Contact URI.

Using the SIP add-on conferencing, the SIP B2BUA on the Oracle Communications Session Border Controller can map the Contact headers it receives to the Contact headers it creates. It can also convert the Refer-To URI to the correct value required for forwarding the REFER request.

### **SIP B2BUA Functionality**

This section describes the role of the Oracle Communications Session Border Controller's SIP B2BUA in the add-on conferencing scenario that requires Contact header mapping for the Refer-To header.

When the Oracle Communications Session Border Controller starts up, the SIP B2BUA reads and parses the list of options in the SIP configuration. If the refer to uri prefix is an appropriate value (it is not an empty string), the Oracle Communications Session Border Controller will have a text prefix value the media server can use to denote a conference ID in its Contact header. With this information, the SIP B2BUA sets up a Contact header mapping.

You configure the Oracle Communications Session Border Controller to enable Contact header mapping for the Refer-To header by editing the SIP config options parameter. The SIP B2BUA on the Oracle Communications Session Border Controller can then map the Contact headers it receives to the Contact headers it creates.

### **Contact Header Processing**

The Contact header mapping matches a Contact header that contains the refer to URI prefix to the corresponding Contact header that the Oracle Communications Session Border Controller's SIP B2BUA re-originates. Contact headers that do not contain the refer to URI prefix are not mapped (so that performance of the Oracle Communications Session Border Controller is minimally affected).

Only the Contact header in an INVITE request and its 200 OK response are checked for the refer to URI prefix and added to the Contact header mapping. Contact headers appearing in other SIP requests/responses are not checked.

### **Target Mapping and Conferences**

If the Oracle Communications Session Border Controller is configured to enable Contact header mapping for the Refer-To header, then Contact header target maps are established for each individual call. The Oracle Communications Session Border Controller's SIP B2BUA uses these maps to allow the media server to connect the conference initiator with the conferenced-in parties.

Prior to terminating the call (hanging up), the conference initiator can contact other parties and invite those additional parties to join the conference. These other parties can join the existing conference because the target mapping for the conference is still in effect on the Oracle Communications Session Border Controller.

Once the conference initiator hangs up, the Oracle Communications Session Border Controller discards the mapping from the conference.

### **Refer-To Header Processing**

When a Refer-To header is present in a REFER request that arrives at the SIP B2BUA after the incoming request is properly translated according to its SIP-NAT, the SIP B2BUA follows these steps:

1. The SIP B2BUA parses the Refer-To URI.
2. If the user part of the Refer-To URI contains the refer to URI prefix, the SIP B2BUA searches the Contact header mapping for a match of the user part of the URI.

If the user part of the Refer-To URI does not contain the refer to URI prefix, the SIP B2BUA leaves the existing Refer-To URI unchanged.

3. If the user part of the Refer-To URI contains the refer to URI prefix and a match of the Refer-To URI is found, the SIP B2BUA replaces the existing Refer-To URI with the URI of the corresponding Contact URI stored in the matched record. This replacement enables the NAT function to properly decode the replacement URI and change it back to the form originally received by the Oracle Communications Session Border Controller. As a result, the correct conference ID is restored in the Refer-To header prior to the request being sent to its next hop.

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If the user part of the Refer-To URI contains the refer to URI prefix but a matched URI cannot be found, the SIP B2BUA will leave the existing Refer-To URI unchanged and will write a WARNING level log message to record the failure.

### Add-on Conferencing Configuration

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-config  
ACMEPACKET(sip-config)#
```

From this point, you can configure SIP config parameters. To view all SIP config parameters, enter a ? at the system prompt.

4. Type **options** followed by a Space.

5. After the Space, type the add-on conferencing information in the following format:

```
options refer-to-uri-prefix="conf"
```

For example:

```
ACMEPACKET(sip-config)# options refer-to-uri-prefix="conf"
```

6. Press Enter.

## SIP REFER Method Call Transfer

---

In prior releases, the Oracle Communications Session Border Controller supports the SIP REFER method by proxying it to the other UA in the dialog. A handling mode has been developed for the REFER method so that the Oracle Communications Session Border Controller automatically converts a received REFER method into an INVITE method, thus allowing the Oracle Communications Session Border Controller to transfer a call without having to proxy the REFER back to the other UA.

This function can be configured for a specified SIP interface, a realm, or a session agent. When all three elements have the SIP REFER method call transfer functionality configured, the session-agent configuration takes precedence over realm-config and sip-interface configurations. If session-agent is not configured, and realm-config and sip-interface are, realm-config takes precedence.

The Oracle Communications Session Border Controller has a configuration parameter giving it the ability to provision the handling of REFER methods as call transfers. The parameter is called refer-call-transfer. When this feature is enabled, the Oracle Communications Session Border Controller creates an INVITE message whenever it receives a REFER. The Oracle Communications Session Border Controller sends this INVITE message to the address in the Refer-To header. Included in the INVITE message is all the unmodified information contained in the REFER message. The previously negotiated codec is also still used in the new INVITE message. NOTIFY and BYE messages are sent to the UA upon call transfer completion.

If a REFER method is received containing no Referred-By header, the Oracle Communications Session Border Controller adds one, allowing the Oracle Communications Session Border Controller to support all call agent screen applications.

In addition, the SIP REFER method call transfer feature supports the following:

- Both unattended and attended call transfers
- Both successful and unsuccessful call transfers
- Early media from the Referred-To party to the transferee



- REFER method transfer from different sources within the destination realm
- The REFER event package as defined in RFC 3515. This applies for situations where multiple REFER methods are used within a single dialog.
- Third party initiated REFER method signalling the transfer of a call by associating the REFER method to the dialogue via the REFER TargetDialog.
- The Referred-To party can be both in a different realm (and thus a different steering pool) from the referrer, and in the same realm
- The associated latching should not prohibit the Referred-To party from being latched to while the referee is still sending media.

## Unsuccessful Transfer Scenarios

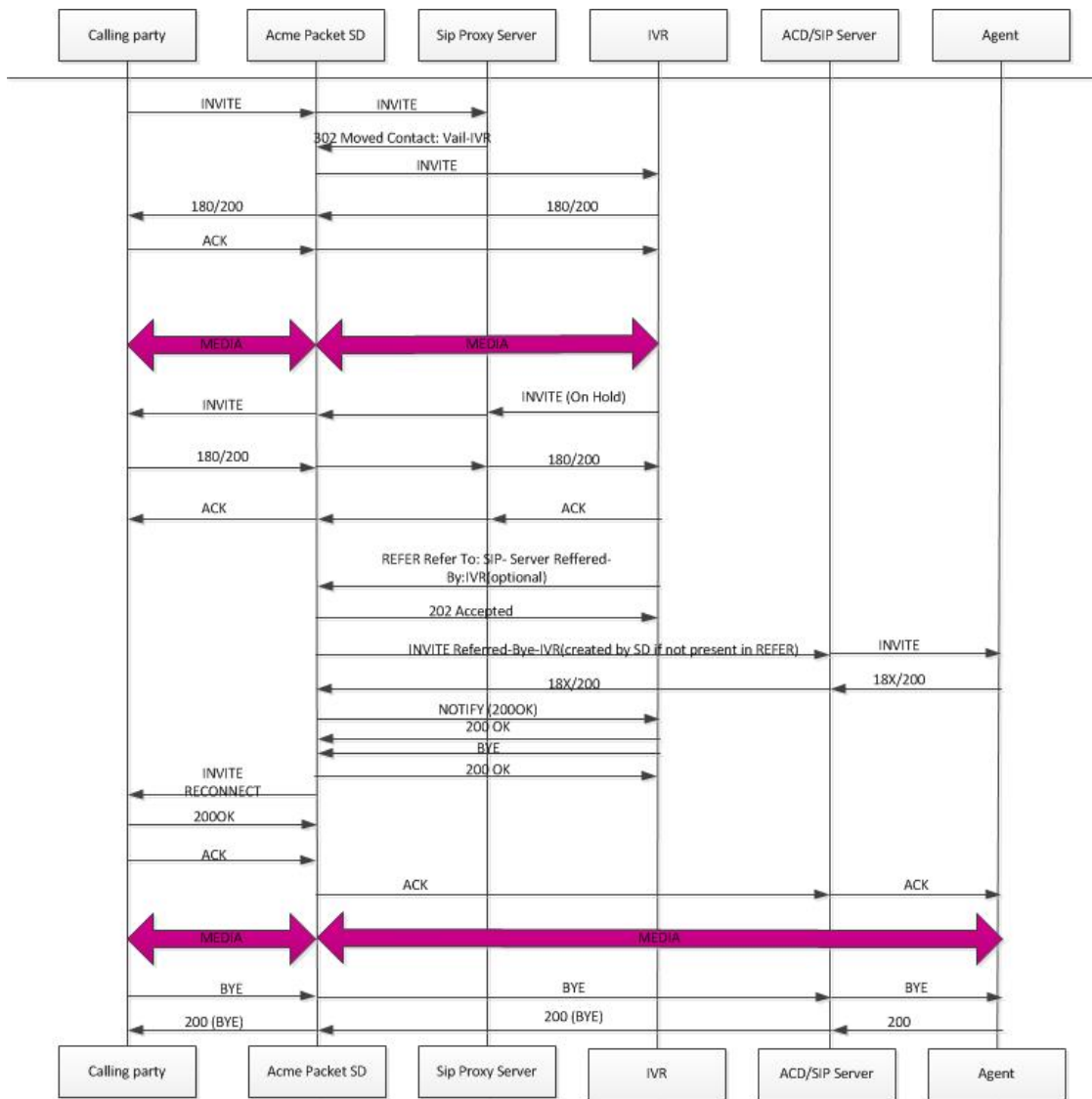
The Oracle Communications Session Border Controller does not successfully handle the following failed, unusual, and unexpected transfer scenarios:

- The new INVITE to the Referred-To party gets challenged, the Oracle Communications Session Border Controller does not answer the challenge. It is treated with the 401/407 response just as any other unsuccessful final response.
- The header of the REFER message contains a method other than INVITE or contains URI-parameters or embedded headers not supported by the Oracle Communications Session Border Controller.
- The Oracle Communications Session Border Controller shall allow the Referred-To URI that happens to resolve to the same next-hop as the original INVITE went to, to do so.
- The Oracle Communications Session Border Controller ignores any MIME attachment(s) within a REFER method.
- The Oracle Communications Session Border Controller recurses (when configured to do so) when the new INVITE sent to the Referred-To party receives a 3xx response.
- The transferee indicated support for 100rel, and the original two parties agreed on using it, yet the Referred-To party does not support it.
- The original parties negotiated SRTP keys.
- The original parties agreed on a codec using a dynamic payload type, and the Referred-To party happens to use a different dynamic payload number for that codec.

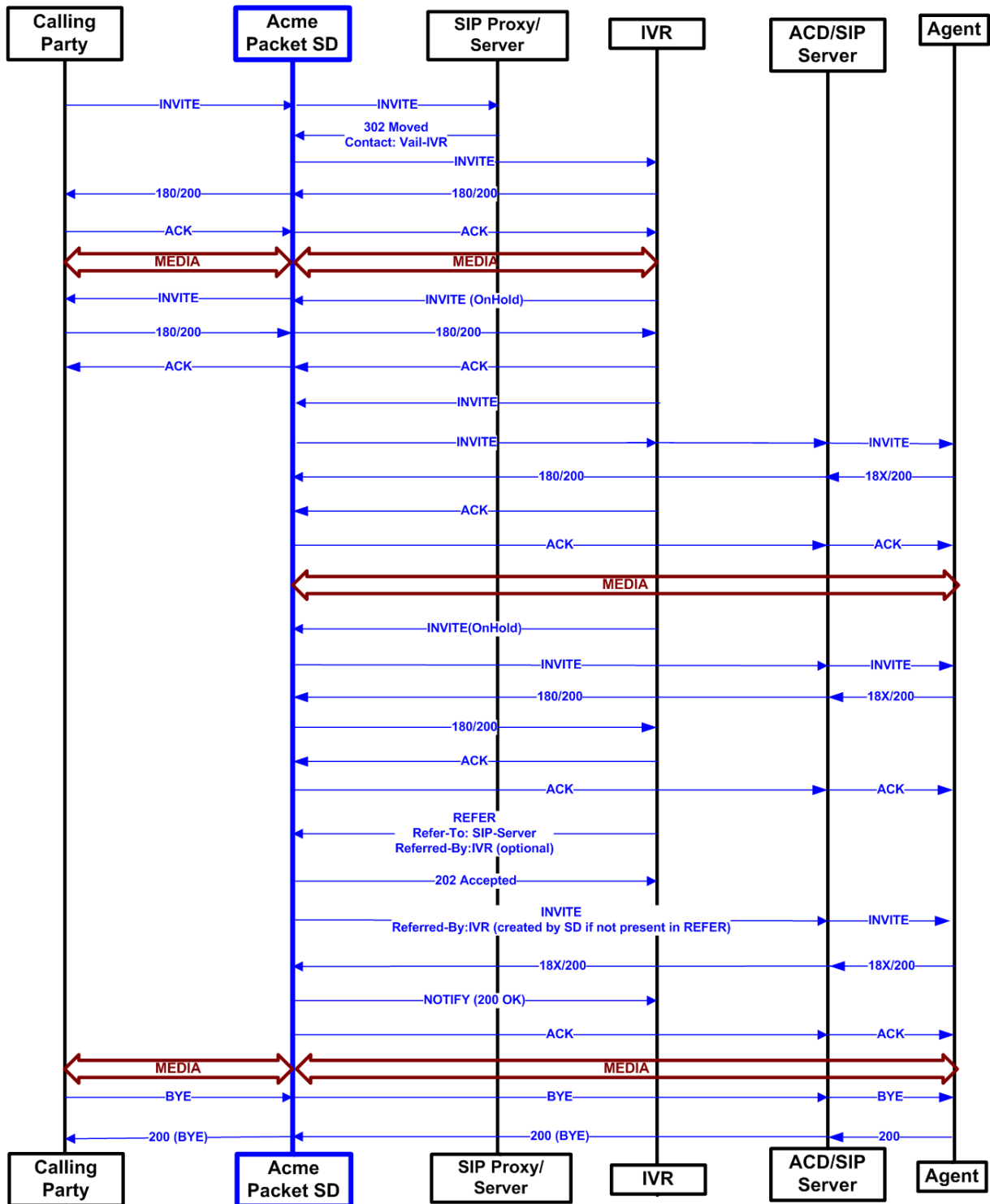
## Call Flows

The following is an example call flow for an unattended call transfer:

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The following is an example call flow of an attended call transfer:



## SIP REFER Method Configuration

To enable SIP REFER method call transfer in the realm-config:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type `media-manager` and press Enter.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

4. **refer-call-transfer**—Set to **enabled** to enable the refer call transfer feature. The default for this parameter is **disabled**.
5. Save and activate your configuration.

To enable SIP REFER method call transfer in the sip-interface:

6. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

7. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

8. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-config) #
```

9. **refer-call-transfer**—Set to **enabled** to enable the refer call transfer feature. The default for this parameter is **disabled**.
10. Save and activate your configuration.

To enable SIP REFER method call transfer in a realm:

11. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

12. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

13. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

14. **refer-call-transfer**—Set to **enabled** to enable the refer call transfer feature. The default for this parameter is **disabled**.
15. Save and activate your configuration.

To enable SIP REFER method call transfer in the session-agent:

16. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

17. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

18. Type **session-agent** and press Enter.

```
ACMEPACKET (media-manager) # session-agent
ACMEPACKET (session-agent) #
```


19. **refer-call-transfer**—Set to **enabled** to enable the refer call transfer feature. The default for this parameter is **disabled**.

20. Save and activate your configuration.

## REFER-Initiated Call Transfer

In prior releases, the Oracle Communications Session Border Controller supported REFER-initiated call transfer either by proxying the REFER to the other User Agent in the dialog, or by terminating the received REFER and issuing a new INVITE to the referred party. These static alternate operational modes could be configured for specific SIP interfaces, realms, or session agents.

Release S-C6.2.0 enhances support with an additional operational mode that determines on a call-by-call basis whether to proxy the REFER to the next hop, or terminate the REFER and issue an INVITE in its stead.

 **Note:** With the release of Version S-C6.2.0, support for REFER-initiated call transfer is no longer available for SIP interfaces; support must be configured for realms and/or session agents.

Version S-C6.2.0 provides a new configuration parameter **dyn-refer-term**, and a revised **refer-call-transfer** parameter (both available in realm-config configuration mode) that specify call transfer modes.

With the **refer-call-transfer** parameter set to **disabled** (the default), all received REFERs are simply proxied to the peer User Agent.

With the **refer-call-transfer** parameter set to **enabled**, the Oracle Communications Session Border Controller terminates all REFERs, generates a new INVITE, and sends the INVITE to the address in the Refer-To header.

With the **refer-call-transfer** parameter set to **dynamic** (a new value introduced with Version S-C6.2.0), the Oracle Communications Session Border Controller determines REFER handling on a call-by-call basis as follows:

1. Check the **refer-call-transfer** value for the session agent from which the REFER was received, or for ingress realm (the realm that received the REFER).

If the value is disabled, proxy the REFER to the peer User Agent, to complete REFER processing.

If the value is enabled, terminate the REFER and issue a new INVITE to the referred party, to complete REFER processing.

If the value is dynamic, identify the next hop session agent or the egress realm.

2. Check the **dyn-refer-term** value for the next hop session agent, or for the egress realm.

If the **dyn-refer-term** value is disabled (the default), proxy the REFER to the next hop to complete REFER processing.

If the **dyn-refer-term** value is enabled, terminate the REFER and issue a new INVITE to the referred party to complete REFER processing

## Supported Scenarios

In the basic scenario for REFER initiated call transfer, a call is established between two User Agents (Alice and Bob). User Agent Bob then sends a REFER request to transfer the call to a third User Agent Eva. With dynamic call-transfer enabled, the Oracle Communications Session Border Controller (SBC) prevents the REFER from being sent to Alice and generates the INVITE to Eva.

If the INVITE to Eva succeeds, the SBC sends a re-INVITE to Alice modifying the SIP session as described in Section 14 of RFC 3261, *SIP: Session Initiation Protocol*. At this point the SBC cancels the original dialog between the SBC and Bob.

If the INVITE to Eva fails, call disposition depends on whether or not Bob issued a BYE after the REFER call transfer. If the Oracle Communications Session Border Controller did receive a BYE from Bob (for instance, a blind transfer), it proxies the BYE to A. Otherwise, the SBC retains the original SIP session and media session, thus allowing Bob to re-establish the call with Alice by sending a re-INVITE. In this case, the SBC sets a timer (32 seconds), after which a BYE will be sent.

## SIP Signaling Services

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If a REFER method is received containing no Referred-By header, the SBC adds one, allowing the SBC to support all call agent screen applications.

In addition, the SIP REFER method call transfer feature supports the following:

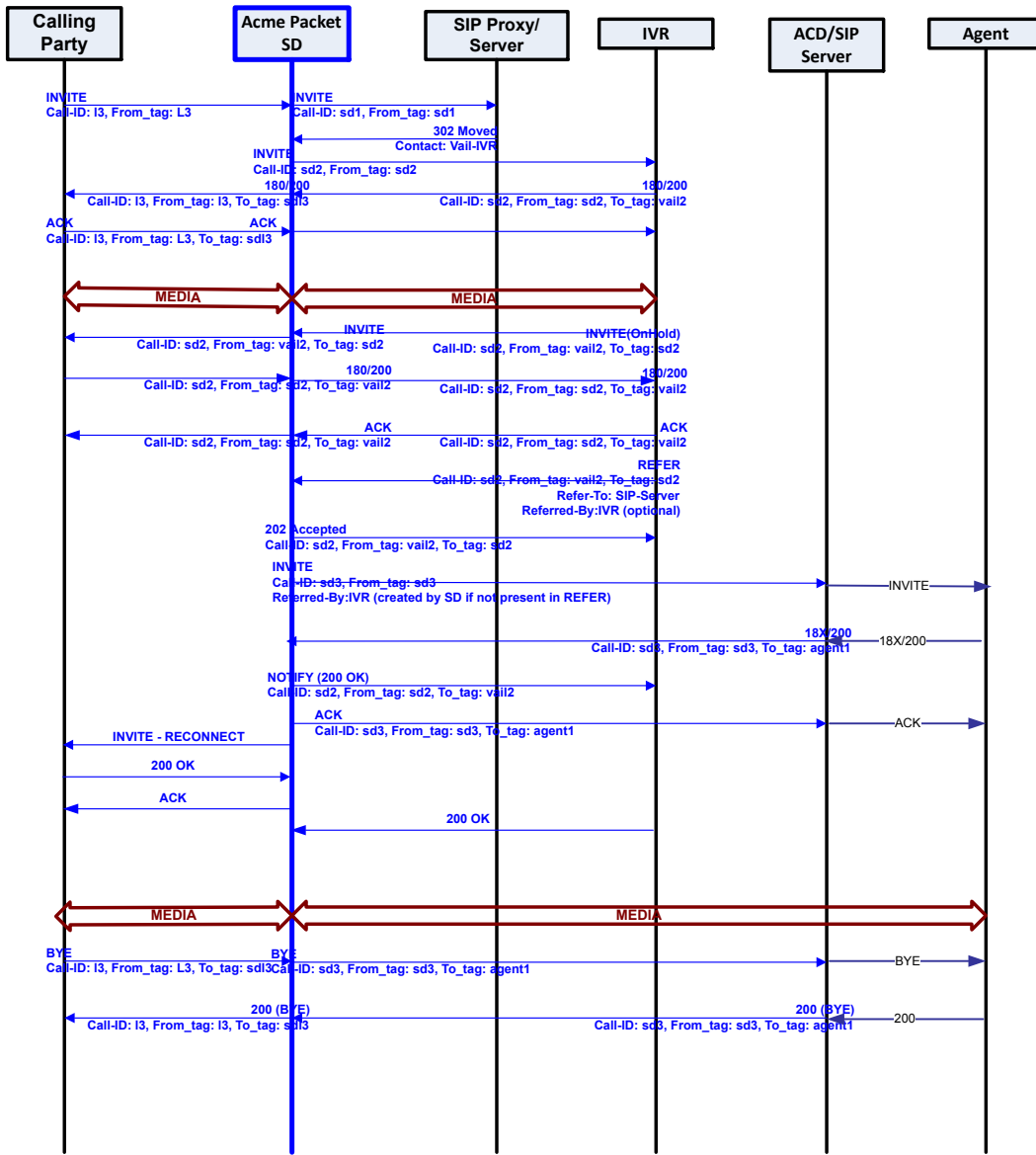
- Both unattended and attended call transfers
- Both successful and unsuccessful call transfers
- Early media from the Referred-To party to the transferee
- REFER method transfer from different sources within the destination realm
- The REFER event package as defined in RFC 3515. This applies for situations where multiple REFER methods are used within a single dialog.
- Third party initiated REFER method signalling the transfer of a call by associating the REFER method to the dialogue via the REFER TargetDialog.
- The Referred-To party can be both in a different realm (and thus a different steering pool) from the referrer, and in the same realm
- The associated latching should not prohibit the Referred-To party from being latched to while the referee is still sending media.

The SBC does not successfully handle the following anomalous transfer scenarios:

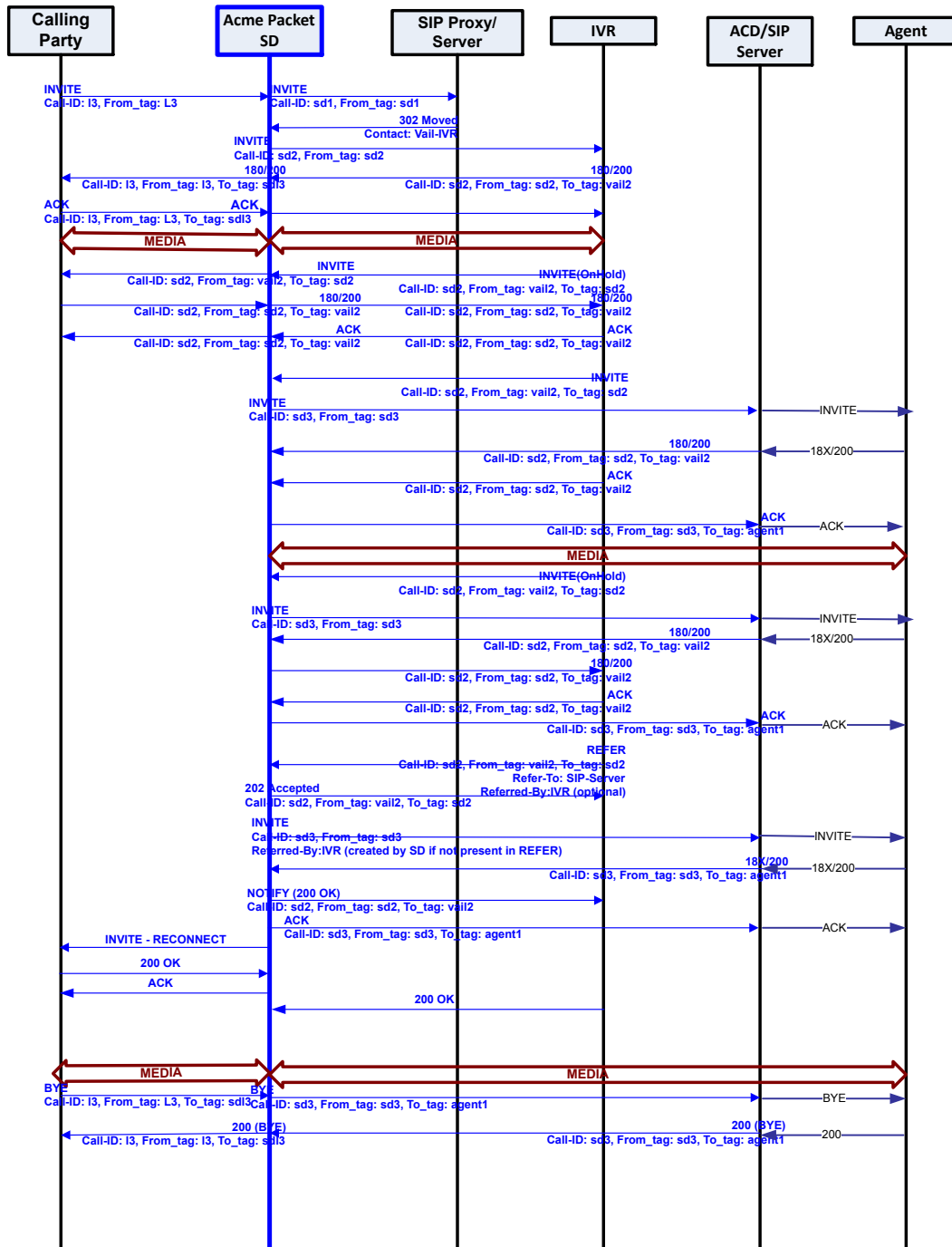
- The new INVITE to the Referred-To party gets challenged — the SBC does not answer the challenge. It is treated with the 401/407 response just as any other unsuccessful final response.
- The header of the REFER message contains a method other than INVITE or contains URI-parameters or embedded headers not supported by the SBC.
- The SBC shall allow the Referred-To URI that happens to resolve to the same next-hop as the original INVITE went to, to do so.
- The SBC ignores any MIME attachment(s) within a REFER method.
- The SBC recurses (when configured to do so) when the new INVITE sent to the Referred-To party receives a 3xx response.
- The transferee indicated support for 100rel, and the original two parties agreed on using it, yet the Referred-To party does not support it.
- The original parties negotiated SRTP keys.

### Call Flows

The following is an example call flow for an unattended call transfer:



The following is an example call flow of an attended call transfer:



## REFER Source Routing

If, after the conclusion of static or dynamic REFER handling, the REFER is terminated and a new INVITE issued, users now can specify a policy lookup behavior based upon either the source realm of the calling party (the INVITE originator), or the source realm of the referring party (the REFER originator).

Behavior is controlled by a new **refer-src-routing** parameter in the **sip-config** configuration element.

**disabled**, the default value, specifies that the Oracle Communications Session Border Controller performs a policy lookup based on the source realm of the calling party.

**enabled** specifies that the Oracle Communications Session Border Controller performs a policy lookup based on the source realm of the referring party.



## REFER Source Routing Configuration

To enable realm-based REFER method call transfer:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **realm-config** and press Enter.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. **refer-call-transfer** — Retain the default (**disabled**) to proxy all REFERs to the next hop. Use **enabled** to terminate all REFERs and issue a new INVITE. Use **dynamic** to specify REFER handling on a call-by-call basis, as determined by the value of the **dyn-refer-term** parameter.
5. **dyn-refer-term** (meaningful only when **refer-call-transfer** is set to dynamic) — Retain the default (**disabled**) to terminate the REFER and issue a new INVITE. Use **enabled** to proxy the REFER to the next hop.
6. Save and activate your configuration.

To enable session-agent-based REFER method call transfer:

7. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

8. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

9. Type **session-agent** and press Enter.

```
ACMEPACKET(media-manager)# session-agent
ACMEPACKET(session-agent)#
```

10. **refer-call-transfer** — Retain the default (**disabled**) to proxy all REFERs to the next hop. Use **enabled** to terminate all REFERs and issue a new INVITE. Use **dynamic** to specify REFER handling on a call-by-call basis, as determined by the value of the **dyn-refer-term** parameter.
11. Save and activate your configuration.

To specify policy lookup for a newly generated INVITE:

12. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

13. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

14. Type **sip-config** and press Enter.

```
ACMEPACKET(configure)# sip-config
ACMEPACKET(sip-config)#
```

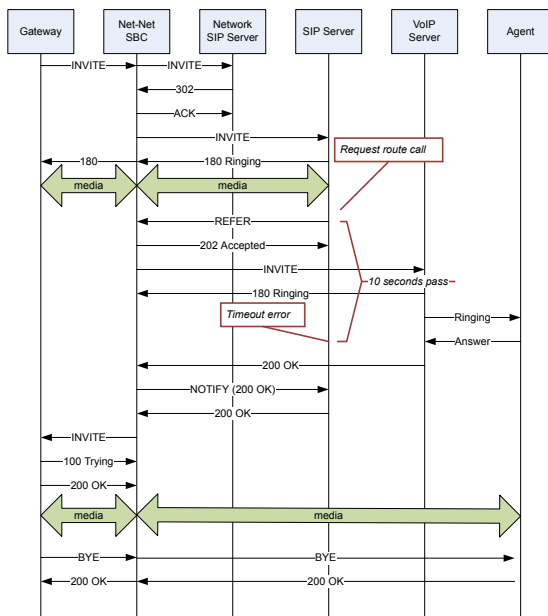
15. **refer-src-routing** — Retain the default (**disabled**) to perform a policy lookup based upon the source realm of the calling party (the issuer of the original INVITE). Use **enabled** to perform a policy lookup based upon the source realm of the referring party (the issuer of the REFER).
16. Save and activate your configuration.

## 180 & 100 NOTIFY in REFER Call Transfers

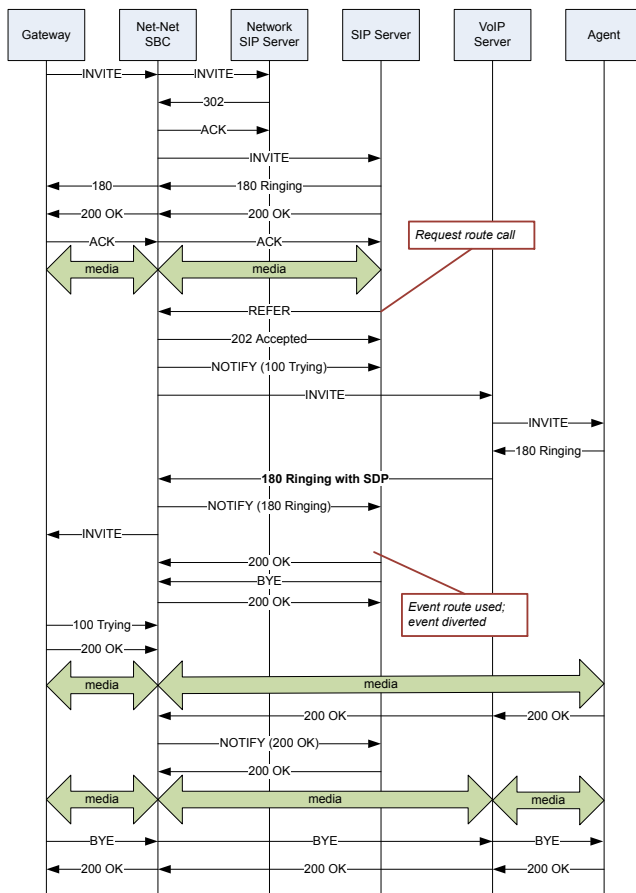
When you configure your Oracle Communications Session Border Controller to support REFER call transfers, you can enable it to send a NOTIFY message after it has sent either a 202 Accepted or sent a 180 Ringing message. If your network contains elements that comply with RFC 5589, and so expect the NOTIFY message after the 202 Accepted and each provisional 180 Ringing, you want to set the **refer-notify-provisional** to either **initial** or **all**, according to your needs.

Without this parameter changed from its default (**none**), the Oracle Communications Session Border Controller does not return send the NOTIFY until it receives the 200 OK response from the agent being called. If the time between the REFER and the NOTIFY exceeds time limits, this sequencing can cause the Oracle Communications Session Border Controller’s NOTIFY to go undetected by devices compliant with RFC 5589. Failures during the routing process can result.

You can see how a sample call flow works without setting the **refer-notify-provisional** parameter.



When you compare the call flow above to the one depicting the scenario when the Oracle Communications Session Border Controller has the **refer-notify-provisional** changed from its default, you can see that the Oracle Communications Session Border Controller now response with a NOTIFY in response to the 202 Accepted and it sends another after the 180 Ringing. This causes the event to be diverted successfully.



## Sample Messages

In compliance with RFC 5589, the NOTIFY message with 100 Trying as the message body looks like the sample below. Note that the expires value in the subscription state header is populated with a value that equals 2\* TIMER C, where the default value of TIMER C is 180000 milliseconds.

```
NOTIFY sips:4889445d8kjdk3@atlanta.example.com;gr=723jd2d SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>;tag=1928301774
From: <sips:3ld812adjkw@biloxi.example.com;gr=3413kj2ha>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 73 NOTIFY
Contact: <sips:3ld812adjkw@biloxi.example.com;gr=3413kj2ha>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, tdialog
Event: refer
Subscription-State: active;expires=360
Content-Type: message/sipfrag
```

```
Content-Length: ...
SIP/2.0 100 Trying
```

Also in compliance with RFC 5589, the NOTIFY message with 180 Ringing as the message body looks like the sample below. Again, the expires value in the subscription state header is populated with a value that equals 2\* TIMER C, where the default value of TIMER C is 180000 milliseconds.

```
NOTIFY sips:4889445d8kjdk3@atlanta.example.com;gr=723jd2d SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>;tag=1928301774
From: <sips:3ld812adkjwt@biloxi.example.com;gr=3413kj2ha>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 73 NOTIFY
Contact: <sips:3ld812adkjwt@biloxi.example.com;gr=3413kj2ha>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, tdialog
Event: refer
Subscription-State: active;expires=360
Content-Type: message/sipfrag
Content-Length: ...
SIP/2.0 180 Ringing
```

Also in compliance with RFC 5589, the NOTIFY message with 200 OK as the message body looks like the sample below.

```
NOTIFY sips:4889445d8kjdk3@atlanta.example.com;gr=723jd2d SIP/2.0
Via: SIP/2.0/TLS 192.0.2.4;branch=z9hG4bKnas432
Max-Forwards: 70
To: <sips:transferor@atlanta.example.com>;tag=1928301774
From: <sips:3ld812adkjwt@biloxi.example.com;gr=3413kj2ha>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 74 NOTIFY
Contact: <sips:3ld812adkjwt@biloxi.example.com;gr=3413kj2ha>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, tdialog
Event: refer
Subscription-State: terminated;reason=noresource
Content-Type: message/sipfrag
Content-Length: ...
SIP/2.0 200 OK
```

## 180 and 100 NOTIFY Configuration

You can apply the **refer-notify-provisional** setting to realms or to session agents. This section shows you how to apply the setting for a realm; the same steps and definitions apply to session agents.

If you do not want to insert NOTIFY messages into the exchanges that support REFER call transfers, you can leave the **refer-notify-provisional** set to **none**. This means that the Oracle Communications Session Border Controller will send only the final result NOTIFY message. Otherwise, you want to choose one of the two settings described in the instructions below.

To enable 100 and 180 NOTIFY messages in REFER call transfers:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET(configure)# media-manager
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. **refer-notify-provisional**—Choose from one of the following settings, where the Oracle Communications Session Border Controller:
  - **initial**—Sends an immediate 100 Trying NOTIFY, and the final result NOTIFY
  - **all**—Sends an immediate 100 Trying NOTIFY, plus a notify for each non-100 provisional messages the Oracle Communications Session Border Controller receives; and the final result NOTIFY

```
ACMEPACKET (realm-config) # refer-notify-provisional all
```
5. Save your work.

## SIP REFER Re-Invite for Call Leg SDP Renegotiation

Enhancing the original implementation of SIP REFER termination introduced in Release S-C6.0.0, this change to Oracle Communications Session Border Controller behavior allows for SDP renegotiation between both parties of a transferred call.

### Scenario

In a call transfer initiated by SIP REFER, a call is established between two user agents, UA-A and UA-B. UA-B then sends a REFER request to transfer the call to UA-C. The challenge is that UA-A and UA-B had already been communicating using mutually agreed-on codec, while UA-C might not be using an entirely different codec.

To solve this problem, the Oracle Communications Session Border Controller causes a new SIP session and new media session to be created between UA-A and UA-C. The Oracle Communications Session Border Controller removes any resources allocated for use between UA-A and UA-B, and then severs its connection with UA-B. The session between UA-A and UA-C continues.

## Alterations to SIP REFER

The original implementation of the SIP REFER feature made available in Oracle Communications Session Border Controller Release S-C6.0.0 resulted in instances where SDP parameters were not being communicate properly. Issues arose when the Oracle Communications Session Border Controller maintained the original dialog with the user agent that did not support REFER and failed to communicate and SDP changes to that endpoint.

The alterations to SIP REFER made available in Oracle Communications Session Border Controller Release S-C(X)6.1.0M2 solve those issues. Now, the Oracle Communications Session Border Controller sends re-INVITE with the negotiated SDP to the user agent for which the Oracle Communications Session Border Controller performs the call transfer.

### Implementation Details

This section describes the details of how the Oracle Communications Session Border Controller behaves in SIP REFER scenarios with the changes made in Oracle Communications Session Border Controller Release S-C(X)6.1.0M2. The Oracle Communications Session Border Controller makes the new call between Party A and Party C appear as though A were participating to allow the Oracle Communications Session Border Controller's natural media setup occur in the same way as if the REFER had actually been sent to A and A had sent a new INVITE.

When the Oracle Communications Session Border Controller receives a REFER request and determines it needs to handle it locally, it creates a new INVITE made to look like one from Party A. And the Oracle Communications Session Border Controller actually processes this INVITE as though it were from Party A. As a result, new SIP and new media sessions are created with new media ports for Parties A and C. When the INVITE to Party C receives a final response, the Oracle Communications Session Border Controller sends the result to Party B using a SIP NOTIFY request.

If the new INVITE succeeds, the old context and flows disappear and the new context and flows for the A-to-C connection remain in place. Because of the new media ports, the Oracle Communications Session Border Controller sends a re-INVITE to Party A, directing media to the new port and forwarded to Party C. Next, the original dialog

## SIP Signaling Services

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with Party B needs to be terminated; if the Oracle Communications Session Border Controller has not received Party B's BYE, it will wait five second and then send Party B a BYE.

If the INVITE to Party C fails, the new SIP and media sessions are deleted as are the new context and flows. The Oracle Communications Session Border Controller treats Party A differently depending on whether or not a BYE was received from Party B. If a BYE was received from Party B, then the Oracle Communications Session Border Controller sends a BYE to Party A. If not, the original SIP and media sessions as well as the context and media flows remain in tact. This way, Party B can re-establish the call with Party A using a re-INVITE. In the case, the Oracle Communications Session Border Controller waits 32 second before sending a BYE.

If the Oracle Communications Session Border Controller receives a BYE while processing the INVITE to Party C, it sends a CANCEL message to Party C in an attempt to cancel the call. The BYE passes to Party B, and associated sessions, contexts, and flows terminate normally. Still, the Oracle Communications Session Border Controller waits for the final response to the INVITE to Party C. If the Oracle Communications Session Border Controller receives a successful response, it sends an ACK and then a BYE to terminate the abandoned call. If the Oracle Communications Session Border Controller receives an unsuccessful final response, it uses its normal response error handling processes. In either of these last two cases, all sessions, context, and flow are deleted.

Please note that the Oracle Communications Session Border Controller does not remove the a=sendonly attribute from the SDP it sends to Party A during the A-to-B call, and extra media ports are not allocated for the original media session.

## SIP REFER with Replaces

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To support enterprise and call center applications, the Oracle Communications Session Border Controller provides the ability for one party participating in a three-way call to request direct connectivity between the other two parties and to leave the call silently when that connectivity is established. SIP supports this function using the Replaces header in a REFER message, also known as REFER with Replaces.

The most common application of REFER with Replaces handling occurs in a high-level sequence like this:

1. The customer calls a customer service line and reaches—via the Oracle Communications Session Border Controller—an IVR/ACD (Interactive Voice Response system/Automatic Call Distribution system). In some architectures, these are two separate elements.
2. Based on the customer's selection from the menu of options, the IVR/ACD contacts an agent via the Oracle Communications Session Border Controller.
3. Since the ultimate goal is for the IVR/ACD to drop out of the path, it sends a REFER with Replaces to the Oracle Communications Session Border Controller. This message indicates the Oracle Communications Session Border Controller should replace the IVR/ACD endpoint in the call leg with the agent's endpoint.
4. The Oracle Communications Session Border Controller processes the REFER with Replaces, issuing ReINVITES to the customer with the agent's parameters.
5. The IVR/ACD drops out of the media path once the bridged call between the customer and the agent is established.

Note that direct media connectivity between endpoints must be possible in order for the REFER with Replaces to be carried out properly. For example, if both endpoints (such as the customer and agent from the example above) are behind the same firewall, direct media connectivity should be possible. However, if one endpoint is behind a firewall and the other is not, then direct media connectivity may not be possible.

For licensing capacity purposes, note that a bridged session counts as a single call.

## SIP REFER with Replaces Configuration

You enable SIP REFER with Replaces handling either in the realm configuration or in the session agent configuration. This section show you how to configure the feature for session agent, though the steps are the same for adding this feature to a realm.

To enable sending ReINVITES to a referred agent on an existing session/dialog:

1. In Superuser mode, type configure terminal and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET (session-router) # session-agent
ACMEPACKET (session-agent) #
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the ACLI **select** command) the configuration that you want to edit.

4. **options**—Set the options parameter by typing options, a Space, and then the option name **refer-reinvite**. Then press Enter.

```
ACMEPACKET (session-agent) # options +refer-reinvite
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

## SIP REFER-to-BYE

The Oracle Communications Session Border Controller's SIP REFER-to-BYE capability addresses situations when other network elements do not support the REFER method but do offer blind transfer in a SIP BYE request. The target number is encoded in a Reason header of the BYE request. In such cases, the Oracle Communications Session Border Controller terminates the REFER and passes the Refer-To number in a Reason header of the BYE.

You configure both SIP interfaces and SIP session agents with the refer-to-by option to use this function:

- SIP interface—You add this ability to SIP interfaces facing the SIP elements that need to receive a BYE instead of a REFER. This setting only applies when the next hop is not a session agent.
- SIP session agent—The SIP session agent takes precedence over the SIP interface. You add this ability to SIP session agents that need to receive a BYE instead of a REFER. If the next hop SIP element—the remote target in the dialog—is a session agent, in other words, you need to configure the option for it. Note that when you use this option for SIP session agents, the SIP interface or realm on which the REFER is received takes precedence over the REFER-to-BYE capability.

When a REFER request arrives and the REFER-to-BYE capability applies, the Oracle Communications Session Border Controller responds to it with a 202 Accepted and sends a NOTIFY to terminate the implicit refer subscription. This NOTIFY contains a message/sipfrag body with SIP/2.0 200 OK. Upon receiving the response to this NOTIFY, the Oracle Communications Session Border Controller sends a BYE with an added Reason header (encoded with the Refer-To number) to the other end.

The network element that does not accept REFERs takes the BYE with the Reason header and issues a new initial INVITE that initiates transfer, which the Oracle Communications Session Border Controller sees as starting a new and independent session.

## SIP Roaming

This section explains how to configure SIP roaming. SIP roaming lets subscribers move from one active SIP device to another (at the same site or multiple sites) and retain service at the last registering device.

### Overview

The Oracle Communications Session Border Controller supports multiple active registrations for the same user. The softswitch makes decisions regarding the current location of the user and the handling of requests from devices that

## SIP Signaling Services

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are not currently identified as the user location. When there are multiple NATs, the Oracle Communications Session Border Controller is still required to let the softswitch be able to differentiate it.

The Oracle Communications Session Border Controller's SIP roaming ability supports the following features:

- Multiple active registrations from the same user can be cached, allowing subscribers to move from one active SIP device to another (at the same site or multiple sites) and still retain service at the last registering device. With the SIP roaming feature, one person, using multiple devices, can be contacted at all of the devices. These multiple devices (with their unique contact information) register to indicate that they are available for anyone that wants to contact that one person.
- The Oracle Communications Session Border Controller can also inform network devices (such as softswitches) of private SIP device IPv4 addresses (endpoints) and the public firewall address of the user location.

## Process Overview

Caller 1 wants to contact Person A. Caller 1 sends a message to `persona@acmepacket.com`, but Person A has configured more than one SIP-enabled device to accept messages sent to that address. These devices have unique addresses of `desk@10.0.0.4` and `phone2@10.0.0.5`. Person A has `desk@10.0.0.4` and `phone2@10.0.0.5` registered with the Oracle Communications Session Border Controller for anything addressed to `persona@acmepacket.com`.

With the SIP roaming feature, the Oracle Communications Session Border Controller accepts and stores both registrations for `persona@acmepacket.com`. That way, when someone wants to get in touch with Person A, the messages are sent to both devices (`desk@10.0.0.4` and `phone2@10.0.0.5`) until Person A answers one of them. You do not need to configure your Oracle Communications Session Border Controller for this functionality; your Oracle Communications Session Border Controller automatically provides it.

### Using Private IPv4 Addresses

In addition to supporting multiple registries, the Oracle Communications Session Border Controller (SBC) can also distinguish user locations by their private IPv4 address and the IPv4 address of the public firewall. Using this information, the SBC adds private endpoint and public firewall information to Contact headers.

For example, entering this information causes a Contact header that formerly appeared as the following:

```
Contact:<sip:0274116202@63.67.143.217>
```

to subsequently appear as the following:

```
Contact:<sip:0274116202@63.67.143.217;ep=192.168.1.10;fw=10.1.10.21>
```

The SBC SIP proxy reads this information and populates the contact-endpoint and contact-firewall fields with the appropriate values.

### Example 1 With a NAT Firewall

The Oracle Communications Session Border Controller (SBC) SIP proxy is configured with the following changeable parameters:

- endpoint= IP address of the SIP UA
- useradd= IP address of the Firewall Public IP address or the source layer 3 IP address of Register message
- userport= IP address port number of the Firewall Public IP address or the source layer 3 IP address port of Register message
- SBC address=63.67.143.217
- firewall public address=10.1.10.21
- firewall public address port=10000
- SIP endpoint behind firewall=192.168.1.10

SIP message Contact header:

```
Contact:<sip:0274116202@63.67.143.217; endpoint=192.168.1.10; useradd=10.1.10.21; userport=10000; transport=udp>
```



## Example 2 Without a NAT Firewall

The Oracle Communications Session Border Controller SIP proxy is configured with the following changeable parameters:

- useradd= IP address of the SIP UA or the source layer 3 IP address of Register message
- userport= IP address port number of the SIP UA or the source layer 3 IP address port of Register message
- Oracle Communications Session Border Controller address=63.67.143.217
- SIP endpoint=192.168.1.10
- SIP endpoint IP address port=5060

SIP message Contact header:

```
Contact:<sip:0274116202@63.67.143.217; useradd=192.168.1.10; userport=5060; transport=udp>
```

For SIP, the softswitch responsibility is that the URI SD put in the Contact of the REGISTER message should be reflected in the 200-OK response to the REGISTER request. The Contact header of the response should have an expires header parameter indicating the lifetime of the registration.

The following example shows a Oracle Communications Session Border Controller Send:

```
Contact: <sep: 0274116202@63.67.143.217 endpoint=192.168.1.10; useradd=10.1.10.21; userport=10000>;
```

The following examples shows the softswitch Respond:

```
Contact: <sep: 0274116202@63.67.143.217 endpoint=192.168.1.10; useradd=10.1.10.21; userport=10000>; expires=360
```

The contact field for endpoint and firewall parameters only appear in the following:

- Contact header of a REGISTER request sent from the Oracle Communications Session Border Controller to the softswitch server
- Contact header of a REGISTER response sent from the softswitch server to the Oracle Communications Session Border Controller
- Request-URI of an initial INVITE sent from the UT CSA server to the Oracle Communications Session Border Controller

An active endpoint is deleted when it does not register within the registration-interval setting or receives a 401 Unauthorized.

## SIP Roaming Configuration

You can configure the SIP configuration's options parameter to indicate that you want to use the private IP address of the SIP device that the user is using and/or the public firewall address that identifies the location of the device. If defined, these options will be added as parameters to all Contact headers.

You can identify the endpoint and/or firewall information using the following options:

- contact-endpoint=<value> where <value> is the endpoint address or label
- contact-firewall=<value> where <value> is the firewall address or label

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) #
```

## SIP Signaling Services

---

From this point, you can configure SIP config parameters. To view all SIP config parameters, enter a ? at the system prompt.

4. Type **options** followed by a Space.
5. After the Space, type the information for an endpoint or a firewall, or both:

```
contact-endpoint="<label>  
contact-firewall="<label>  
"contact-endpoint="<label>", contact-firewall="<label>"
```

6. Press Enter.

For example, if you want your Oracle Communications Session Border Controller to add private endpoint and public firewall information to Contact headers, and you want to label this information as ep and fw, you would enter the following information in the ACLI.

```
ACMEPACKET(configure)# session-router  
ACMEPACKET(session-router)# sip-config  
ACMEPACKET(sip-config)# options "contact-endpoint="ep", contact-  
firewall="fw"
```

## Embedded Header Support

---

This section explains how to configure embedded header support. The Oracle Communications Session Border Controller supports methods of extracting an embedded P-Asserted-Identity header from a contact header to support E911 when integrated with certain vendor's systems. See RFC 3455 *Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)* for more information.

The embedded header support feature watches for a specified embedded header contained in a Contact header received in a 3XX message. When the specified embedded header is found, the full <header=value> pair is inserted as a unique header in a redirected INVITE message that exits the Oracle Communications Session Border Controller. If the outgoing INVITE message were to contain the specified header, regardless of the use of this feature, the value extracted from the 3XX message replaces the INVITE message's specified header value.

If an incoming Contact header in a 3XX message looks like:

```
Contact: <ESRN@IPv4_Intrado_GW;user=phone?P-Asserted-Identity=%3Csip:+1-  
ESQK@IPv4_My_EAG;user=phone%3E>
```

Then, if you configure your Oracle Communications Session Border Controller to parse for the embedded P-Asserted-Identity header to write as a unique header in the outgoing invite message, the outgoing INVITE and P-Asserted-Identity headers will look like:

```
INVITE SIP: ESRN@IPv4_Intrado_GW;user=phone  
P-Asserted-Identity: +1-ESQK@IPv4_My_EAG;user=phone
```

## Embedded Header Support Configuration

Embedded header support is enabled in the session agent configuration.

To configure embedded header support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# session-agent  
ACMEPACKET(session-agent)#
```

4. Select the session agent where you want this feature.

```
ACMEPACKET(session-agent) # select
<hostname>:
1: asd          realm=      ip=1.0.0.0
2: SIPSA       realm=      ip=10.10.102.1
selection:2
ACMEPACKET(session-agent) #
```

5. **request-uri-headers**—Enter a list of embedded headers extracted from the Contact header that will be inserted in the re INVITE message. To configure this parameter for multiple headers, enclose the headers in double quotes and separate them with spaces. This completes the configuration of embedded header support.

```
ACMEPACKET(session-agent) # request-uri-headers P-Asserted-Identity
```

## Static SIP Header and Parameter Manipulation

This section explains the SIP header and parameter manipulation feature, which lets the Oracle Communications Session Border Controller add, modify, and delete SIP headers and parts of SIP headers called SIP header elements. SIP header elements are the different subparts of the header, such as the header value, header parameter, URI parameter and so on (excluding the header name).

To enable the SIP header and parameter manipulation functionality, you create header manipulation rulesets in which you specify header manipulation rules, as well as optional header element rules that operate on specified header elements. You then apply the header manipulation ruleset as inbound or outbound for a session agent or SIP interface.

### Header Manipulation Rules

Header manipulation rules operate on the header you specify when you configure the rule. A header manipulation rule can also be configured with a list of element rules, each of which would specify the actions you want performed for a given element of this header.

### Header Element Rules

Header element rules perform operations on the elements of a header. Header elements include all subparts of a header; excluding the header name. For example, header value, header parameter, URI parameter, and so on.

### About SIP Header and Parameter Manipulation

Using the SIP header manipulation ruleset, you can cause the Oracle Communications Session Border Controller to:

- Delete a header based on header name match.
- Delete a header based on header name match as well as header value match.
- Add a header.
- Modify the elements of a header (by configuring header element rules):

Add an element to a header.

For example, add a parameter to a header or add a URI parameter to the URI in a header.

Delete an element from a header.

For example, delete a parameter from a header or delete a URI parameter from the URI in a header.

Modify an element of a header.

For example, replace a FQDN with an IPv4 address in a header or replace the value of a parameter in the header.

Delete a message body part

For example, delete the body part if the Content-Type is application/ISUP.

### HMR \$LOCAL\_PORT for Port Mapping

When you configure SIP HMR and set an element-rule's new-value parameter to \$LOCAL\_PORT, the Oracle Communications Session Border Controller maps this value to the real port it uses for each signaling exchange.

#### Role in Trunk Group URI Feature

SIP header and parameter manipulation plays a role in the trunk group URI feature. You need to set the new-value parameter to one of the trunk group values when configuring SIP header rules, if using this feature. (In addition you can configure session agents and session agents groups on the Oracle Communications Session Border Controller to insert trunk group URI parameters in the SIP contact header.

For all trunk group URI support, you must set the appropriate parameters in the SIP header manipulation configuration and in the session agent or session agent group configurations.

For trunk group URI support, the SIP header and parameter manipulation configuration tells the Oracle Communications Session Border Controller where and how to manipulate the SIP message to use originating (access) and terminating (egress) trunk group URI parameters.

### SIP Header and Parameter Manipulation Configuration

This section explains how to configure SIP header and parameter manipulation. First you create a SIP header manipulation ruleset, then the header manipulation rules and optional header element rules you want that ruleset to contain. You then configure a session agent or a SIP interface to use the SIP header and parameter manipulation ruleset in the inbound and outbound directions.

#### Creating SIP Header Manipulation Rulesets

To configure the SIP header manipulation ruleset:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-router path.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-manipulation** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-manipulation  
ACMEPACKET (sip-manipulation) #
```

4. **name**—Enter the name you want to use for this ruleset.


5. **header-rules**—Define the header manipulation rules you want to include in this ruleset.

Type **header-rules** and press Enter.

```
ACMEPACKET (sip-manipulation) # header-rules  
ACMEPACKET (sip-header-rules) #
```

**name**—Enter the name of the header to which this rule applies. (The name you enter here must match a header name.)

This is a case-insensitive string that is compared to the header name for matching. You need to create a rule using the long form of the header name and a rule using the compact form of the header name.

 **Note:** The Request-URI header is identified as request-uri.

**action**—Enter the action you want applied to the header specified in the name parameter. The default value is none. Valid options are:

- **add**—Add a new header, if that header does not already exist.
- **delete**—Delete the header, if it exists.
- **manipulate**—Elements of this header will be manipulated according to the element rules configured.

- **store**—Store the header.
- **none**—No action to be taken.

**match-value**—Enter the value to be matched (only an exact match is supported) with a header value. The action you specify is only performed if the header value matches.

**msg-type**—Enter the message type to which this header rule applies. The default value is any. Valid options are:

- **any**—Both Requests and Reply messages
- **request**—Request messages only
- **reply**—Reply messages only


Type show to display the header rule configuration values.

6. **element-rules**—Define the element rules you want to use to be performed on the elements of the header specified by the header rule.

Type element-rules and press Enter.

```
ACMEPACKET(sip-header-rules) # element-rules
ACMEPACKET(sip-element-rules) #
```

**name**—Enter the name of the element to which this rule applies.

 **Note:** The name parameter usage depends on the element type you enter in step 6. For uri-param, uri-user-param, and header-param it is the parameter name to be added, replaced, or deleted. For all other types, it serves to identify the element rule and any name can be used.

**type**—Enter the type of element on which to perform the action. The default value is none. Valid options are:

- **header-value**—Enter value of the header.
- **header-param-name**—Header parameter name.
- **header-param**—Parameter portion of the header.
- **uri-display**—Display of the SIP URI.
- **uri-user**—User portion of the SIP URI.
- **uri-host**—Host portion of the SIP URI.
- **uri-port**—Port number portion of the SIP URI.
- **uri-param-name**—Name of the SIP URI param.
- **uri-param**—Parameter included in the SIP URI.
- **uri-header-name**—SIP URI header name
- **uri-header**—Header included in a request constructed from the URI.
- **uri-user-param**—User parameter of the SIP URI.

**action**—Enter the action you want applied to the element specified in the name parameter, if there is a match value. The default value is none. Valid options are:

- **none**—No action is taken.
- **add**—Add a new element, if it does not already exist.
- **replace**—Replace the elements.
- **delete-element**—Delete the specified element if it exists. Based on the match value if entered in step 6f.
- **delete-header**—Delete the specified header, if it exists.
- **store**—Store the elements.

**match-val-type**—Enter the type of value that needs to be matched to the match-field entry for the action to be performed. The default value is ANY. Valid options are:

- **IP**—Element value in the SIP message must be a valid IP address to be compared to the match-value field entry. If the match-value field is empty, any valid IP address is considered a match. If the element value is not a valid IP address, it is not considered a match.

## SIP Signaling Services

- **FQDN**—Element value in the SIP message must be a valid FQDN to be compared to the match-value field entry. If the match-value field is empty, any valid FQDN is considered a match. If the element value is not a valid FQDN, it is not considered a match.
- **ANY**—Element value in the SIP message is compared with the match-value field entry. If the match-value field is empty, all values are considered a match.

match-value—Enter the value you want to match against the element value for an action to be performed.

new-value—Enter the value for a new element or to replace a value for an existing element. You can enter an expression that includes a combination of absolute values, pre-defined parameters, and operators.

- Absolute values, with which you can use double quotes for clarity. You must escape all double quotes and back slashes that are part of an absolute value, and enclose the absolute value in double quotes.

For example:

sip:”+\$STRUNK\_GROUP+”.\$STRUNK\_GROUP\_CONTEXT

- Pre-defined parameters always start with a \$. Valid pre-defined parameters are:

Parameter	Description
\$ORIGINAL	Original value of the element is used.
\$LOCAL_IP	IP address of the SIP interface on which the message was received for inbound manipulation; or sent on for outbound manipulation.
\$REMOTE_IP	IP address the message was received from for inbound manipulation; or being sent to for outbound manipulation.
\$REMOTE_VIA_HOST	Host from the top Via header of the message is used.
\$STRUNK_GROUP	Trunk group is used.
\$STRUNK_GROUP_CONTEXT	Trunk group context is used.

- Operators are:

Operator	Description
+	Append the value to the end. For example: acme”+”packet generates acmepacket
+^	Prepends the value. For example: acme”+^”packet generates packetacme
-	Subtract at the end. For example: 112311”-”11 generates 1123
_^	Subtract at the beginning. For example: 112311”-^”11 generates 2311

Examples of entries for the **new-value** field.

```
$ORIGINAL+acme
$ORIGINAL+”my name is john”
```

```
$ORIGINAL+"my name is \"john\""  
$ORIGINAL-^781+^617
```

Type `show` to display the element rule configuration values.

Type `done` to save them.

Repeat steps 6b through 6j to create additional rules.

Type `exit` to return to the header-rules parameters.

7. **methods**—Enter the SIP method names to which you want to apply this header rule. If entering multiple method names, separate them with commas. For example:

```
INVITE,ACK,BYE
```

This field is empty by default. If you leave the method field empty, the header-rule is applied to all methods.

8. Type `exit` to return to the sip-manipulation level.
9. Save your work using the ACLI `done` command.
10. If you want to save this configuration, exit out of configuration mode and type `save-config`.

### Configuring a Session Agent

You can configure a session agent to use the SIP header manipulation ruleset.

To configure a session agent:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter to access the session-router path.

```
ACMEPACKET(configure)# session-router
```

3. Type `session-agent` and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# session-agent  
ACMEPACKET(session-agent)#
```

4. **in-manipulationid**—Enter the name of the SIP header manipulation ruleset you want to apply to inbound SIP packets.

```
ACMEPACKET(session-agent)# in-manipulationid route-stripper
```

5. **out-manipulationid**—Enter the name of the SIP header manipulation ruleset you want to apply to outbound SIP packets.

```
ACMEPACKET(session-agent)# out-manipulationid route-stripper
```

6. Save your work using the ACLI `done` command.
7. If you want to save this configuration, exit out of configuration mode and type `save-config`.

### Configuring a SIP Interface

You can configure a interface to use a SIP header manipulation ruleset.

To configure a SIP interface:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter to access the session-router path.

```
ACMEPACKET(configure)# session-router
```

3. Type `sip-interface` and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router) # sip-interface
ACMEPACKET(sip-interface) #
```

4. **in-manipulationid**—Enter the name of the SIP header manipulation ruleset you want to apply to SIP packets in the ingress direction.

```
ACMEPACKET(sip-interface) # in-manipulationid
```

5. **out-manipulationid**—Enter the name of the SIP header manipulation ruleset you want to apply to SIP packets in the egress direction.

```
ACMEPACKET(sip-interface) # out-manipulationid
```

6. Save your work using the ACLI **done** command.

7. If you want to save this configuration, exit out of configuration mode and type **save-config**.

### Example 1 Stripping All Route Headers

This example explains how to strip all route headers from a SIP packet. First, you create a header manipulation ruleset, in the example it is called route-stripper. Then you configure the list of header manipulation rules you need to strip route headers. In this case, you only need one rule named Route (to match the Route header name) with the action set to Delete.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) # sip-manipulation
ACMEPACKET(sip-manipulation) # name route-stripper
ACMEPACKET(sip-manipulation) # header-rules
ACMEPACKET(sip-header-rules) # name Route
ACMEPACKET(sip-header-rules) # action Delete
ACMEPACKET(sip-header-rules) # done
header-rule
  name                Route
  action              delete
  match-value
  msg-type            any
ACMEPACKET(sip-header-rules) # ex
ACMEPACKET(sip-manipulation) # done
sip-manipulation
  name                route-stripper
  header-rule
    name              Route
    action            delete
    match-value
    msg-type          any
```

### Example 2 Stripping an Existing Parameter and Adding a New One

This example explains how to strip the user parameter from the Contact header URI and add the acme parameter with value as LOCAL IP, only for requests. First you create a header manipulation ruleset, in the example it is called param-stripper1. You then configure a list of header rules you need. In this case, you only need one rule named Contact (to match the Contact header name), with action set to manipulate (indicating the elements of this header would be manipulated). Next, you configure a list of element rules for the Contact header rule.

In this case you configure two element rules; one to strip the uri parameter user (the rule name user matches the param name user) and the other to add the uri parameter acme (the rule name acme matches the param name acme).

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) # sip-manipulation
ACMEPACKET(sip-manipulation) # name param-stripper1
ACMEPACKET(sip-manipulation) # header-rules
ACMEPACKET(sip-header-rules) # name Contact
ACMEPACKET(sip-header-rules) # action manipulate
ACMEPACKET(sip-header-rules) # msg-type request
ACMEPACKET(sip-header-rules) # element-rules
```



```

ACMEPACKET (sip-element-rules) # name user
ACMEPACKET (sip-element-rules) # type uri-param
ACMEPACKET (sip-element-rules) # action delete-element
ACMEPACKET (sip-element-rules) # done
element-rule
    name                user
    type                uri-param
    action              delete-element
    match-val-type     any
    match-value
    new-value
ACMEPACKET (sip-element-rules) # name acme
ACMEPACKET (sip-element-rules) # action add
ACMEPACKET (sip-element-rules) # type uri-param
ACMEPACKET (sip-element-rules) # new-value "$LOCAL_IP"
ACMEPACKET (sip-element-rules) # done
element-rule
    name                acme
    type                uri-param
    action              add
    match-val-type     any
    match-value
    new-value          "$LOCAL_IP"
ACMEPACKET (sip-element-rules) # ex
ACMEPACKET (sip-header-rules) # done
header-rule
    name                Contact
    action              manipulate
    match-value
    msg-type            request
    element-rule
        name            user
        type            uri-param
        action          delete-element
        match-val-type any
        match-value
        new-value
    element-rule
        name            acme
        type            uri-param
        action          add
        match-val-type any
        match-value
        new-value      "$LOCAL_IP"
ACMEPACKET (sip-header-rules) # ex
ACMEPACKET (sip-manipulation) # done
sip-manipulation
    name                param-stripper1
    header-rule
        name            Contact
        action          manipulate
        match-value
        msg-type        request
        element-rule
            name            user
            type            uri-param
            action          delete-element
            match-val-type any
            match-value
            new-value
element-rule
    name                acme
    type                uri-param
    action              add

```

match-val-type	any
match-value	
new-value	"\$LOCAL_IP"

For example, if the IP address of the SIP interface (\$LOCAL\_IP) is 10.1.2.3 and the Oracle Communications Session Border Controller receives the following Contact header:

```
Contact: <sip:1234@10.4.5.6;user=phone>
```

The header rule is applied to strip the user parameter from the Contact header URI and add the acme parameter with the value 10.1.2.3:

```
Contact: <sip:1234@10.4.5.6;acme=10.1.2.3>
```

## SIP HMR (Header Manipulation Rules)

---

SIP header manipulation can also be configured in a way that makes it possible to manipulate the headers in SIP messages both statically and dynamically. Using this feature, you can edit response headers or the Request-URI in a request, and change the status code or reason phrase in SIP responses.

Static SIP Header and Parameter Manipulation allows you to set up rules in your Oracle Communications Session Border Controller configuration that remove and/or replace designated portions of specified SIP headers. SIP HMR allows you to set up dynamic header manipulation rules, meaning that the Oracle Communications Session Border Controller has complete control over alterations to the header value. More specifically:

- The Oracle Communications Session Border Controller can search header for dynamic content or patterns with the header value. It can search, for example, for all User parts of a URI that begin with 617 and end with 5555 (e.g., 617...5555).
- The Oracle Communications Session Border Controller can manipulate any part of a patterns match with any part of a SIP header. For example, 617 123 5555 can become 617 231 5555 or 508 123 0000, or any combination of those.

To provide dynamic header manipulation, the Oracle Communications Session Border Controller uses regular expressions to provide a high degree of flexibility for this feature. This allows you to search a specific URI when you do not know that value of the parameter, but want to use the matched parameter value as the header value. It also allows you to preserve matched sections of a pattern, and change what you want to change.

You can apply header manipulation to session agents, SIP interfaces, and realms. You do so by first setting up header manipulations rules, and then applying them in the configurations where they are needed. Within the header manipulation rules, there are sets of element rules that designate the actions that need to be performed on a given header.

Each header rule and each element rule (HMR) have a set of parameters that you configure to identify the header parts to be manipulated, and in what way the Oracle Communications Session Border Controller is to manipulate them. These parameters are explained in detail, but the parameter that can take regular expression values is **match-value**. This is where you set groupings that you want to store, match against, and manipulate.

Generally, you set a header rule that will store what you want to match, and then you create subsequent rules that operate on this stored value. Because header rules and element rules are applied sequentially, it is key to note that a given rule performs its operations on the results of all the rules that you have entered before it. For example, if you want to delete a portion of a SIP header, you would create Rule 1 that stores the value for the purpose of matching, and then create Rule 2 that would delete the portion of the header you want removed. This prevents removing data that might be used in the other header rules.

Given that you are using regular expression in this type of configuration, this tightly sequential application of rules means that you must be aware of the results to be yielded from the application of the regular expressions you enter. When you set a regular expression match value for the first rule that you enter, the Oracle Communications Session Border Controller takes the results of that match, and then a second rule might exist that tells the Oracle Communications Session Border Controller to use a new value if it the second rule's match value finds a hit (and only 10 matches, 0-9, are permitted) for the results (yield) from applying the first rule.

Consider the example of the following regular expression entry made for a **match-value** parameter: 'Trunk(.+)', which might be set as that match value in the first rule you configure. Given a SIP element rule called uri-param and the param-name tgid, it can yield two values:

- Grouping 0—The entire matching string (Trunk1)
- Grouping 1—The grouping (1)

In turn, these groupings can be referenced in an element rule by using this syntax:

```
$<header rule name >.$<element rule name>.$<value>
```

Additional syntax options that can be used with this feature are:

- \$headerName[['index']]
- \$headerName[['index']][\$index]
- \$headerName[['index']][\$elementName]
- \$headerName[['index']][\$elementName][\$index]

## Guidelines for Header and Element Rules

Header rules and element rules share these guidelines:

- References to groupings that do not exist result in an empty string.
- References to element rule names alone result in a Boolean condition of whether the expression matched or not.
- A maximum of ten matches are allowed for a regular expression. Match 0 (grouping 0) is always the match of the entire matching string; subsequent numbers are the results for other groups that match.

## Precedence

The Oracle Communications Session Border Controller applies SIP header rules in the order you have entered them. This guards against the Oracle Communications Session Border Controller removing data that might be used in the other header rules.

This ordering also provides you with ways to use manipulations strategically. For example, you might want to use two rules if you want to store the values of a regular expression. The first rule would store the value of a matched regular expression, and the second could delete the matched value.

In addition to taking note of the order in which header rules are configured, you now must also configure a given header rule prior to referencing it. For example, you must create Rule1 with the action store for the Contact header BEFORE you can create Rule2 which uses the stored value from the Contact header.

## Duplicate Header Names

If more than one header exists for the header name you have configured, the Oracle Communications Session Border Controller stores the value where it can be referenced with the optional syntax `$<header rule name>[index]`. Additional stored header values are indexed in the order in which they appear within the SIP message, and there is no limit to the index.

Possible index values are:

- ~ — The Oracle Communications Session Border Controller references the first matching header
- \* — The Oracle Communications Session Border Controller references all headers
- ^ — The Oracle Communications Session Border Controller references the last stored header in the header rule

## Performing HMR on a Specific Header

HMR has been enhanced so that you can now operate on a specific instance of a given header. The syntax you use to accomplish this is similar to that you used to refer to a specific header rule stored value instance.

Using the header-name parameter, you can now add a trailing [`<index>`] value after the header name. This [`<index>`] is a numerical value representing the specific instance of the header on which to operate. However, the Oracle

Communications Session Border Controller takes no action if the header does not exist. You can also use the caret (^) to reference the last header of that type (if there are multiple instances)

The count for referencing is zero-based, meaning that the first instance of the header counts as 0.

Note that the header instance functionality has no impact on HMR's add action, and you cannot use this feature to insert headers into a specific location. Headers are added to the end of the list, except that Via headers are added to the top.

### Multiple SIP HMR Sets

In general you use SIP HMR by configuring rules and then applying those rules to session agents, realms, or SIP interfaces in the inbound or outbound direction. In addition, the Oracle Communications Session Border Controller has a set method for how certain manipulation rules take precedence over others. For instance, inbound SIP manipulation rules defined in a session agent take precedence over any configured for a realm, and the rules for a realm take precedence over SIP interface manipulation rules.

The multiple SIP HMR feature gives you the ability to:

- Apply multiple inbound and outbound manipulations rules to a SIP message
- Provision the order in which the Oracle Communications Session Border Controller applies manipulation rules

The **action** parameter in the header rules configuration now takes the value `sip-manip`. When you set the parameter to `sip-manip`, you then configure the **new-value** parameter with the name of a SIP manipulation rule that you want to invoke. The values for the **match-value**, **comparison-type**, and **methods** parameters for invoked rule are all supported. This means that the manipulation defined by the rules identified in the **new-value** parameter are carried out when the values for the **match-value**, **comparison-type**, and **methods** parameters are true.

The relationship between manipulation rules and manipulation rule sets is created once you load your configuration, meaning that the order in which you enter them does not matter. It also means that the Oracle Communications Session Border Controller cannot dynamically perform validation as you enter rules, so you should use the CLI **verify-config** command to confirm your manipulation rules contain neither invalid nor circular references. Invalid references are those that point to SIP manipulation rules that do not exist, and circular references are those that create endless loops of manipulation rules being carried out over and over. If you load a configuration exhibiting either of these issues, the Oracle Communications Session Border Controller forces the action value for the rule to **none** and the rule will not be used.

### MIME Support

Using the SIP HMR feature set, you can manipulate MIME types in SIP message bodies. While you can manipulate the body of SIP messages or a specific content type using other iterations of SIP HMR, this version gives you the power to change the MIME attachment of a specific type within the body by using regular expressions. To achieve this, you use the **find-replace-all** action type, which enables the search for a particular string and the replacement of all matches for that type. Although you use **find-replace-all** to manipulate MIME attachments, it can also be used to achieve other goals in SIP HMR.

Note that using **find-replace-all** might consume more system resources than other HMR types. Therefore this powerful action type should only be used when another type cannot perform the type of manipulation you require.

### Find and Replace All

To manipulate a particular portion of the MIME attachment, for example when removing a certain attribute within the content type of `application/sdp`, the Oracle Communications Session Border Controller (SBC) would need to search the content multiple times because:

- SDP can have more than one media line
- The SIP message body can contain more than one `application/sdp`.

The **find-replace-all** action type works for SIP header rules and for element rules. You can use it for all manipulation types from the entire header value, to the URI specific parameters, to MIME attachment.

For this action type, it does not matter what you configure the comparison type, which is atypical for actions types, as the comparison type is vital to the others. **find-replace-all**, however, binds the comparison type to the pattern rule. Thus, the SBC treats the match value as a regular expression, and it ignores any configured comparison type value in favor of the pattern rule. This type of action is both a comparison and action: For each regular expression match within the supplied string, the SBC substitutes the new value for that match. Yet if you want to replace a certain portion of the regular expression and not the entire matched expression, you need to use a subgroup of expressions and the right syntax to indicate the sub-group replacement index.

You can indicate the sub-group replacement syntax by adding the string `[:n:]` to the end of the regular expression—where `n` is a number between 0 and 9. For example, given the following settings:

- `action=find-replace-all`
- `match-value=sip:(user)@host[:1:]`
- `new-value=bob`

you create a new rule to replace only the user portion of the URI that searches for the regular expression and replaces all instances of the user subgroup with the value bob.

Taking advantage of the **find-replace-all**'s recursive nature, you can replace all the 0 digits in a telephone number with 1:

- `action=find-replace-all`
- `match-value=0`
- `new-value=1`

So for the user portion of a URI—or for any other string—with a value 1-781-308-4400 would be replaced as 1-781-318-4411.

If you leave the **new-value** parameter blank for **find-replace-all**, the SBC replaces the matched sub-group with an empty string—an equivalent of deleting the sub-group match. You can also replace empty sub-groups, which is like inserting a value within the second sub-group match. For example, `user()@host.com[:1:]` with a configured **new-value** `_bob` yields `user_bob@host.com`.

When you use **find-replace-all**, you cannot use the following **parameter-type** values: **uri-param-name**, **uri-header-name**, and **header-param-name**. These values are unusable because the SBC only uses case-sensitive matches for the match-value to find the parameter name within the URI. Since it can only be found by exact match, the SBC does not support finding and replacing that parameter.

## Escaped Characters

SIP HMR's support for escaped characters allows for searches for values you would be unable to enter yourself. Because they are necessary to MIME manipulation, support for escaped characters now includes:

- `\f`
- `\n`
- `\r`
- `\t`
- `\v`

## New Reserved Word

To allow you to search for carriage returns and new lines, the SIP HMR MIME feature also adds support for the reserved word `$CRLF`. Because you can search for these values and replace them, you also must be able to add them back in when necessary. Configuring `$CRLF` in the **new-value** parameter always resolves to `/r/n`, which you normally cannot otherwise enter through the ACLI.

## About the MIME Value Type

Introduced to modify the MIME attachment, SIP HMR supports a **mime** value for the **type** parameter in the element rules configuration. Like the **status-code** and **reason-phrase** values, you can only use the **mime** type value against a specific header—which in this case, is Content-Type.

When you set the element rule type to **mime**, you must also configure the **parameter-name** with a value. This step is a requirement because it sets the content-type the Oracle Communications Session Border Controller manipulates in a specific part of the MIME attachment. You cannot leave this parameter blank; the Oracle Communications Session Border Controller does not let you save the configuration if you do. When you use the **store** action on a multi-part MIME attachment that has different attachment types, the Oracle Communications Session Border Controller stores the final instance of the content-type because it does not support storing multiple instances of element rule stored values.

In the event you do not know the specific content-type where the Oracle Communications Session Border Controller will find the **match-value**, you can wildcard the **parameter-name** by setting with the asterisk (\*) as a value. You cannot, however, set partial content-types (i.e., application/\*). So configured, the Oracle Communications Session Border Controller loops through the MIME attachment’s content types.

You can set the additional **action** types listed in this table with the described result:

Action Type	Description
delete-element	Removes the matched mime-type from the body. If this is the last mime-type within in message body, the Oracle Communications Session Border Controller removes the Content-Type header.
delete-header	Removes all body content and removes the Content-Type header.
replace	Performs a complete replacement of the matched mime-type with the new-value you configure.
find-replace-all	Searches the specifies mime-type’s contents and replaces all matching regular expressions with the new-value you configure
store	Stores the final instance of the content-type (if there are multi-part MIME attachments of various attachment types)
add	Not supported

MIME manipulation does not support manipulating headers in the individual MIME attachments. For example, the Oracle Communications Session Border Controller cannot modify the Content-Type given a portion of a message body like this one:

```

--boundary-1
Content-Type: application/sdp
v=0
o=use1 53655765 2353687637 IN IP4 192.168.1.60
s=-
c=IN IP4 192.168.1.60
t=0 0
m=audio 10000 RTP/AVP 8
a=rtpmap:8 PCMA/8000/1
a=sendrecv
a=ptime:20
a=maxptime:200
    
```

## Back Reference Syntax

You can use back reference syntax in the **new-value** parameter for header and element rules configurations. Denoted by the use of \$1, \$2, \$3, etc. (where the number refers to the regular expression’s stored value), you can

reference the header and header rule's stored value without having to use the header rule's name. It instead refers to the stored value of this rule.

For example, when these settings are in place:

- header-rule=changeHeader
- action=manipulate
- match-value=(.+)([^\;])

you can set the **new-value** as sip:\$2 instead of **sip:\$changeHeader.\$2**.

You can use the back reference syntax for:

- Header rule **actions** **manipulate** and **find-replace-all**
- Element rule **actions** **replace** and **find-replace-all**

Using back reference syntax simplifies your configuration steps because you do not need to create a store rule and then manipulate rule; the manipulate rule itself performs the store action if the **comparison-type** is set to **pattern-rule**.

## Notes on the Regular Expression Library

In the regular expression library, the dot (.) character no longer matches new lines or carriage returns. Conversely, the not-dot does match new lines and carriage returns. This change provides a safety mechanism preventing egregious backtracking of the entire SIP message body when there are no matches. Thus, the Oracle Communications Session Border Controller reduces backtracking to a single line within the body. In addition, there is now support for:

Syntax	Description
\s	Whitespace
\S	Non-whitespace
\d	Digits
\D	Non-digits
\R	Any \r, \n, \r\n
\w	Word
\W	Non-word
\A	Beginning of buffer
\Z	End of buffer
\	Any character including newline, in the event that the dot (.) is not

In addition, there is:

- Escaped character shortcuts (\w\W\S\s\d\D\R) operating inside brackets [...]

## SIP Message-Body Separator Normalization

The Oracle Communications Session Border Controller supports SIP with Multipurpose Internet Mail Extension (MIME) attachments — up to a maximum payload size of 64KB — and has the ability to allow more than the required two CRLFs between the SIP message headers and the multipart body's first boundary. The first two CRLFs that appear in all SIP messages signify the end of the SIP header and the separation of the header and body of the message, respectively. Sometimes additional extraneous CRLFs can appear within the preamble before any text.

The Oracle Communications Session Border Controller works by forwarding received SIP messages regardless of whether they contain two or more CRLFs. Although three or more CRLFs are legal, some SIP devices do not accept more than two.

## SIP Signaling Services

---

The solution to ensuring all SIP devices accept messages sent from the Oracle Communications Session Border Controller is to strip all CRLFs located at the beginning of the preamble before the appearance of any text, ensuring that there are no more than two CRLFs between the end of the last header and the beginning of the body within a SIP message. You enable this feature by adding the new `stripPreambleCrlf` option to the global SIP configuration.

To enable the stripping of CRLFs in the preamble:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET# (configure)
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
ACMEPACKET# (session-router)
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) #
```

4. **options**—Set the options parameter by typing `options`, a Space, the option name `stripPreambleCrlf` with a plus sign.

```
ACMEPACKET (sip-config) # options +stripPreambleCrlf
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the global SIP configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

## SIP Header Pre-Processing HMR

By default, the Oracle Communications Session Border Controller (SBC) performs in-bound SIP manipulations after it carries out header validation. Adding the **inmanip-before-validate** option in the global SIP configuration allows the SBC to perform HMR on received requests prior to header validation. Because there are occasional issues with other SIP implementations—causing invalid headers to be used in messages they send to the SBC—it can be beneficial to use HMR to remove or repair these faulty headers before the request bearing them are rejected.

When configured to do so, the SBC performs pre-validation header manipulation immediately after it executes the top via check. Inbound SIP manipulations are performed in order of increasing precedence: SIP interface, realm, and session agent.

The fact that the top via check happens right before the SBC carries out pre-validation header manipulations means that you cannot use this capability to repairs the first via header if it is indeed invalid. If pre-validation header manipulation were to take place at another time during processing, it would not be possible to use it for SIP session agents. The system learns of matching session agents after top via checking completes.

For logistical reasons, this capability does not extend to SIP responses. Inbound manipulation for responses cannot be performed any sooner that it does by default, a time already preceding any header validation.

To enable SIP header pre-processing:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET# (configure)
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
ACMEPACKET# (session-router)
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.



```
ACMEPACKET(session-router) # sip-config
ACMEPACKET(sip-config) #
```

4. **options**—Set the options parameter by typing options, a Space, the option name inmanip-before-validate with a plus sign.

```
ACMEPACKET(sip-config) # options +inmanip-before-validate
```

This value allows a the SBC to perform pre-validation header manipulation in order of increasing precedence: SIP interface, realm, and session agent.

5. Save and activate the configuration.

## Best Practices

This section lists practices that AOracle recommends you follow for successful implementation of this feature.

- Define all storage rules first.

This recommendation is made because each subsequent header rule processes against the same SIP message, so each additional header rules works off of the results from the application of the rule that precedes it.

In general, you want to store values from the original SIP header rather than from the iteratively changed versions.

- Implement rules at the element rule rather than the header rule level.

Header rules should only be a container for element rules.

- When you are creating rules to edit a header, add additional element rules to modify a single header rather than try to create multiple header rules each with one element rule. That is, create multiple element rules within a header rule rather than creating multiple header rules.
- Do not use header or element rule names that are all capital letters (i.e., \$IP\_ADDRESS). Capitals currently refer to predefined rules that are used as macros, and they might conflict with a name that uses capital letters.

## About Regular Expressions

Two of the most fundamental ideas you need to know in order to work with regular expressions and with this feature are:

- Regular expressions are a way of creating strings to match other string values.
- You can use groupings in order to create stored values on which you can then operate.

To learn more about regex, you can visit the following Web site, which has information and tutorials that can help to get you started:<http://www.regular-expressions.info/>.

Many of the characters you can type on your keyboard are literal, ordinary characters—they present their actual value in the pattern. Some characters have special meaning, however, and they instruct the regex function (or engine which interprets the expressions) to treat the characters in designated ways. The following table outlines these “special characters” or metacharacters.

Character	Name	Description
.	dot	Matches any one character, including a space; it will match one character, but there must be one character to match.  Literally a . (dot) when bracketed ([.]), or placed next to a \ (backslash).
*	star/asterisk	Matches one or more preceding character (0, 1, or any number), bracketed carrier class, or group in parentheses. Used for quantification.  Typically used with a . (dot) in the format .* to indicate that a match for any character, 0 or more times.

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Character	Name	Description
		Literally an * (asterisk) when bracketed ([*]).
+	plus	Matches one or more of the preceding character, bracketed carrier class, or group in parentheses. Used for quantification.  Literally a + (plus sign) when bracketed ([+]).
	bar/vertical bar/pipe	Matches anything to the left or to the right; the bar separates the alternatives. Both sides are not always tried; if the left does not match, only then is the right attempted. Used for alternation.
{	left brace	Begins an interval range, ended with } (right brace) to match; identifies how many times the previous singles character or group in parentheses must repeat.  Interval ranges are entered as minimum and maximums ({minimum,maximum}) where the character/group must appear a minimum of times up to the maximum. You can also use these character to set magnitude, or exactly the number of times a character must appear; you can set this, for example, as the minimum value without the maximum ({minimum,}).
?	question mark	Signifies that the preceding character or group in parentheses is optional; the character or group can appear not at all or one time.
^	caret	Acts as an anchor to represent the beginning of a string.
\$	dollar sign	Acts as an anchor to represent the end of a string.
[	left bracket	Acts as the start of a bracketed character class, ended with the ] (right bracket). A character class is a list of character options; one and only one of the characters in the bracketed class must appear for a match. A - (dash) in between two character enclosed by brackets designates a range; for example [a-z] is the character range of the lower case twenty-six letters of the alphabet.  Note that the ] (right bracket) ends a bracketed character class unless it sits directly next to the [ (left bracket) or the ^ (caret); in those two cases, it is the literal character.
(	left parenthesis	Creates a grouping when used with the ) (right parenthesis). Groupings have two functions:  They separate pattern strings so that a whole string can have special characters within it as if it were a single character.

Character	Name	Description
		They allow the designated pattern to be stored and referenced later (so that other operations can be performed on it).

## Expression Building Using Parentheses

You can now use parentheses ( ) when you use HMR to support order of operations and to simplify header manipulation rules that might otherwise prove complex. This means that expressions such as (sip + urp) - (u + rp) can now be evaluated to sip. Previously, the same expression would have evaluated to sipurprp. In addition, you previously would have been required to create several different manipulation rules to perform the same expression.

## SIP Manipulation Configuration

This section explains the parameters that appear in the subelements for the SIP manipulations configuration. Within the SIP manipulations configuration, you can set up SIP header rules, and within those header rules you can configure element rules.

This section also contains several configuration examples for different applications of the HMR feature.

### Configuring SIP Header Manipulation Rules

To configure dynamic SIP header manipulation rules:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-manipulation** and press Enter.

```
ACMEPACKET(session-router)# sip-manipulation
ACMEPACKET(sip-manipulation)#
```

4. Type **header-rules** and press Enter.

```
ACMEPACKET(sip-manipulation)# header-rules
```

5. **name**—Enter the unique identifier for this SIP HMR. There is no default for this value.

6. **header-name**—Enter the name of the header on which you want the Oracle Communications Session Border Controller (SBC) to use this HMR. There is no default for this parameter.

Set this parameter to @status-line, where the at-sign (@)—not allowed in SIP header names—to prevent undesired matches with header having the name status-code.

7. **msg-type**—Specify the type of message to which this SIP HMR will be applied. The default value is **any**. The valid values are:

- any | request | reply

8. **methods**—Enter the method type to use when this SIP HMR is used, such as INVITE, ACK, or CANCEL. When you do not set the method, the SBC applies the rule across all SIP methods.

9. **comparison-type**—Enter the way that you want SIP headers to be compared from one of the available. This choice dictates how the SBC processes the match rules against the SIP header. the default is **refer-case-sensitive**. The valid values are:

- boolean | refer-case-sensitive | refer-case-insensitive | pattern-rule | case-sensitive | case-insensitive

10. **action**—Enter the action that you want this rule to perform on the SIP header. The default value is **none**. The valid values are:

- add | delete | manipulate | store | none

Remember that you should enter rules with the action type store before you enter rules with other types of actions.

When you set the action type to store, the SBC always treats the match value you enter as a regular expression. As a default, the regular expression is uses for the match value is `.+` (which indicates a match value of at least one character), unless you set a more specific regular expression match value.

- 11. match-value**—Enter the value to match against the header value in SIP packets; the SBC matches these against the entire SIP header value. This is where you can enter values to match using regular expression values. Your entries can contain Boolean operators.

When you configure HMR (using SIP manipulation rules, elements rules, etc.), you can now use escape characters in the **match-value** parameter to support escaping Boolean and string manipulation operators..

You can also escape the escape character itself, so that it is used as a literal string. For example, the SBC now treats the string `\+1234` as `+1234`.

The following are escape characters: `+`, `-`, `+`, `-`, `&`, `|`, `\`, `(`, `)`, `.`, `$`, `^`, and `“`.

You can also use two variables, `$REMOTE_PORT` and `$LOCAL_PORT`, which resolve respectively to the far-end and remote UDP or TCP port value.

- 12. new-value**—When the action parameter is set to add or to manipulate, enter the new value that you want to substitute for the entire header value. This is where you can set stored regular expression values for the SBC to use when it adds or manipulates SIP headers.

When you configure HMR (using SIP manipulation rules, elements rules, etc.), you can now use escape characters in the **new-value** parameter to support escaping Boolean and string manipulation operators..

You can also escape the escape character itself, so that it is used as a literal string. For example, the SBC now treats the string `\+1234` as `+1234`.

The following are escape characters: `+`, `-`, `+`, `-`, `&`, `|`, `\`, `(`, `)`, `.`, `$`, `^`, and `“`.

You can also use two variables, `$REMOTE_PORT` and `$LOCAL_PORT`, which resolve respectively to the far-end and remote UDP or TCP port value.

### Configuring SIP Header Manipulation Element Rules

Element rules are a subset of the SIP header manipulation rules and are applied at the element type level rather than at the entire header value.

To configure dynamic SIP header manipulation rules:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router  
ACMEPACKET(session-router)#
```

3. Type **sip-manipulation** and press Enter.

```
ACMEPACKET(session-router)# sip-manipulation  
ACMEPACKET(sip-manipulation)#
```

4. Type **header-rules** and press Enter.

```
ACMEPACKET(sip-manipulation)# header-rules
```

5. Type **element-rules** and press Enter.

```
ACMEPACKET(sip-header-rules)# element-rules  
ACMEPACKET(sip-element-rules)#
```

6. **name**—Enter the unique identifier for this element rule. There is no default for this value.

7. **parameter-name**—Enter the SIP header parameter/element on which you want the Oracle Communications Session Border Controller (SBC) to use this rule. There is no default for this parameter.
8. **type**—Specify the type of parameter to which this element rule will be applied. The default value is **none**. The valid values are:
- header-value | header-param-name | header-param | uri-display | uri-user | uri-user-param | uri-host | uri-port | uri-param-name | uri-param | uri-header-name | uri-header
- To configure HMR so that there is impact only on the status-line; the value will be used for matching according to the **comparison-type**:
- **status-code**—Designates the status code of the response line; accepts any string, but during the manipulation process only recognizes the range from 1 to 699.
  - **reason-phrase**—Designates the reason of the response line; accepts any string.
9. **match-val-type**—Enter the value type that you want to match when this rule is applied. The default value is **ANY**. Valid values are:
- IP | FQDN | ANY
10. **comparison-type**—Enter the way that you want SIP headers to be compared from one of the available. This choice dictates how the SBC processes the match rules against the SIP header parameter/element. The default is **refer-case-sensitive**.
- boolean | refer-case-sensitive | refer-case-insensitive | pattern-rule
11. **action**—Enter the action that you want this rule to perform on the SIP header parameter/element. The default is **none**. The valid rules are:
- add | replace | delete-element | delete-header | store | none
- Remember that you should enter rules with the action type store before you enter rules with other types of actions.
- When you set the action type to store, the SBC always treats the match value you enter as a regular expression. As a default, the regular expression is uses for the match value is `.+` (which indicates a match value of at least one character), unless you set a more specific regular expression match value.
12. **match-value**—Enter the value to match against the header value in SIP packets; the SBC matches these against the value of the parameter/element. This is where you can enter values to match using regular expression values, or stored pattern matches. Your entries can contain Boolean operators.
- When you configure HMR (using SIP manipulation rules, elements rules, etc.), you can now use escape characters in the **match-value** parameter to support escaping Boolean and string manipulation operators..
- You can also escape the escape character itself, so that it is used as a literal string. For example, the SBC now treats the string `\+1234` as `+1234`.
- The following are escape characters: `+`, `-`, `+`, `-`, `&`, `|`, `\`, `(`, `)`, `.`, `$`, `^`, and `“`.
- You can also use two variables, `$REMOTE_PORT` and `$LOCAL_PORT`, which resolve respectively to the far-end and remote UDP or TCP port value.
13. **new-value**—When the action parameter is set to add or to manipulate, enter the new value that you want to substitute for the entire header value. This is where you can set stored regular expression values for the SBC to use when it adds or manipulates parameters/elements.
- When you configure HMR (using SIP manipulation rules, elements rules, etc.), you can now use escape characters in the **new-value** parameter to support escaping Boolean and string manipulation operators..
- You can also escape the escape character itself, so that it is used as a literal string. For example, the SBC now treats the string `\+1234` as `+1234`.
- The following are escape characters: `+`, `-`, `+`, `-`, `&`, `|`, `\`, `(`, `)`, `.`, `$`, `^`, and `“`.
- You can also use two variables, `$REMOTE_PORT` and `$LOCAL_PORT`, which resolve respectively to the far-end and remote UDP or TCP port value.

### Status-Line Manipulation and Value Matching

The Oracle Communications Session Border Controller's HMR feature has been enhanced to support the ability to change the status code or reason phrase in SIP responses. This addition—the ability to edit status-lines in responses—builds on HMR's existing ability to edit response headers or the Request-URI in a request.

This section shows you how to configure SIP HMR when you want the Oracle Communications Session Border Controller to drop a 183 Session Progress response when it does not have SDP, though flexibility is built into this feature so that you can use it to achieve other ends. In addition, you can now set the SIP manipulation's **match-value** parameter with Boolean parameters (AND or OR).

### Setting the Header Name

SIP header rules (part of the SIP manipulation configuration) now support a new value for the header-name parameter. The value is `@status-line`, where the at-sign (`@`)—not allowed in SIP header names—prevents undesired matches with header having the name status-code.

To set the header name for SIP header rules:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-manipulation** and press Enter.

```
ACMEPACKET (session-router) # sip-manipulation
ACMEPACKET (sip-manipulation) #
```

4. Type **header-rules** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # header-rules
ACMEPACKET (sip-header-rules) #
```

5. **header-name**—Enter the new value for the **header-name** parameter: `@status-line`.

### Setting the Element Type

In the element rules (a subset of the SIP header rules configuration), you can now set the **type** parameter to either of the following values, both of which will only have an impact on the status-line:

- **status-code**—Designates the status code of the response line; accepts any string, but during the manipulation process only recognizes the range from 1 to 699
- **reason-phrase**—Designates the reason of the response line; accepts any string

Like other rule types you can set, the Oracle Communications Session Border Controller matches against the value for these using case-sensitive, case-insensitive, or pattern-rule matching (set in the **comparison-type** parameter for the element rule).

To set the element type:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-manipulation** and press Enter.

```
ACMEPACKET (session-router) # sip-manipulation
ACMEPACKET (sip-manipulation) #
```

4. Type **header-rules** and press Enter.

```
ACMEPACKET(session-router) # header-rules
ACMEPACKET(sip-header-rules) #
```

5. Type **element-rule** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(sip-header-rules) # element-rules
ACMEPACKET(sip-element-rules) #
```

6. **type**—Enter either status-code or reason-phrase, the value of which will be used for matching according to the **comparison-type**.

### Setting the Match Value

Note that for the SIP header rules and for the SIP element rules, the **match-value** parameter can now be set with these Boolean operators:

- and (for which the syntax is the ampersand &)
- or (for which the syntax is the pipe|)

However, you can only use Boolean operators in this value when you set the **comparison-type** parameter to **pattern-rule** and are evaluating stored matches. The Oracle Communications Session Border Controller evaluates these Boolean expressions from left to right, and does not support any grouping mechanisms that might change the order of evaluation. For example, the Oracle Communications Session Border Controller evaluates the expression  $A \& B | C$  (where  $A=true$ ,  $B=false$ , and  $C=true$ ) as follows:  $A \& B = false$ ;  $false | true = true$ .

You can set the match-value for the SIP header rules or for the SIP element rules.

To set a match value in the SIP header rules configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

3. Type **sip-manipulation** and press Enter.

```
ACMEPACKET(session-router) # sip-manipulation
ACMEPACKET(sip-manipulation) #
```

4. Type **header-rules** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router) # header-rules
ACMEPACKET(sip-header-rules) #
```

5. **match-value**—Enter the value to match against the header value in SIP packets; the Oracle Communications Session Border Controller matches these against the entire SIP header value. This is where you can enter values to match using regular expression values; your entries can contain Boolean operators.

To set a match value in the SIP element rules configuration:

6. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

7. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

8. Type **sip-manipulation** and press Enter.

```
ACMEPACKET(session-router) # sip-manipulation
ACMEPACKET(sip-manipulation) #
```

9. Type **header-rules** and press Enter.

```
ACMEPACKET(session-router)# header-rules
ACMEPACKET(sip-header-rules)#
```

10. Type **element-rule** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(sip-header-rules)# element-rules
ACMEPACKET(sip-element-rules)#
```

11. **match-value**—Enter the value to match against the header value in SIP packets; the Oracle Communications Session Border Controller matches these against the value of the parameter/element. This is where you can enter values to match using regular expression values, or stored pattern matches; your entries can contain Boolean operators.

### Setting the Response Code Block

To enable the SIP HMR enhancements, you need to set an option in SIP interface configuration that keeps the Oracle Communications Session Border Controller from sending the response you designate.

Note that this example sets the `dropResponse` option to 699, where 699 is an arbitrary code used to later match the HMR.

To enable SIP response blocking for a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

If you are adding support for this feature to a pre-existing SIP interface, then you must select (using the ACLI **select** command) the configuration that you want to edit.

4. **options**—Set the options parameter by typing **options**, a Space, the option name **dropResponse** with a plus sign in front of it, type the equal sign and the code(s) or range(s) you want blocked. If there is more than one, separate your entries with a colon. Then press Enter.

```
ACMEPACKET(sip-interface)# options +dropResponse=699
```

**If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.**

5. Save and activate your configuration.

### Configuring MIME Support

The **find-replace-all** action has been added to the header rules. Element rules support the **find-replace-all** action and the **mime** type.

To set the header rule with the find-replace-all action:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```



3. Type **sip-manipulation** and press Enter.

```
ACMEPACKET(session-router)# sip-manipulation
ACMEPACKET(sip-manipulation)#
```

4. Type **header-rules** and press Enter.

```
ACMEPACKET(sip-manipulation)# header-rules
ACMEPACKET(sip-header-rules)#
```

5. **action**—Set the action parameter to **find-replace-all** if you want to enable SIP HMR MIME manipulation.

6. Save and activate your configuration.

To set the element rule with the find-replace-all action and MIME type:

7. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

8. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

9. Type **sip-manipulation** and press Enter.

```
ACMEPACKET(session-router)# sip-manipulation
ACMEPACKET(sip-manipulation)#
```

10. Type **header-rules** and press Enter.

```
ACMEPACKET(sip-manipulation)# header-rules
ACMEPACKET(sip-header-rules)#
```

11. Type **element-rules** and press Enter.

```
ACMEPACKET(sip-header-rules)# element-rules
```

12. ACMEPACKET(sip-element-rules)#

13. **action**—Set the action parameter to **find-replace-all** if you want to enable SIP HMR MIME manipulation.

14. **type**—Set the type parameter to **mime** if you want to enable SIP HMR MIME manipulation.

15. Save and activate your configuration.

## Testing Pattern Rules

The Oracle Communications Session Border Controller it yields the results you require. This command is useful for testing the regex values that you devise because it will tell you whether that value is valid or not.

This command is called **test-pattern-rule**.

To test a pattern rule:

1. Type **test-pattern-rule** and press Enter.

```
ACMEPACKET# test-pattern-rule
ACMEPACKET(test-pattern-rule)#
```

2. **expression**—Enter the regular expression that you want to test. If there is a match, then the Oracle Communications Session Border Controller will inform you of it; you will also be informed if there is no match.

The string against which the Oracle Communications Session Border Controller is matching is not the string parameter that you can use for this command; it is the string value of the regular expression you entered.

```
ACMEPACKET(test-pattern-rule)# expression '*.*;tgid=(.+) .*'
```

3. **string**—Enter the string against which you want to compare the regular expression.

```
ACMEPACKET(test-pattern-rule)# string sip:+17024260002@KCMGGWC;user=phone SIP/2.0;tgid=Trunk1  
expression made 3 matches against string
```

4. **show**—Use the **show** command within test-pattern-rules to view the test pattern that you entered, whether there was a match, and the number of matches.

```
ACMEPACKET(test-pattern-rule)# show  
Pattern Rule:  
Expression : .*(;tgid=(.+)).*  
String      : sip:+17024260002@KCMGGWC;user=phone SIP/2.0;tgid=Trunk1  
Matched     : TRUE  
Matches:  
$0 sip:+17024260002@KCMGGWC;user=phone SIP/2.0;tgid=Trunk1  
$1 ;tgid=Trunk1  
$2 Trunk1
```

### Configuring SIP HMR Sets

This section shows you how to configure your multiple SIP HMR sets.

Remember to run the ACLI **verify-config** command prior to activating your configuration so the Oracle Communications Session Border Controller can detect any invalid or circular references.

To set the parameters enabling the use of SIP HMR sets:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal  
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router  
ACMEPACKET(session-router)#
```

3. Type **sip-manipulation** and press Enter.

```
ACMEPACKET(session-router)# sip-manipulation  
ACMEPACKET(sip-manipulation)#
```

4. Type **header-rules** and press Enter.

```
ACMEPACKET(session-router)# header-rules  
ACMEPACKET(sip-header-rules)#
```

5. Type **element-rule** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(sip-header-rules)# element-rules  
ACMEPACKET(sip-element-rules)#
```

6. **action**—Enter the sip-manip value to enable use this rule for a SIP HMR set. This value then invoke the rule identified in the new-value parameter.
7. **new-value**—To use SIP HMR sets, enter the name of the manipulation rule you want invoked for the set.
8. Save and activate your configuration.

### Configuration Examples

This section shows you several configuration examples for HMR. This section shows the configuration for the various rules that the Oracle Communications Session Border Controller applied, and sample results of the manipulation. These examples present configurations as an entire list of fields and settings for each ruleset, nested header rules and nested element rules. If a field does not have any operation within the set, the field is shown with the setting at the default or blank.

### Example 1 Removing Headers

For this manipulation rule, the Oracle Communications Session Border Controller removes the Custom header if it matches the pattern rule. It stores the defined pattern rule for the goodBye header. Finally, it removes the goodBye header if the pattern rule from above is a match.

This is a sample of the configuration:

```

sip-manipulation
  name
  header-rule
    name
    header-name
    action
    comparison-type
    match-value
    msg-type
    new-value
    methods
  header-rule
    name
    header-name
    action
comparison-type
  boolean
  match-value
  msg-type
  new-value
  methods
header-rule
action
  name
  delete
  comparison-type
  match-value
  msg-type
  new-value
  methods
removeHeader
removeCustom
Custom
delete
boolean
^This is my.*
request
INVITE
goodByeHeader
Goodbye
store
boolean
^Remove (.+)
request
INVITE
goodBye
pattern-rule
$goodByeHeader
request
INVITE

```

This is a sample of the result:

```

Request-Line: INVITE sip:service@192.168.200.60:5060;tgid=123 SIP/2.0
  Message Header
  Via: SIP/2.0/UDP
192.168.200.61:5060;branch=z9hG4bK0g639r10fgc0aakk26s1.1
  From: sipp <sip:sipp@192.168.1.60:5060>;tag=SDclrm601-1
  To: sut <sip:service@192.168.1.61:5060>
  Call-ID: SDclrm601-d01673bcacfcc112c053d95971330335-06a3gu0
  CSeq: 1 INVITE
  Contact: <sip:sipp@192.168.200.61:5060;transport=udp>
  Display: sipp <sip:user@192.168.1.60:5060;up=abc>;hp=123
  Params: sipp <sip:sipp1@192.168.1.60:5060>
  Params: sipp <sip:sipp2@192.168.1.60:5060>
  Edit: disp <sip:user@192.168.1.60:5060>
  Max-Forwards: 69
  Subject: Performance Test
  Content-Type: application/sdp
  Content-Length: 140

```

### Example 2 Manipulating the Request URI

For this manipulation rules, the Oracle Communications Session Border Controller stores the URI parameter tgid in the Request URI. Then if the pattern rule matches, it adds a new header (x-customer-profile) with the a new header value tgid to the URI parameter in the request URI.

This is a sample of the configuration:

## SIP Signaling Services

```
sip-manipulation
  name                               CustomerTgid
  header-rule
    name                               ruriRegex
    header-name                         request-uri
    action                               store
    comparison-type                     pattern-rule
    match-value
    msg-type                             request
new-value
  methods                               INVITE
  element-rule
    name                               tgidParam
    parameter-name                      tgid
    type                                 uri-param
    action                               store
    match-val-type                       any
    comparison-type                     pattern-rule
    match-value
    new-value
header-rule
  name                               addCustomer
  header-name                         X-Customer-Profile
  action                               add
  comparison-type                     pattern-rule
  match-value                         $ruriRegex.$tgidParam
  msg-type                             request
  new-value                           $ruriRegex.$tgidParam.$0
  methods                               INVITE
header-rule
  name                               delTgid
  header-name                         request-uri
  action                               manipulate
  comparison-type                     pattern-rule
  match-value                         $ruriRegex.$tgidParam
  msg-type                             request
  new-value
  methods                               INVITE
  element-rule
    name                               tgidParam
    parameter-name                      tgid
    type                                 uri-param
    action                               delete-element
    match-val-type                       any
    comparison-type                     case-sensitive
    match-value                         $ruriRegex.$tgidParam.
$0
  new-value
```

This is a sample of the result:

```
Request-Line: INVITE sip:service@192.168.200.60:5060 SIP/2.0
Message Header
Via: SIP/2.0/UDP 192.168.200.61:5060;branch=z9hG4bK0g6plv3088h03acgh6c1.1
  From: sipp <sip:sipp@192.168.1.60:5060>;tag=SDclrg601-1
  To: sut <sip:service@192.168.1.61:5060>
  Call-ID: SDclrg601-f125d8b0ec7985c378b04cab9f91cc09-06a3gu0
  CSeq: 1 INVITE
  Contact: <sip:sipp@192.168.200.61:5060;transport=udp>
  Goodbye: Remove Me
  Custom: This is my custom header
  Display: sipp <sip:user@192.168.1.60:5060;up=abc>;hp=123
Params: sipp <sip:sipp1@192.168.1.60:5060>
Params: sipp <sip:sipp2@192.168.1.60:5060>
```

```

Edit: disp <sip:user@192.168.1.60:5060>
Max-Forwards: 69
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 140
X-Customer-Profile: 123

```

### Example 3 Manipulating a Header

For this manipulation rule, the Oracle Communications Session Border Controller stores the pattern matches for the Custom header, and replaces the value of the Custom header with a combination of the stored matches and new content.

This is a sample of the configuration:

```

sip-manipulation
  name
  header-rule
    name
    header-name
    action
    comparison-type
    match-value
    msg-type
    new-value
    methods
header-rule
  name
  header-name
  action
  comparison-type
  match-value
  msg-type
  new-value
methods
  element-rule
    name
    parameter-name
    type
    action
    match-val-type
    comparison-type
    match-value
new-value
  $customSearch.$1+edited+$customSearch.$3

```

This is a sample of the result:

```

Request-Line: INVITE sip:service@192.168.200.60:5060;tgid=123 SIP/2.0
  Message Header
  Via: SIP/2.0/UDP
192.168.200.61:5060;branch=z9hG4bK20q2s820boghbacgs6o0.1
  From: sipp <sip:sipp@192.168.1.60:5060>;tag=SDelra601-1
  To: sut <sip:service@192.168.1.61:5060>
  Call-ID: SDelra601-4bb668e7ec9eeb92c783c78fd5b26586-06a3gu0
  CSeq: 1 INVITE
  Contact: <sip:sipp@192.168.200.61:5060;transport=udp>
  Goodbye: Remove Me
  Custom: This is my edited header
  Display: sipp <sip:user@192.168.1.60:5060;up=abc>;hp=123
  Params: sipp <sip:sipp1@192.168.1.60:5060>
  Params: sipp <sip:sipp2@192.168.1.60:5060>
  Edit: disp <sip:user@192.168.1.60:5060>
  Max-Forwards: 69
  Subject: Performance Test

```

```
Content-Type: application/sdp
Content-Length: 140
```

### Example 4 Storing and Using URI Parameters

For this manipulation rule, the Oracle Communications Session Border Controller stores the value of the URI parameter tag from the From header. It also creates a new header FromTag with the header value from the stored information resulting from the first rule.

This is a sample of the configuration:

```
sip-manipulation
  name                               storeElemParam
  header-rule
    name                               Frohmr
    header-name                         From
    action                               store
    comparison-type                     case-sensitive
    match-value
    msg-type                             request
    new-value
    methods                              INVITE
    element-rule
      name                               elementRule
      parameter-name                     tag
      type                               uri-param
      action                               store
      match-val-type                     any
      comparison-type                   case-sensitive
      match-value
      new-value
header-rule
  name                               newHeader
  header-name                         FromTag
  action                               add
  comparison-type                     pattern-rule
  match-value                         $FromHR.$elementRule
  msg-type                             any
  new-value                             $FromHR.$elementRule.$0
  methods
```

This is a sample of the result:

```
Request-Line: INVITE sip:service@192.168.200.60:5060;tgid=123 SIP/2.0
  Message Header
  Via: SIP/2.0/UDP
192.168.200.61:5060;branch=z9hG4bK4oda2e2050ih7acgh6c1.1
  From: sipp <sip:sipp@192.168.1.60:5060>;tag=SDflre601-1
  To: sut <sip:service@192.168.1.61:5060>
  Call-ID: SDflre601-f85059e74e1b443499587dd2dee504c2-06a3gu0
  CSeq: 1 INVITE
  Contact: <sip:sipp@192.168.200.61:5060;transport=udp>
  Goodbye: Remove Me
  Custom: This is my custom header
  Display: sipp <sip:user@192.168.1.60:5060;up=abc>;hp=123
  Params: sipp <sip:sipp1@192.168.1.60:5060>
  Params: sipp <sip:sipp2@192.168.1.60:5060>
  Edit: disp <sip:user@192.168.1.60:5060>
  Max-Forwards: 69
  Subject: Performance Test
  Content-Type: application/sdp
Content-Length: 140
  FromTag: 1
```

### Example 5 Manipulating Display Names

For this manipulation rule, the Oracle Communications Session Border Controller stores the display name from the Display header. It replaces the two middle characters of the original display name with a new string. Then it also replaces the From header's display name with "abc 123" if it matches sipp.

This is a sample of the configuration:

```

sip-manipulation
  name
  header-rule
    name
    header-name
    action
    comparison-type
    match-value
    msg-type
    new-value
    methods
    element-rule
      name
      parameter-name
      type
      action
      match-val-type
      comparison-type
      match-value
      new-value
  comparison-type
  pattern-rule
  match-value
  new-value
header-rule
  name
  header-name
  action
  comparison-type
  match-value
  msg-type
  new-value
  methods
  element-rule
    name
    parameter-name
    type
    action
    match-val-type
    comparison-type
    match-value
  $displayName
  new-value
  $displayName.$1+lur+$storeDisplay.$displayName.$3
header-rule
  name
  header-name
  action
  comparison-type
  match-value
  msg-type
  new-value
  methods
  element-rule
    name
    parameter-name
    type
    action
    match-val-type
    comparison-type
    modFrom
    From
    manipulate
    pattern-rule
    request
    INVITE
    fromDisplay
    uri-display
    replace
    any
    pattern-rule

```

match-value	sipp
new-value	"\"abc 123\" "

This is a sample of the result:

```
Request-Line: INVITE sip:service@192.168.200.60:5060;tgid=123 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP
192.168.200.61:5060;branch=z9hG4bK681kot109gp04acgs6o0.1
    From: "abc 123" <sip:sipp@192.168.1.60:5060>;tag=SD79ra601-1
    To: sut <sip:service@192.168.1.61:5060>
    Call-ID: SD79ra601-a487f1259e2370d3dbb558c742d3f8c4-06a3gu0
    CSeq: 1 INVITE
    Contact: <sip:sipp@192.168.200.61:5060;transport=udp>
    Goodbye: Remove Me
    Custom: This is my custom header
    Display: slurp <sip:user@192.168.1.60:5060;up=abc>;hp=123
    Params: sipp <sip:sipp1@192.168.1.60:5060>
    Params: sipp <sip:sipp2@192.168.1.60:5060>
    Edit: disp <sip:user@192.168.1.60:5060>
    Max-Forwards: 69
    Subject: Performance Test
    Content-Type: application/sdp
    Content-Length: 140
```

### Example 6 Manipulating Element Parameters

For this more complex manipulation rule, the Oracle Communications Session Border Controller:

- From the Display header, stores the display name, user name, URI parameter up, and header parameter hp
- Adds the header parameter display to the Params header, with the stored value of the display name from the first step
- Add the URI parameter user to the Params header, with the stored value of the display name from the first step
- If the URI parameter match succeeds in the first step, replaces the URI parameter up with the Display header with the value def
- If the header parameter match succeeds in the first step, deletes the header parameter hp from the Display header

This is a sample of the configuration:

```
sip-manipulation
  name
  header-rule
    name
    header-name
    action
    comparison-type
    match-value
    msg-type
    new-value
    methods
    element-rule
      name
      parameter-name
      type
      action
      match-val-type
      comparison-type
      match-value
      new-value
  element-rule
    name
    parameter-name
    type
    action
```



	match-val-type	any
	comparison-type	pattern-rule
	match-value	
	new-value	
element-rule		
	name	uriParam
	parameter-name	up
	type	uri-param
	action	store
	match-val-type	any
	comparison-type	pattern-rule
	match-value	
	new-value	
element-rule		
	name	headerParam
	parameter-name	hp
	type	header-param
	action	store
	match-val-type	any
	comparison-type	pattern-rule
	match-value	
	new-value	
header-rule		
	name	EditParams
	header-name	Params
	action	manipulate
	comparison-type	case-sensitive
	match-value	
	msg-type	request
	new-value	
	methods	INVITE
	element-rule	
	name	addHeaderParam
	parameter-name	display
	type	header-param
	action	add
match-val-type	any	
	comparison-type	case-sensitive
	match-value	
	new-value	\$StoreDisplay.
\$displayName.\$0		
	element-rule	
	name	addUriParam
	parameter-name	user
	type	uri-param
	action	add
	match-val-type	any
	comparison-type	case-sensitive
	match-value	
	new-value	
\$StoreDisplay.\$userName.\$0		
	header-rule	
	name	EditDisplay
	header-name	Display
	action	manipulate
	comparison-type	case-sensitive
	match-value	
	msg-type	request
	new-value	
	methods	INVITE
	element-rule	
	name	replaceUriParam
	parameter-name	up
	type	uri-param

```

        action                replace
        match-val-type        any
        comparison-type        pattern-rule
        match-value            $StoreDisplay.$uriParam
        new-value              def
    element-rule
        name                    delHeaderParam
        parameter-name          hp
        type                    header-param
        action                  delete-element
        match-val-type          any
        comparison-type          pattern-rule
        match-value             $StoreDisplay.$headerParam
        new-value

```

This is a sample of the result:

```

Request-Line: INVITE sip:service@192.168.200.60:5060;tgid=123 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP
192.168.200.61:5060;branch=z9hG4bK7okvei0028jgdacgh6c1.1
    From: sipp <sip:sipp@192.168.1.60:5060>;tag=SD89rm601-1
    To: sut <sip:service@192.168.1.61:5060>
    Call-ID: SD89rm601-b5b746cef19d0154cb1f342cb04ec3cb-06a3gu0
    CSeq: 1 INVITE
    Contact: <sip:sipp@192.168.200.61:5060;transport=udp>
    Goodbye: Remove Me
    Custom: This is my custom header
    Display: sipp <sip:user@192.168.1.60:5060;up=def>
    Params: sipp <sip:sipp1@192.168.1.60:5060;user=user>;display=sipp
    Params: sipp <sip:sipp2@192.168.1.60:5060;user=user>;display=sipp
    Edit: disp <sip:user@192.168.1.60:5060>
    Max-Forwards: 69
    Subject: Performance Test
    Content-Type: application/sdp
    Content-Length: 140

```

### Example 7 Accessing Data from Multiple Headers of the Same Type

For this manipulation rule, the Oracle Communications Session Border Controller stores the user name from the Params header. It then adds the URI parameter c1 with the value stored from the first Params header. Finally, it adds the URI parameter c2 with the value stored from the second Params header.

This is a sample of the configuration:

```

sip-manipulation
  name                    Params
  header-rule
    name                    storeParams
    header-name              Params
    action                  store
    comparison-type          case-sensitive
    match-value
    msg-type                 request
    new-value
    methods                  INVITE
  element-rule
    name                    storeUserName
    parameter-name          user
    type                    uri-user
    action                  store
    match-val-type          any
    comparison-type          case-sensitive
    match-value
    new-value

```

```

header-rule
    name
    header-name
    action
    comparison-type
    match-value
    msg-type
    new-value
    modEdit
    Edit
    manipulate
    pattern-rule
    request

methods
    INVITE
    element-rule
        name
        parameter-name
        type
        action
        match-val-type
        comparison-type
        match-value
        new-value
        addParam1
        c1
        uri-param
        add
        any
        case-sensitive
        $storeParams[0].

$storeUserName.$0
    element-rule
        name
        parameter-name
        type
        action
        match-val-type
        comparison-type
        match-value
        new-value
        addParam2
        c2
        uri-param
        add
        any
        case-sensitive
        $storeParams[1].

$storeUserName.$0

```

This is a sample of the result:

```

Request-Line: INVITE sip:service@192.168.200.60:5060;tgid=123 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP
192.168.200.61:5060;branch=z9hG4bK9g855p30cos08acgs6o0.1
  From: sipp <sip:sipp@192.168.1.60:5060>;tag=SD99ri601-1
  To: sut <sip:service@192.168.1.61:5060>
  Call-ID: SD99ri601-6f5691f6461356f607b0737e4039caec-06a3gu0
  CSeq: 1 INVITE
  Contact: <sip:sipp@192.168.200.61:5060;transport=udp>
  Goodbye: Remove Me
  Custom: This is my custom header
  Display: sipp <sip:user@192.168.1.60:5060;up=abc>;hp=123
  Params: sipp <sip:sipp1@192.168.1.60:5060>
  Params: sipp <sip:sipp2@192.168.1.60:5060>
  Edit: disp <sip:user@192.168.1.60:5060;c1=sipp1;c2=sipp2>
  Max-Forwards: 69
  Subject: Performance Test
  Content-Type: application/sdp
  Content-Length: 140

```

### Example 8 Using Header Rule Special Characters

For this manipulation rule, the Oracle Communications Session Border Controller:

- Stores the header value of the Params header with the given pattern rule, and stores both the user name of the Params header and the URI parameter abc
- Adds the URI parameter lpu with the value stored from the previous Params header
- If any of the Params headers match the pattern rule defined in the first step, adds the URI parameter apu with the value aup
- If all of the Params headers match the pattern rule defined in the first step, adds the URI parameter apu with the value apu

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- If the first Params headers does not match the pattern rule for storing the URI parameter defined in the first step, adds the URI parameter not with the value 123
- If the first Params headers matches the pattern rule for storing the URI parameter defined in the first step, adds the URI parameter yes with the value 456

This is a sample of the configuration:

```
sip-manipulation
  name                               specialChar
  header-rule
    name                             searchParams
    header-name                       Params
    action                             store
    comparison-type                   pattern-rule
    match-value                       .*sip:(.+)*
    msg-type                           request
    new-value
    methods                             INVITE
  element-rule
    name                               userName
    parameter-name
    type                               uri-user
    action                             store
    match-val-type                     any
    comparison-type                   case-sensitive
    match-value
    new-value
  element-rule
    name                               emptyUriParam
    parameter-name                     abc
    type                               uri-param
    action                             store
    match-val-type                     any
    comparison-type                   pattern-rule
    match-value
    new-value
  header-rule
    name                               addUserLast
    header-name                       Edit
    action                             manipulate
    comparison-type                   case-sensitive
    match-value
    msg-type                           request
    new-value
    methods                             INVITE
  element-rule
    name                               lastParamUser
    parameter-name                     lpu
    type                               uri-param
    action                             add
    match-val-type                     any
    comparison-type                   case-sensitive
    match-value
    new-value $searchParams[^\].$userName.$0
  element-rule
    name                               anyParamUser
    parameter-name                     apu
    type                               uri-param
    action                             add
    match-val-type                     any
    comparison-type                   pattern-rule
    match-value                       $searchParams[~]
    new-value                           aup
  element-rule
```

```

        name                allParamUser
        parameter-name      apu
        type                 header-param
        action               add
        match-val-type       any
        comparison-type      pattern-rule
        match-value          $searchParams [*]
        new-value            apu
    element-rule
        name                notParamYes
        parameter-name      not
        type                 uri-param
        action               add
        match-val-type       any
        comparison-type      pattern-rule
        match-value          !$searchParams.
$emptyUriParam
        new-value            123
    element-rule
        name                notParamNo
        parameter-name      yes
        type                 uri-param
        action               add
        match-val-type       any
        comparison-type      pattern-rule
        match-value          $searchParams.
$emptyUriParam
        new-value            456

```

This is a sample of the result:

```

Request-Line: INVITE sip:service@192.168.200.60:5060;tgid=123 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP
192.168.200.61:5060;branch=z9hG4bK681m9t30e0qh6akgj2s1.1
    From: sipp <sip:sipp@192.168.1.60:5060>;tag=SDchrc601-1
    To: sut <sip:service@192.168.1.61:5060>
    Call-ID: SDchrc601-fcf5660a56e2131fd27f12fcbd169fe8-06a3gu0
    CSeq: 1 INVITE
    Contact: <sip:sipp@192.168.200.61:5060;transport=udp>
    Goodbye: Remove Me
    Custom: This is my custom header
    Display: sipp <sip:user@192.168.1.60:5060;up=abc>;hp=123
    Params: sipp <sip:sipp1@192.168.1.60:5060>
    Params: sipp <sip:sipp2@192.168.1.60:5060>
    Edit: disp
<sip:user@192.168.1.60:5060;lpu=sipp2;apu=aup;not=123>;apu=apu
    Max-Forwards: 69
    Subject: Performance Test
    Content-Type: application/sdp
    Content-Length: 140

```

### Example 9 Status-Line Manipulation

This section shows an HMR configuration set up for status-line manipulation.

Given that the object of this example is to drop the 183 Session Progress response when it does not have SDP, your SIP manipulation configuration needs to:

1. Search for the 183 Session Progress response
2. Determine if the identified 183 Session Progress responses contain SDP; the Oracle Communications Session Border Controller searches the 183 Session Progress responses where the content length is zero
3. If the 183 Session Progress response does not contain SDP, change its status code to 699
4. Drop all 699 responses

```

sip-manipulation
  name                               manip
  description
  header-rule
    name                               IsContentLength0
    header-name                         Content-Length
    action                               store
    comparison-type                     pattern-rule
    match-value                         0
    msg-type                             reply
    new-value
    methods
  header-rule
    name                               is183
    header-name                         @status-line
    action                               store
    comparison-type                     pattern-rule
    match-value
    msg-type                             reply
    new-value
    methods
  element-rule
    name                               is183Code
    parameter-name
    type                               status-code
    action                               store
    match-val-type                      any
    comparison-type                     pattern-rule
    match-value                         183
    new-value
  header-rule
    name                               change183
    header-name                         @status-line
    action                               manipulate
    comparison-type                     case-sensitive
    match-value
    msg-type                             reply
    new-value
    methods
  element-rule
    name                               make199
    parameter-name
    type                               status-code
    action                               replace
    match-val-type                      any
    comparison-type                     pattern-rule
    match-value                         $IsContentLength0 &
    new-value
    new-value                           199
  $is183.$is183Code
sip-interface      options dropResponse=699

```

### Example 10 Use of SIP HMR Sets

The following example shows the configuration for SIP HMR with one SIP manipulation configuration loading another SIP manipulation configuration. The goals of this configuration are to:

- Add a new header to an INVITE
- Store the user portion of the Request URI
- Remove all Route headers from the message only if the Request URI is from a specific user

```

sip-manipulation
  name                               deleteRoute
  description                         delete all Route Headers

```

```

header-rule
  name          deleteRoute
  header-name   Route
  action        delete
  comparison-type case-sensitive
  match-value
  msg-type      request
  new-value
  methods       INVITE
sip-manipulation
  name          addAndDelete
  description   Add a New header and delete Route
headers
header-rule
  name          addHeader
  header-name   New
  action        add
  comparison-type case-sensitive
  match-value
  msg-type      request
  new-value     "Some Value"
  methods       INVITE
header-rule
  name          storeRURI
  header-name   request-uri
  action        store
  comparison-type pattern-rule
  match-value
  msg-type      request
  new-value
  methods       INVITE
element-rule
  name          storeUser
  parameter-name
  type          uri-user
  action        store
  match-val-type any
  comparison-type pattern-rule
  match-value   305.*
  new-value
header-rule
  name          deleteHeader
  header-name   request-uri
  action        sip-manip
  comparison-type Boolean
  match-value   $storeRURI.$storeUser
  msg-type      request
  new-value     deleteRoute
  methods       INVITE

```

### Example 11 Use of Remote and Local Port Information

The following example shows the configuration for remote and local port information. The goals of this configuration are to:

- Add LOCAL\_PORT as a header parameter to the From header
- Add REMOTE\_PORT as a header parameter to the From header

```

sip-manipulation
  name          addOrigIp
  description
  header-rule
    name          addIpParam
    header-name   From

```

```

action                manipulate
comparison-type      case-sensitive
match-value
msg-type              request
new-value
methods               INVITE
element-rule
  name                addIpParam
  parameter-name      newParam
  type                header-param
  action              add
  match-val-type      any
  comparison-type     case-sensitive
  match-value
  new-value            $LOCAL_IP
element-rule
  name                addLocalPort
  parameter-name      lport
  type                header-param
  action              add
  match-val-type      any
  comparison-type     case-sensitive
  match-value
  new-value            $LOCAL_PORT
element-rule
  name                addRemotePort
  parameter-name      rport
  type                header-param
  action              add
  match-val-type      any
  comparison-type     case-sensitive
  match-value
  new-value            $REMOTE_PORT

```

### Example 12 Response Status Processing

Given that the object of this example is to drop the 183 Session Progress response when it does not have SDP, your SIP manipulation configuration needs to:

1. Search for the 183 Session Progress response
2. Determine if the identified 183 Session Progress responses contain SDP; the Oracle Communications Session Border Controller searches the 183 Session Progress responses where the content length is zero
3. If the 183 Session Progress response does not contain SDP, change its status code to 699
4. Drop all 699 responses

```

sip-manipulation
  name                manip
  description
  header-rule
    name              IsContentLength0
    header-name       Content-Length
    action             store
    comparison-type   pattern-rule
    match-value       0
    msg-type           reply
    new-value
    methods
  header-rule
    name              is183
    header-name       @status-line
    action             store
    comparison-type   pattern-rule
    match-value

```



```

msg-type          reply
new-value
methods
element-rule
  name            is183Code
  parameter-name
  type            status-code
  action          store
  match-val-type  any
  comparison-type pattern-rule
  match-value     183
  new-value
header-rule
  name            changel83
  header-name     @status-line
  action          manipulate
  comparison-type case-sensitive
  match-value
msg-type          reply
new-value
methods
element-rule
  name            make699
  parameter-name
  type            status-code
  action          replace
  match-val-type  any
  comparison-type pattern-rule
  match-value     $IsContentLength0 &
$sis183.$sis183Code
  new-value       699
sip-interface
  options dropResponse=699

```

The following four configuration examples are based on the this sample SIP INVITE:

```

INVITE sip:service@192.168.1.61:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.60:5060;branch=z9hG4bK-1-0
From: sipp <sip:sipp@192.168.1.60:5060>;tag=1
To: sut <sip:service@192.168.1.61:5060>
Call-ID: 1-15554@192.168.1.60
CSeq: 1 INVITE
Contact: <sip:sipp@192.168.1.60:5060;user=phone>
Max-Forwards: 70
Content-Type: multipart/mixed;boundary=boundary
Content-Length: 466
--boundary
Content-Type: application/sdp
v=0
o=user1 53655765 2353687637 IN IP4 192.168.1.60
s=-
c=IN IP4 192.168.1.60
t=0 0
m=audio 12345 RTP/AVP 18
a=rtpmap:8 G729/8000/1
a=fmtp:18 annexb=no
a=sendrecv
a=ptime:20
a=maxptime:200
--boundary
Content-Type: application/sdp
v=0
o=user1 53655765 2353687637 IN IP4 192.168.1.60
s=-

```

```
c=IN IP4 192.168.1.60
t=0 0
m=video 12345 RTP/AVP 34
a=rtpmap:34 H263a/90000
a=ptime:30
--boundary--
```

### Example 13 Remove a Line from SDP

In this example, the SIP manipulation is configured to remove all p-time attributes from the SDP.

```
sip-manipulation
  name                removePtimeFromBody
  description         removes ptime attribute from all bodies
  header-rule
    name              CTypeManp
    header-name       Content-Type
    action            manipulate
    comparison-type   case-sensitive
    match-value
    msg-type          request
    new-value
    methods           INVITE
  element-rule
    name              remPtime
    parameter-name    application/sdp
    type              mime
    action            find-replace-all
    match-val-type    any
    comparison-type   case-sensitive
    match-value       a=ptime:[0-9]{1,2}(\n|
\r\n)
    new-value
```

The result of manipulating the original SIP INVITE (shown above) with the configured SIP manipulation is:

```
INVITE sip:service@192.168.1.61:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.60:5060;branch=z9hG4bK-1-0
From: sipp <sip:sipp@192.168.1.60:5060>;tag=1
To: sut <sip:service@192.168.1.61:5060>
Call-ID: 1-15554@192.168.1.60
CSeq: 1 INVITE
Contact: <sip:sipp@192.168.1.60:5060;user=phone>
Max-Forwards: 70
Content-Type: multipart/mixed;boundary=boundary
Content-Length: 466
--boundary
Content-Type: application/sdp
v=0
o=user1 53655765 2353687637 IN IP4 192.168.1.60
s=-
c=IN IP4 192.168.1.60
t=0 0
m=audio 12345 RTP/AVP 18
a=rtpmap:18 G729/8000/1
a=fmtp:18 annexb=no
a=sendrecv
a=maxptime:200
--boundary
Content-Type: application/sdp
v=0
o=user1 53655765 2353687637 IN IP4 192.168.1.60
s=-
c=IN IP4 192.168.1.60
t=0 0
```

```
m=video 12345 RTP/AVP 34
a=rtpmap:34 H263a/90000
--boundary-
```

### Example 14 Back Reference Syntax

In this sample of back-reference syntax use, the goal is to change the To user. The SIP manipulation would be configured like the following:

```

sip-manipulation
  name                changeToUser
  description          change user in the To header
  header-rule
    name              ChangeHeader
    header-name       To
    action             manipulate
    comparison-type   case-sensitive
    match-value
    msg-type          request
    new-value
    methods           INVITE
  element-rule
    name              replaceValue
    parameter-name
    type              header-value
    action            replace
    match-val-type    any
    comparison-type   pattern-rule
    match-value       (.+)(service)(.+)
    new-value         $1+Bob+$3

```

The result of manipulating the original SIP INVITE (shown above) with the configured SIP manipulation is:

```

INVITE sip:service@192.168.1.61:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.60:5060;branch=z9hG4bK-1-0
From: sipp <sip:sipp@192.168.1.60:5060>;tag=1
To: sut <sip:Bob@192.168.1.61:5060>
Call-ID: 1-15554@192.168.1.60
CSeq: 1 INVITE
Contact: <sip:sipp@192.168.1.60:5060;user=phone>
Max-Forwards: 70
Content-Type: multipart/mixed;boundary=boundary
Content-Length: 466
...
...
...

```

### Example 15 Change and Remove Lines from SDP

In this sample of changing and removing lines from the SDP, the goal is to convert the G.729 codec to G.729a. The SIP manipulation would be configured like the following:

```

sip-manipulation
  name                std2prop-codec-name
  description          rule to translate standard to
  proprietary codec name
  header-rule
    name              CTypeManp
    header-name       Content-Type
    action            manipulate
    comparison-type   case-sensitive
    match-value
    msg-type          any
    new-value
    methods

```

```

element-rule
  name                g729-annexb-no-std2prop
  parameter-name      application/sdp
  type                mime
  action              find-replace-all
  match-val-type      any
  comparison-type     case-sensitive
  match-value         a=rtpmap:[0-9]{1,3}
(G729/8000/1\r\na=fmtp:[0-9]{1,3} annexb=no) [[:1:]]
  new-value           G729a/8000/1

```

The result of manipulating the original SIP INVITE (shown above) with the configured SIP manipulation is:

```

INVITE sip:service@192.168.1.61:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.60:5060;branch=z9hG4bK-1-0
From: sipp <sip:sipp@192.168.1.60:5060>;tag=1
To: sut <sip:service@192.168.1.61:5060>
Call-ID: 1-15554@192.168.1.60
CSeq: 1 INVITE
Contact: <sip:sipp@192.168.1.60:5060;user=phone>
Max-Forwards: 70
Content-Type: multipart/mixed;boundary=boundary
Content-Length: 466
--boundary
Content-Type: application/sdp
v=0
o=user1 53655765 2353687637 IN IP4 192.168.1.60
s=-
c=IN IP4 192.168.1.60
t=0 0
m=audio 12345 RTP/AVP 8
a=rtpmap:18 G729a/8000/1
a=sendrecv
a=maxptime:200
--boundary
Content-Type: application/sdp
v=0
o=user1 53655765 2353687637 IN IP4 192.168.1.60
s=-
c=IN IP4 192.168.1.60
t=0 0
m=video 12345 RTP/AVP 34
a=rtpmap:34 H263a/90000
--boundary-

```

### Example 16 Change and Add New Lines to the SDP

In this sample of changing and adding lines from the SDP, the goal is to convert non-standard codec H.263a to H.263. The SIP manipulation would be configured like the following:

```

sip-manipulation
  name                prop2std-codec-name
  description         rule to translate proprietary to
standard codec name
  header-rule
    name              CodecManp
    header-name       Content-Type
    action            manipulate
    comparison-type   case-sensitive
    match-value
    msg-type          any
    new-value
    methods
    element-rule
      name            H263a-prop2std

```

parameter-name	application/sdp
type	mime
action	find-replace-all
match-val-type	any
comparison-type	case-sensitive
match-value	a=rtpmap: ([0-9] {1,3})
H263a/.*\r\n	
new-value	a=rtpmap:+\$1+"
H263/90000"+\$CRLF+a=fmtp:+\$1+" QCIF=4"+\$CRLF	

The result of manipulating the original SIP INVITE (shown above) with the configured SIP manipulation is:

```
INVITE sip:service@192.168.1.61:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.60:5060;branch=z9hG4bK-1-0
From: sipp <sip:sipp@192.168.1.60:5060>;tag=1
To: sut <sip:service@192.168.1.61:5060>
Call-ID: 1-15554@192.168.1.60
CSeq: 1 INVITE
Contact: <sip:sipp@192.168.1.60:5060;user=phone>
Max-Forwards: 70
Content-Type: multipart/mixed;boundary=boundary
Content-Length: 466
--boundary
Content-Type: application/sdp
v=0
o=user1 53655765 2353687637 IN IP4 192.168.1.60
s=-
c=IN IP4 192.168.1.60
t=0 0
m=audio 12345 RTP/AVP 8
a=rtpmap:18 G729/8000/1
a=fmtp:18 annexb=no
a=sendrecv
a=maxptime:200
--boundary
Content-Type: application/sdp
v=0
o=user1 53655765 2353687637 IN IP4 192.168.1.60
s=-
c=IN IP4 192.168.1.60
t=0 0
m=video 12345 RTP/AVP 34
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=4
--boundary-
```

## Dialog-Matching Header Manipulation

The most common headers to manipulate using HMR are the To-URI and From-URI. Along with the to-tag, from-tag, and Call-ID values, these are also all headers that represent dialog-specific information that must match the UAC and UAS to be considered part of the same dialog. If these parameters are modified through HMR, the results can be that the UAC or UAS rejects messages.

While it is possible to ensure that dialog parameters match correctly using regular HMR, this feature offers a simpler and less error-prone method of doing so.

In addition, this section describes the addition of built-in SIP manipulations defined by Oracle best practices, and a new method of testing your SIP manipulations.

### About Dialog-Matching Header Manipulations

The goal of this feature is to maintain proper dialog-matching through manipulation of dialog-specific information using HMR. Two fundamental challenges arise when looking at the issue of correctly parameters manipulating dialog-matching:

- Inbound HMR
- Outbound HMR

The new setting **out-of-dialog** (for the **msg-type** parameter) addresses these challenges by offering an intelligent more of dialog matching of messages for inbound and outbound HMR requests. This is a **msg-type** parameter, meaning that it becomes matching criteria for operations performed against a message. If you also specify methods (such as REGISTER) as matching criteria, then the rule is further limited to the designated method.

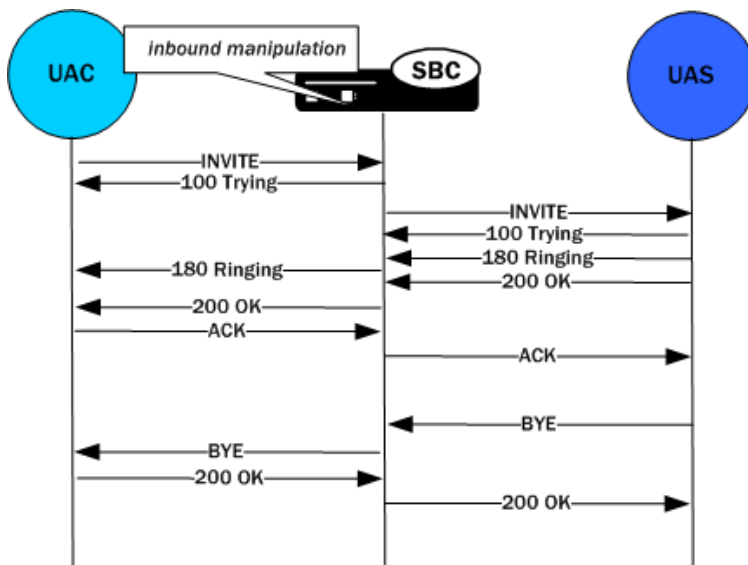
For both inbound and outbound manipulations, using the **out-of-dialog** setting means the message must be a request without a to-tag in order to perform the manipulation.

#### Inbound HMR Challenge

Since inbound manipulations take place before the message reaches the core of Oracle Communications Session Border Controller SIP processing, the SIP proxy takes the manipulated header as what was directly received from the client. This can cause problems for requests leaving the Oracle Communications Session Border Controller for the UAC because the dialog will not match the initial request sent.

So the unmodified header must be cached because for any subsequent request (as in the case of a BYE originating from the terminator; see the diagram below) the Oracle Communications Session Border Controller might need to restore the original value—enabling the UAC to identify the message correctly as being part of the same dialog. For out-of-dialog requests (when the To, From, or Call-ID headers are modified) the original header will be stored in the dialog when the **msg-type out-of-dialog** is used.

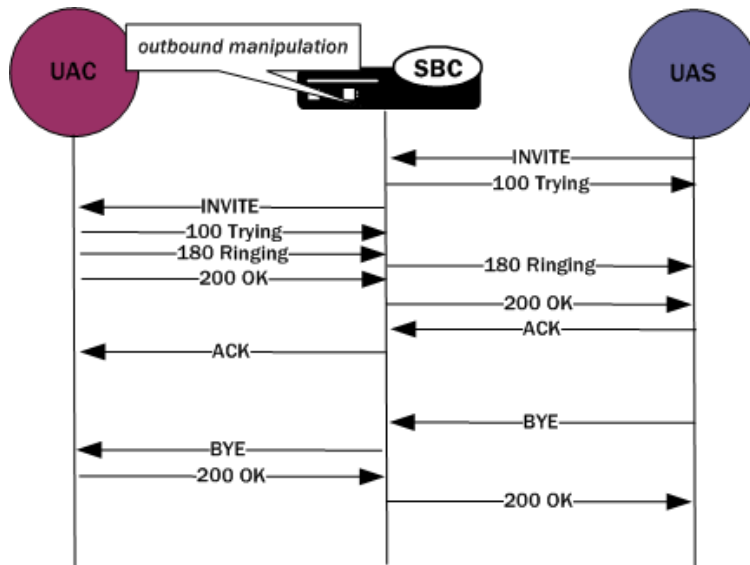
The Oracle Communications Session Border Controller performs the restoration of original headers outside of SIP manipulations. That is, there are no manipulation rules to configure for restore the header to their original context. The Oracle Communications Session Border Controller will recognize the headers have been modified, and restore them to their original state prior to sending the message out on the wire. Restoration takes place prior to outbound manipulations so that any outbound manipulation can those headers once they have been restored.



#### Outbound HMR Challenge

When you use the **out-of-dialog** setting for an outbound manipulation, the Oracle Communications Session Border Controller only executes this specific SIP header rule only if the same SIP header rule was executed against the initial dialog-creating request.

For example, if the INVITE's To header was not manipulated, it would not be correct to manipulate the To header in the BYE request. To do so would render the UAC unable to properly match the dialog. And this also means that the outbound manipulation should be carried out against a To, From, or Call-ID header in the BYE request if it was manipulated in the INVITE.



### Dialog-matching Header Manipulation Configuration

You use the **out-of-dialog** setting in the **msg-type** parameter, part of the SIP header rules configuration.

To enable dialog-matching header manipulation:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-manipulation** and press Enter.

```
ACMEPACKET(session-router)# sip-manipulation
ACMEPACKET(sip-manipulation)#
```

4. Type **mime-rules** and press Enter. If you are adding this feature to an existing configuration, then remember you must select the configuration you want to edit.

```
ACMEPACKET(sip-manipulation)# header-rules
ACMEPACKET(sip-header-rules)#
```

5. **msg-type**—Set this parameter to **out-of-dialog** to enable dialog-matching header manipulation.
6. Save your work.

### Built-In SIP Manipulations

In the course of HMR use, certain rules have become commonly used. Lengthy and complex, these rules do not include any customer-specific information and do they can be used widely. To make using them easier, they have been turned into built-in rules that you can reference in the **in-manipulationid** and **out-manipulationid** parameters that are part of the realm, session agent, and SIP interfaces configurations.

Built-in rules start with the prefix **ACME\_**, so Oracle recommends you name your own rules in a different manner to avoid conflict.

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While the number of built-in manipulation rules is expected to grow, one is supported at the present time: `ACME_NAT_TO_FROM_IP`. When performed outbound, this rule changes:

- The To-URI hostname to the logical `$TARGET_IP` and port to `$TARGET_PORT`
- The From-URI to the logical `$REPLY_IP` and port to be `$REPLY_PORT`

### Built-In SIP Manipulation Configuration

When you want to enable this feature for a realm, session agent, or SIP interface, you configure the `in-manipulationid` or `out-manipulationid` parameters with the rule.

The sample here shows this feature being applied to a session agent, but the realm and SIP interface configurations also have the same parameter you use to set up the feature.

To use built-in SIP manipulations:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type `session-router` and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type `session-agent` and press Enter.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

4. `out-manipulationid`—Enter name of the built-in rule you want to use. Remember that all built-in rules start with `ACME_`.

5. Save your work.

## Testing SIP Manipulations

You can now use a new tool that allows you to test the outcome of your SIP manipulation and header rules without sending real traffic through the Oracle Communications Session Border Controller to see if they work.

To use the tool, you enter the ACLI's `test-sip-manipulation` utility and reference the rule you want to test using this name. Then you enter a mode where you put in a SIP message entered in ASCII. You can cut and paste this message from `sipmsg.log` or from some other location. Using `<Ctrl-D>` stops the SIP message collection and parses it.

The test informs you of any parsing errors found in the SIP message. Once the message is entered, you can execute the SIP manipulation against the message. The output after this step is the modified SIP message after manipulations have been applied. You will also find a debugging option, which displays SIP manipulation logging to the screen as the manipulation takes place.

As a starting point for testing, this tool comes loaded with a default SIP message. It cannot be associated with realms, session agents, or SIP interfaces, and so it also comes with certain resolves reserved words, such as: `$LOCAL_IP`, `$TRUNK_GROUP_CONTEXT`, and `$REMOTE_PORT`. In addition, you can use your settings for testing across terminal sessions; if you choose to save your settings, everything (including the SIP message) will be saved, with the exception of the debugging option.

It is not recommended that you use this tool to add an ISUP message body.

## HMR Import-Export

---

Due to the complexity of SIP manipulations rules and the deep understanding of system syntax they require, it is often difficult to configure reliable rules. This feature provides support for importing and exporting pieces of SIP manipulation configuration in a reliable way so that they can be reused.



## Exporting

The SIP manipulation configuration contains an **export** command. When you use it, the Oracle Communications Session Border Controller sends the configuration you have selected to a designated file. The contents are the same information you see when you use the ACLI **show** command in XML format; it includes the selected configuration and any changes that have been made. Because you can only export one SIP manipulation configuration at a time, you must export each one-by-one if you need more than one.

The file name can be any you selected, and would be most useful if it were to identify its contents in some way. If the file already exists, then the export fails and informs you the file already exists. A successfully-executed export simply returns you to the system prompt.

The system writes exported files to /code/imports, a new location that will be created to avoid overlap with existing backup files. The files will carry the extension `.gz` to show that they have been compressed with gzip.

Your export data will look like this sample:

```
<?xml version='1.0' standalone='yes'?>
<sipManipulation
  name='manip'
  description=''
  lastModifiedBy='admin@console'
  lastModifiedDate='2009-10-16 14:16:29'>
  <headerRule
    headerName='Foo'
    msgType='any'
    name='headerRule'
    action='manipulate'
    cmpType='boolean'
    matchValue='$REGEX("[bB][A-Za-z]{2}")'
    newValue='foo'
    methods='INVITE'>
  </headerRule>
</sipManipulation>
```

To avoid conflict with other objects on the system, key and object ID are not included as part of the exported XML.

## Importing

Using the import command in the SIP manipulation configuration, you can import data from an exported file to a currently-selected configuration. If you have not selected a configuration into which to load the data, a new one will be created. Including the `.gz` extension, you enter the full name of the file you want imported. After it finds the file, the Oracle Communications Session Border Controller unarchives it and parses its contents. If these steps fail, the Oracle Communications Session Border Controller will alert you. If they succeed, then the configuration data loads into the object.

If you have been making changes to the configuration into which data was imported, the Oracle Communications Session Border Controller will inform you prior to importing the data so that you will not lose any of your work. This way, you will be less likely to overwrite unsaved changes.

Once the import is complete, it will be as if you entered the configuration by hand. You only need to save your work (by typing **done**) to save the SIP manipulation to the global SIP configuration. Note that if for some reason the XML is malformed or contained more than one object, the import will fail.

If you attempt to import a configuration with the same key as one that already exists, the system returns an error and prevents you from saving the imported object. In this case, you can delete the object with the same key and then carry out your import, or you can select the object with the same key and perform an import that will overwrite it with new data.

## Displaying Imports

You can display imported SIP manipulations data at the system prompt. The command lists all files in the exported files directory, and also tells you if there are none.

### Using FTP to Move Files

You can also place exported SIP manipulation configuration files on the Oracle Communications Session Border Controller using FTP. You need to use the same `/code/imports` directory to do so.

### Removing Files

Using the `delete-import` command with the name of the file you want to delete removes it from the system. Using this command, you can delete files that are no longer useful to you. Carrying out this command is final and there is no warning before you go ahead with the deletion. A failed deletion (for instance, because there is no such file) will produce an error message; a successful deletion simply returns you to the system prompt.

## Unique HMR Regex Patterns and Other Changes

---

In addition to the HMR support it offers, the Oracle Communications Session Border Controller can now be provisioned with unique regex patterns for each logical remote entity. This supplement to pre-existing HMR functionality saves you provisioning time and saves Oracle Communications Session Border Controller resources in instances when it was previously necessary to define a unique SIP manipulation per PBX for a small number of customer-specific rules.

### Manipulation Pattern Per Remote Entity

On the Oracle Communications Session Border Controller, you can configure logical remote entities (session agents, realms, and SIP interfaces) with a manipulation pattern string that the system uses as a regular expression. Then the SIP manipulation references this regular expression using the reserved word `$MANIP_PATTERN`. At runtime, the Oracle Communications Session Border Controller looks for the logical entity configured with a manipulation pattern string in this order of preference: session agent, realm, and finally SIP interface.

On finding the logical entity configured with the manipulation string, the Oracle Communications Session Border Controller dynamically determines the expression. When there is an invalid reference to a manipulation pattern, the pattern-rule expression that results will turn out to be the default expression (which is `\,+`).

When the `$MANIP_PATTERN` is used in a manipulation rule's `new-value` parameter, it resolves to an empty string, equivalent of no value. Even though this process ends with no value, it still consumes system resources. And so Oracle recommends you do not use `$MANIP_PATTERN` as a `new-value` value.

In the following example, the SIP manipulation references the regular expression from a realm configuration:

```
realm-config
  identifier                net200
  description
  addr-prefix              0.0.0.0
  network-interfaces      public:0
  ...
  manipulation-pattern     Lorem(.+)
sip-manipulation
  name                     manip
  description
  header-rules
    name                   headerRule
    header-name            Subject
    action                 manipulate
    match-value            $MANIP_PATTERN
    msg-type               request
    comparison-type       pattern-rule
    new-value              Math
    methods                INVITE
```

## Reject Action

When you use this action type and a condition matching the manipulation rule arises, the Oracle Communications Session Border Controller rejects the request (though does not drop responses) and increments a counter.

- If the **msg-type** parameter is set to **any** and the message is a response, the Oracle Communications Session Border Controller increments a counter to show the intention to reject the message—but the message will continue to be processed.
- If the **msg-type** parameter is set to **any** and the message is a request, the Oracle Communications Session Border Controller performs the rejection and increments the counter.

The **new-value** parameter is designed to supply the status code and reason phrase corresponding to the reject. You can use the following syntax to supply this information: `status-code[:reason-phrase]`. You do not have to supply the status code and reason phrase information; by default, the system uses 400:Bad Request.

If you do supply this information, then the status code must be a positive integer between 300 and 699. The Oracle Communications Session Border Controller then provides the reason phrase corresponding to the status code. And if there is no reason phrase, the system uses the one for the applicable reason class.

You can also customize a reason phrase. To do so, you enter the status code followed by a colon (:), being sure to enclose the entire entry in quotation marks (") if your reason code includes spaces.

When the Oracle Communications Session Border Controller performs the **reject** action, the current SIP manipulation stops processing and does not act on any of the rules following the **reject** rule. This course of action is true for nested SIP manipulations that might have been constructed using the **sip-manip** action type.

### Reject Action Configuration

To support the **reject** action, two parameters in the **session-router-config** allow you to set how many messages in a certain amount of time cause the Oracle Communications Session Border Controller to generate an SNMP trap.

To set the reject message number and time window:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-router** and press Enter.

```
ACMEPACKET(session-router)# session-router
ACMEPACKET(session-router-config)#
```

4. **reject-message-threshold**—Enter the minimum number of message rejections allowed in the **reject-message-window** time on the Oracle Communications Session Border Controller (when using the SIP manipulation action reject) before generating an SNMP trap. The default is 0, meaning this feature is disabled and no trap will be sent.

5. **reject-message-window**—Enter the time in seconds that defines the window for maximum message rejections allowed before generating an SNMP trap.

6. Save your work.

### About Counters

The Oracle Communications Session Border Controller tracks messages that have been flagged for rejection using the **reject** action type. In the **show sipd** display, refer to the Rejected Messages category; there is no distinction between requests and responses.

```
ACMEPACKET# show sipd
13:59:07-102
```

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SIP Status	-- Period --			----- Lifetime -----		
	Active	High	Total	Total	PerMax	High
Sessions	0	0	0	0	0	0
Subscriptions	0	0	0	0	0	0
Dialogs	0	0	0	0	0	0
CallID Map	0	0	0	0	0	0
Rejections	-	-	0	0	0	
ReINVITES	-	-	0	0	0	
Media Sessions	0	0	0	0	0	0
Media Pending	0	0	0	0	0	0
Client Trans	0	0	0	0	0	0
Server Trans	0	0	0	0	0	0
Resp Contexts	0	0	0	0	0	0
Saved Contexts	0	0	0	0	0	0
Sockets	0	0	0	0	0	0
Req Dropped	-	-	0	0	0	
DNS Trans	0	0	0	0	0	0
DNS Sockets	0	0	0	0	0	0
DNS Results	0	0	0	0	0	0
Rejected Msgs	0	0	0	0	0	0

Session Rate = 0.0  
Load Rate = 0.0  
Remaining Connections = 20000 (max 20000)

## SNMP Support

The Oracle Communications Session Border Controller provides SNMP support for the Rejected Messages data, so you can access this information externally. The new MIB objects are:

```
apSysRejectedMessages      OBJECT-TYPE
    SYNTAX                   Counter32
    MAX-ACCESS               read-only
    STATUS                   current
    DESCRIPTION
        "Number of messages rejected by the SD due to matching
criteria."
    ::= { apSysMgmtMIBGeneralObjects 18 }
apSysMgmtRejectedMesagesThresholdExeededTrap      NOTIFICATION-TYPE
    OBJECTS                   { apSysRejectedMessages }
    STATUS                   current
    DESCRIPTION
        " The trap will be generated when the number of rejected messages
exceed the configured threshold within the configured window."
    ::= { apSystemManagementMonitors 57 }
apSysMgmtRejectedMessagesGroup      OBJECT-GROUP
    OBJECTS {
        apSysRejectedMessages
    }
    STATUS                   current
    DESCRIPTION
        "Objects to track the number of messages rejected by the SD."
    ::= { apSystemManagementGroups 18 }
apSysMgmtRejectedMessagesNotificationsGroup      NOTIFICATION-GROUP
    NOTIFICATIONS {
        apSysMgmtRejectedMesagesThresholdExeededTrap
    }
    STATUS                   current
    DESCRIPTION
        "Traps used for notification of rejected messages"
    ::= { apSystemManagementNotificationsGroups 26 }
apSmgmtRejectedMessagesCap
    AGENT-CAPABILITIES
    PRODUCT-RELEASE         "Acme Packet SD"
    STATUS                   current
```

```

DESCRIPTION      "Acme Packet Agent Capability for enterprise
                  system management MIB."
SUPPORTS         APSYSMGMT-MIB
INCLUDES         {
                  apSysMgmtRejectedMessagesGroup,
                  apSysMgmtRejectedMessagesNotificationsGroup
                  }
 ::= { apSmgmtMibCapabilities 37 }

```

## Log Action

When you use this action type and a condition matching the manipulation rule arises, the Oracle Communications Session Border Controller logs information about the current message to a separate log file. This log files will be located on the same core in which the SIP manipulation occurred. On the core where sipt runs, a logfile called `matched.log` will appear when this action type is executed.

The `matched.log` file contains a timestamp, received and sent Oracle Communications Session Border Controller network interface, sent or received IP address:port information, and the peer IP address:port information. It also specifies the rule that triggered the log action in this syntax: `rule-type[rule:name]`. The request URI, Contact header, To Header, and From header are also present.

```

-----
Apr 17 14:17:54.526 On [0:0]192.168.1.84:5060 sent to 192.168.1.60:5060
element-rule[checkRURIPort]
INVITE sip:service@192.168.1.84:5060 SIP/2.0
From: sipp <sip:+2125551212@192.168.1.60:5060>;tag=3035SIPpTag001
To: sut <sip:service@192.168.1.84>
Contact: sip:sipp@192.168.1.60:5060

```

## Changes to Storing Pattern Rule Values

Release S-C6.2.0 introduces changes to the framework for storing regular expression results within manipulation rules, altering the way the **store** action works. These changes are beneficial to performance.

In previous releases, when the **store** action is used, the Oracle Communications Session Border Controller stores all values matching the regular expression defined in the **match-value** parameter for all headers. At runtime, the system evaluates all stored values to find the correct index.

Now, you no longer need to specify the **store** action. The simple fact of referencing another rule tells the system it must store a value. When SIP manipulation is used, the system first checks to see if any values require storing. The **add** action is an exception to this process; storing happens after a header is added.

When referring to a rule, that rule still needs to have a regular expression defined in the `match-value` and the comparison type set to `pattern-rule`; else the default expression will be used.

## Removal of Restrictions

The following restrictions related to HMR have been removed in Release S-C6.2.0:

- The action **find-replace-all** now executes all element rules. Previously, no child rules were executed.
- The action **sip-manip** now executes existing all element rules. Previously, no child rules were executed.
- The action **store** now executes existing all element rules. Previously, only child rules with the **store** action were executed.
- The action **add** now executes existing all element rules. Previously, only child rules with the **add** action were executed.

## Name Restrictions for Manipulation Rules

Historically, you have been allowed to configure any value for the name parameter within a manipulation rule. This method of naming caused confusion when referencing rules, so now manipulation rules name must follow a specific syntax. They must match the expression `^[[:alpha:]]+[[:alnum:]]_+$` and contain at least one lower case letter.

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In other words, the name must:

- Start with a letter, and then it can contain any number of letters, numbers, or underscores
- Contain at least one lower case letter

All pre-existing configurations will continue to function normally. If you want to change a manipulation rule, however, you are required to change its name if it does not follow the new format.

The CLI **verify-config** command warns you if the system has loaded a configuration containing illegal naming syntax.

Please note that the software allows you to make changes to HMRs, including configuring new functionality to existing rules, as long as you do not change the rule name. This results in an important consideration surrounding HMRs with hyphens in previously configured rule names.

- You can reference stored values in new value names. (Recall that stored values may be rule names.)
- You can perform subtraction in new value names.

If you use a rule names with hyphens within the REGEX of new value names, the system cannot determine whether the hyphen is part of the rule name or is intended to invoke subtraction within the REGEX. For this reason, you need to use great care with legacy HMR naming that includes hyphens.

As a general rule, create new rule names that follow the new rule naming guidelines if you intend to use new functionality in those rules.

## New Value Restrictions

To simplify configuration and remove possible ambiguity, the use of boolean and equality operators (**=**, **<=**, **<**, etc.) for **new-value** parameter values has been banned. Since there was no specific functionality tied to their use, their ceasing to be use will have no impact to normal SIP manipulation operations.

## Header Manipulation Rules for SDP

---

The Oracle Communications Session Border Controller supports SIP header and parameter manipulation rules for four types of SIP message contents:

- headers
- elements within headers
- ASCII-encoded Multipurpose Internet Mail Extensions (MIME) bodies
- binary-encoded MIME ISDN User Part (ISUP) bodies

While Session Description Protocol (SDP) offers and answers can be manipulated in a fashion similar to ASCII-encoded MIME, such manipulation is primitive in that it lacks the ability to operate at the SDP session- and media-levels.

In addition, the system supports a variant of Header Manipulation Rules (HMR) operating on ASCII-encoded SDP bodies, with specific element types for descriptors at both the session-level and media-level, and the ability to apply similar logic to SDP message parts as is done for SIP header elements.

The configuration object, **mime-sdp-rules**, under **sip-manipulation** specifically addresses the manipulation of SDP parts in SIP messages. Just as existing header-rules are used to manipulate specific headers of a SIP message, **mime-sdp-rules** will be used to manipulate the SDP specific mime-attachment of a SIP message.

## Platform Support

HMR for SDP is available on the following platforms: Acme Packet 38x0 and Acme Packet 4500.

## SDP Manipulation

**mime-sdp-rules** function in a similar fashion as header-rules. They provide

- parameters used to match against specific SIP methods and/or message types
- parameters used to match and manipulate all or specified parts of an SDP offer or answer
- a means of comparing search strings or expressions against the entire SDP
- different action types to allow varying forms of manipulation

Since only a single SDP can exist within a SIP message, users need not specify a content-type parameter as is necessary for a mime-rule. A mime-sdp-rule operates on the single SDP within the SIP message. If no SDP exists with the message, one can be added. If the message already contains a mime attachment, adding SDP results in a multipart message.

All manipulations performed against all or parts of the SDP are treated as UTF-8 ASCII encoded text. At the parent-level (mime-sdp-rule) the **match-value** and **new-value** parameters execute against the entire SDP as a single string.

An add action only succeeds in the absence of SDP because a message is allowed only a single SDP offer or answer. A delete operation at the mime-sdp-rule level will remove the SDP entirely.

Note that on an inbound sip-manipulation, SDP manipulations interact with the Oracle Communications Session Border Controller codec-policy. SDP manipulations also interact with codec reordering and media setup. It is very possible to make changes to the SDP such that the call can not be setup due to invalid media parameters, or settings that will affect the ability to transcode the call. Consequently, user manipulation of the SDP can prove risky, and should be approached with appropriate caution.

Three configuration-objects, sdp-session-rule, sdp-media-rule, and mime-header-rule, exist under the mime-sdp-rule. These objects provide finer grained control of manipulating parts of the SDP.

### sdp-session-rule

An sdp-session-rule groups all SDP descriptors, up until the first media line, into a single entity, thus allowing the user to perform manipulation operations on a session-specific portion of the SDP.

Like the mime-sdp-rule, all match-value and new-value operations performed at this level are executed against the entire session group as a complete string. Given the sample SDP below, if an sdp-session-rule is configured, the match-value and new-values operate only on the designated portion.

```
v=0
o=mhandley 2890844526 2890842807 IN IP4 126.16.64.4
s=SDP Seminar
i=A Seminar on the session description protocol
u=http://www.cs.ucl.ac.uk/staff/M.Handley/sdp.03.ps
e=mjh@isi.edu (Mark Handley)
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
a=recvonly
m=audio 49170 RTP/AVP 0
m=video 51372 RTP/AVP 31
m=application 32416 udp wb
a=orient:portrait
```

Nested under the sdp-session-rule configuration object is an sdp-line-rule object, the object that identifies individual descriptors within the SDP. The types of descriptors used at the sdp-session-rule level are v, o, s, i, u, e, p, c, b, t, r, z, k, and a, the descriptors specific to the entire session description.

This level of granularity affords the user a very simple way to making subtle changes to the session portion of the SDP. For instance, it is very common to have to change the connection line at the session level.

The add and delete actions perform no operation at the sdp-session-rule level.

### sdp-media-rule

An sdp-media-rule groups all of the descriptors that are associated with a specific media-type into single entity, thus allowing the user to perform manipulation operations on a media-specific portion of the SDP. For example, a user can construct an sdp-media-rule to change an attribute of the audio media type.

Like a mime-sdp-rule, all match-value and new-value operations performed at this level are executed against the entire media-group as a complete string. Given the sample SDP below, if a media-level-descriptor is configured to operate against the application group, the match-value and new-values would operate only on designated portion.

```
v=0
o=mhandley 2890844526 2890842807 IN IP4 126.16.64.4
s=SDP Seminar
i=A Seminar on the session description protocol
u=http://www.cs.ucl.ac.uk/staff/M.Handley/sdp.03.ps
e=mjh@isi.edu (Mark Handley)
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
a=recvonly
m=audio 49170 RTP/AVP 0
m=video 51372 RTP/AVP 31
m=application 32416 udp wb
a=orient:portrait
```

A configuration parameter **media-type** is used to specify the media group on which to operate. It contains all of the descriptors including the m-line up to the next m-line. This parameter is a string field and must match the media-type exactly as it appears within the SDP. The special media-type media can be used to refer to all media types. This is particularly useful when performing an add operation, when the user wants to add a media section between the first and second medias, but does not know what media type they are. Otherwise, during an add operation, the media section would be added before the specified media-type (if no index parameter was provided).

The types of descriptors used at the sdp-media-rule level are m, i, c, b, k, and a, the descriptors specific to the media description.

This level of granularity affords the user a very simple way to making subtle changes to the media portion of the SDP. For instance, it is very common to have to change the name of an audio format (for example G729 converted to g729b), or to add attributes specific to a certain media-type.

The index operator is supported for the media-type parameter (for example, media-type audio[1]). Like header rules, if no index is supplied, this means operate on all media-types that match the given name. For specifying specific media-types, the non-discrete indices are also supported (for example, ^ - last). Adding a media-type, without any index supplied indicates that the media should be added at the beginning. The special media-type media uses the index as an absolute index to all media sections, while a specific media-type will index relative to that given media type.

For sdp-media-rules set to an action of add where the media-type is set to media, the actual media type is obtained from the new-value, or more specifically, the string after m= and before the first space.

Given the following SDP:

```
v=0
o=mhandley 2890844526 2890842807 IN IP4 126.16.64.4
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
m=audio 49170 RTP/AVP 0
m=audio 48324 RTP/AVP 8
m=video 51372 RTP/AVP 31
```

With the sdp-media-rule:

```
sdp-media-rule
name                smr
media-type          audio[1]
action              manipulate
comparison-type     case-sensitive
match-value
new-value            "m=audio 1234 RTP/AVP 8 16"
```

This rule operates on the 2nd audio line, changing the port and adding another codec, resulting in the SDP:



```
v=0
o=mhandley 2890844526 2890842807 IN IP4 126.16.64.4
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
m=audio 49170 RTP/AVP 0
m=audio 1234 RTP/AVP 8 16
m=video 51372 RTP/AVP 31
```

The following rule, however:

```
sdp-media-rule
  name                smr
  media-type          media[1]
  action              add
  comparison-type     case-sensitive
  match-value
  new-value           "m=video 1234 RTP/AVP 45"
```

adds a new video media-type at the 2nd position of all media-lines, resulting in the SDP:

```
v=0
o=mhandley 2890844526 2890842807 IN IP4 126.16.64.4
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
m=audio 49170 RTP/AVP 0
m=video 1234 RTP/AVP 45
m=audio 48324 RTP/AVP 8
m=video 51372 RTP/AVP 31
```

### sdp-line-rule

Unlike header-rules, sdp descriptors are not added in the order in which they are configured. Instead they are added to the SDP adhering to the grammar defined by RFC 4566 (as is shown below).

```
Session description
  v= (protocol version)
  o= (originator and session identifier)
  s= (session name)
  i=* (session information)
  u=* (URI of description)
  e=* (email address)
  p=* (phone number)
  c=* (connection information -- not required if included in
      all media)
  b=* (zero or more bandwidth information lines)
  One or more time descriptions ("t=" and "r=" lines; see
  below)
  z=* (time zone adjustments)
  k=* (encryption key)
  a=* (zero or more session attribute lines)
  Zero or more media descriptions (see below)

Time description
  t= (time the session is active)
  r=* (zero or more repeat times)

Media description, if present
  m= (media name and transport address)
  i=* (media title)
  c=* (connection information -- optional if included at
      session level)
  b=* (zero or more bandwidth information lines)
  k=* (encryption key)
  a=* (zero or more media attribute lines)
```

\* after the equal sign denotes an optional descriptor.

## SIP Signaling Services

---

This hierarchy is enforced meaning that if you configure a rule which adds a session name descriptor followed by a rule which adds a version descriptor, the SDP will be created with the version descriptor first, followed by the session name.

The only validation that will occur is the prevention of adding duplicate values. In much the same way that header-rules prevents the user from adding multiple To headers, the descriptor rule will not allow the user to add multiple descriptors; unless multiple descriptors are allowed, as is in the case of b, t, r and a.

There exists a parameter **type** under the sdp-line-rule object that allows the user to specify the specific line on which to perform the operation. For example: v, o, s, i, u, e, p, c, b, t, r, z, k, a, and m. Details on these types can be found in RFC 4566.

For those descriptors, of which there may exist zero or more (b, t, r, and a) entries, indexing grammar may be used to reference the specific instance of that attribute. This indexing grammar is consistent with that of header-rules for referring to multiple headers of the same type.

Given the example SDP below:

```
v=0
o=mhandley 2890844526 2890842807 IN IP4 126.16.64.4
s=SDP Seminar
i=A Seminar on the session description protocol
u=http://www.cs.ucl.ac.uk/staff/M.Handley/sdp.03.ps
e=mjh@isi.edu (Mark Handley)
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
r=604800 3600 0 90000
r=7d 1h 0 25h
a=recvonly
m=audio 49170 RTP/AVP 0
m=video 51372 RTP/AVP 31
m=application 32416 udp wb
a=orient:portrait
```

and the following sdp-line-rule:

```
sdp-line-rule
  name          removeRepeatInterval
  type          r[1]
  action        delete
```

The rule `removeRepeatInterval` removes the second repeat interval descriptor within the SDP.

The behavior of all SDP rules follow the same behavior of all manipulation rules in that they are executed in the order in which they are configured and that each rule executes on the resultant of the previous rule.

Each descriptor follows its own grammar and rules depending on the type specified. The values of the descriptor are evaluated at runtime since the new-values themselves are evaluated at runtime. At this time no validation of the grammar for each of the types is performed. The user is responsible for properly formatting each of the descriptors according to their specifications.

For instance, the version (v) descriptor can be removed from the SDP but leaving all descriptors for that SDP, causing the SDP to become invalid. This is consistent with the way header-rules operate, in that there is no validation for the specific headers once they have been manipulated through HMR.

## Regular Expression Interpolation

An interpolated regular expression is a regular expression that is compiled and evaluated at runtime. Today all regular expressions are compiled at configuration time in order to improve performance. There are cases where a regular expression is determined dynamically from data within a SIP message. In these circumstances the regular expression is unknown until the time of execution.

In order to have a regular expression be interpolated at runtime, it must be contained within a set of `{}`. An interpolated expression can have any number of regular expressions and strings appended together. Any characters to the left or right of the curly braces will be appended to the value within the curly braces. The curly braces are

effectively two operators treated as one (interpolate the value contained within and then concatenate the values to the left and right of the curly braces). If the comparison-type is set to pattern-rule and the match-value contains a value that matches the grammar below, then it will be treated as an interpolated expression.

```
([^\|^\)\{\$[\^0-9]+\^})*\}
```

The example below demonstrates using a user defined variable within a regular expression of another rule at runtime.

```
element-rule
    name                someRule
    type                header-value
    action              replace
    comparison-type     pattern-rule
    match-value         ^sip:{$rule1.$0}@(.+)$
    new-value           sip:bob@company.com
```

If the value of \$rule1.\$0 evaluates to alice then it will successfully match against the string sip:alice@comcast.net. An interpolated expression can be as simple as “{\$rule1.\$0}” or as complex as ^sip:{\$rule1.\$0}@{\$rule2[1].\$2}\$. It is not possible to interpolate a normal regular expression since the grammar will not allow the user to enter such an expression. Only variables can be contained with the curly braces.

The resultant of interpolated expressions can be stored in user defined variables. Given the same example from above, if the rule someRule was referenced by another rule, the value of sip:alice@comcast.net would be stored within that rule.

Interpolation only makes a single pass at interpolation, but does so every time the Rule executes. In other words, if the Rule is applied to the Route header, it will interpolate again for each Route header instance. What this means is that the value within the curly braces will only be evaluated once. For instance, if the value {\$someRule.\$1} evaluates to {\$foobar.\$2} the Oracle Communications Session Border Controller (SBC) will treat \$foobar.\$2 as a literal string which it will compile as a regular expression. The SBC will not recursively attempt to evaluate \$foobar.\$2, even if it was a valid user defined variable.

Interpolated regular expressions will evaluate to TRUE if and only if both the regular expression itself can be compiled and it successfully matches against the compared string.

## Regular Expressions as Boolean Expressions

Regular expressions can be used as boolean expressions today if they are the only value being compared against a string, as is shown in the case below.

```
mime-rule
    name                someMimeRule
    content-type        application/text
    action              replace
    comparison-type     pattern-rule
    match-value         ^every good boy .*
    new-value           every good girl does fine
```

However, regular expressions can not be used in conjunction with other boolean expressions to form more complex boolean expressions, as is shown below.

```
mime-rule
    name                someMimeRule
    content-type        application/text
    action              replace
    comparison-type     boolean
    match-value         $someRule & ^every good boy .*
    new-value           every good girl does fine
```

There are many cases where the user has the need to compare some value as a regular expression in conjunction with another stored value. It is possible to perform this behavior today, however it requires an extra step in first storing the value with the regular expression, followed by another Manipulation Rule which compares the two boolean expressions together (e.g. \$someRule & \$someMimeRule).

In order to simplify the configuration of some sip-manipulations and to make them more efficient this functionality is being added.

Unfortunately, it is not possible to just use the example as is shown above. The problem is there are many characters that are commonly used in regular expressions that would confuse the HMR expression parser (such as \$, and +). Therefore delimiting characters need to be used to separate the regular expression from the other parts of the expression.

To treat a regular expression as a boolean expression, it needs to be enclosed within the value `$REGEX(<expression>,<compare_string>=$ORIGINAL)`; where `<expression>` is the regular expression to be evaluated. `<compare_string>` is the string to compare against the regular expression. This second argument to the function is defaulted to `$ORIGINAL` which is the value of the of the specific Manipulation Rule object. It can be overridden to be any other value the user desires.

The proper configuration for the example above to use regular expressions as boolean expressions is

```
mime-rule
  name          someMimeRule
  content-type  application/text
  action        replace
  comparison-type boolean
  match-value   $someRule & $REGEX("^every good boy .*")
  new-value     every good girl does fine
```

It is also possible to use expressions as arguments to the `$REGEX` function. These expressions will in turn be evaluated prior to executing the `$REGEX` function. A more complex example is illustrated below.

```
header-rule
  name          checkPAU
  header-name   request-uri
  action        reject
  comparison-type boolean
  match-value   (!$REGEX($rule1[0], $FROM_USER)) &
               (!$REGEX($rule2[0], $PAI_USER))
  msg-type      request
  new-value     403:Forbidden
  methods       INVITE, SUBSCRIBE, MESSAGE, PUBLISH,
               OPTIONS, REFER
```

It should be noted that when using `$REGEX()` in a boolean expression, the result of that expression is not stored in the user variable. The `comparison-type` must be set to `pattern-rule` in order to store the result of a regular expression.

The arguments to the `$REGEX()` function are interpolated by default. This is the case since the arguments themselves must be evaluated at runtime. The following example is also valid.

```
mime-rule
  name          someMimeRule
  content-type  application/text
  action        replace
  comparison-type boolean
  match-value   $someRule & $REGEX("^every good
               {$rule1[0].$0} .*")
```

## Moving Manipulation Rules

Users can move rules within any manipulation-rule container. Any manipulation rule which contains sub-rules will now offer the ACLI command `move <from index> <to index>`. For example, given the order and list of rules below:

1. rule1
2. rule2
3. rule3
4. rule4

Moving rule3 to position 1 can be achieved by executing `move 3 1`. The resulting order will then be: rule3, rule1, rule2, rule4. A move operation causes a shift (or insert before) for all other rules. If a rule from the top or middle moves to the bottom, all rules above the bottom are shifted up to the position of the rule that was moved. If a rule from the bottom or middle moves to the top, all rules below are shifted down up to the position of the rule that was moved. Positions start from 1.

A valid from-index and to-index are required to be supplied as arguments to the move action. If a user enters a range that is out of bounds for either the from-index or to-index, the CLI will inform the user that the command failed to execute and for what reason.

With respect to the issue of creating an invalid sip-manipulation by incorrectly ordering the manipulation rules, this issue is handled by the Oracle Communications Session Border Controller validating the rules at configuration time and treating them as invalid prior to runtime. This may or may not affect the outcome of the sip-manipulation as a configured rule may not perform any operation if it refers to a rule that has yet to be executed. It is now the user's responsibility to reorder the remaining rules in order to make the sip-manipulation valid once again.

It is important to note that rules of a different type at the same level are all part of the same list. To clarify; header-rules, mime-rules, mime-isup-rules and mime-sdp-rules all share the same configuration level under sip-manipulation. When selecting a move from-index and to-index for a header-rule, one must take into consideration the location of all other rules at the same level, since the move is relative to all rules at that level, and not relative to the particular rule you have selected (for example, the header-rule).

Since the list of rules at any one level can be lengthy, the `move` command can be issued one argument at a time, providing the user with the ability to select indices. For instance, typing `move` without any arguments will present the user with the list of all the rules at that level. After selecting an appropriate index, the user is then prompted with a to-index location based on the same list provided.

For Example:

```
ACMEPACKET(sip-mime-sdp-rules) # move
select a rule to move
```

```
1: msr sdp-type=any; action=none; match-value=; msg-type=any
```

```
2: addFoo header-name=Foo; action=none; match-value=; msg-type=any
```

```
3: addBar header-name=Bar; action=none; match-value=; msg-type=any
```

```
selection: 2
```

```
destination: 1
```

```
Rule moved from position 2 to position 1
```

```
ACMEPACKET(sip-mime-sdp-rules) #
```

## Rule Nesting and Management

There will be cases where the user wants to take a stored value from the SDP and place it in a SIP header, and vice-versa. All header-rules, element-rules, mime-rules, mime-isup-rules, isup-param-rules, mime-header-rules and mime-sdp-rules are inherited from a Manipulation Rule. A Sip Manipulation is of type Manipulation which contains a list of Manipulation Rules. Each Manipulation Rule can itself contain a list of Manipulation Rules. Therefore when configuring manipulation rules, they will be saved in the order which they have been configured. This is different from the way other configuration objects are configured. Essentially, the user has the option of configuring which type of object they want and when they are done, it gets added to the end of the sip-manipulation, such that order is preserved. This will mean that any Manipulation Rule at the same level can not share the same name. For example, names of header-rules can't be the same as any of the mime-sdp-rule ones or mime-isup-rule. This allows the user to reference stored values from one rule type in another at the same level.

## ACLI Configuration Examples

The following eight sections provide sample SDP manipulations.

### Remove SDP

```
sip-manipulation
  name          stripSdp
  description    remove SDP from SIP message
  mime-sdp-rule
    name        sdpStrip
    msg-type    request
    methods     INVITE
    action      delete
    comparison-type case-sensitive
    match-value
    new-value
```

### Remove Video from SDP

```
sip-manipulation
  name          stripVideo
  description    strip video codecs from SIP
                 message
  mime-sdp-rule
    name        stripVideo
    msg-type    request
    methods     INVITE
    action      manipulate
    comparison-type case-sensitive
    match-value
    new-value
    sdp-media-rule
      name      removeVideo
      media-type video
      action    delete
      comparison-type case-sensitive
match-value
  new-value
```

### Add SDP

```
sip-manipulation
  name          addSdp
  description    add an entire SDP if one does
                 not exist
  mime-sdp-rule
    name        addSdp
    msg-type    request
    methods     INVITE
    action      add
    comparison-type case-sensitive
    match-value
    new-value    "v=0\r\no=mhandley
2890844526 2890842807 IN IP4 "+$LOCAL_IP+"\r\ns=SDP Seminar\r\ni=A
Seminar on the session description protocol\r\nu=http:
//www.cs.ucl.ac.uk/staff/M.Handley/sdp.03.ps\r\nne=mjh@isi.edu
(Mark Handley)\r\nc=IN IP4 "+$LOCAL_IP+"\r\nt=2873397496
2873404696\r\na=recvonly\r\nm=audio 49170 RTP/AVP 0\r\n"
```

### Manipulate Contacts

This rule changes the contact in the SDP to the value contained in the Contact header.

```
sip-manipulation
  name          changeSdpContact
  description    changes the contact in the SDP to the
value of the contact header
  header-rule
```

```

name storeContact
header-name Contact
action store
comparison-type pattern-rule
msg-type request
methods INVITE
match-value
new-value
element-rule
    name storeHost
    parameter-name
    type uri-host
    action store
    match-val-type ip
    comparison-type pattern-rule
    match-value
    new-value
mime-sdp-rule
name changeConnection
msg-type request
methods INVITE
action manipulate
comparison-type case-sensitive
match-value
new-value
sdp-session-rule
name changeCLine
action manipulate
comparison-type case-sensitive
match-value
new-value
sdp-line-rule
name updateConnection
type c
action replace
comparison-type case-sensitive
match-value $storeContact.$storeHost
new-value $storeContact.$storeHost.$0

```

## Remove a Codec

This rule changes the contact in the SDP to the value contained in the Contact header.

```

sip-manipulation
name removeCodec
description remove G711 codec if it exists
mime-sdp-rule
name removeCodec
msg-type request
methods INVITE
action manipulate
comparison-type case-sensitive
match-value
new-value
sdp-media-rule
name removeG711
media-type audio
action manipulate
comparison-type case-sensitive
match-value
new-value
sdp-line-rule
name remove711
type m

```

```

        action                replace
        comparison-type      pattern-rule
        match-value           ^(audio [0-9]
                               {1,5} RTP.*) ( [07]
                               \b) (.*)$
                               $1+$3
        new-value
sdp-line-rule
        name                  stripAttr
        type                   a
        action                 delete
        comparison-type      pattern-rule
        match-value           ^(rtpmap|fmtpt):
                               [07]\b$
        new-value

```

### Change Codec

```

sip-manipulation
    name                convertCodec
    description         changeG711toG729
    mime-sdp-rule
        name            changeCodec
        msg-type        request
        methods         INVITE
        action          manipulate
        comparison-type case-sensitive
        match-value
        new-value
    sdp-media-rule
        name            change711to729
        media-type      audio
        action          manipulate
        comparison-type case-sensitive
        match-value
        new-value
    sdp-line-rule
        name            change711
        type             m
        action          replace
        comparison-type pattern-rule
        match-value     ^(audio [0-9]{4,5}
                        RTP/AVP.*) ( 0) (.*)$
new-value             $1+" 18"+$3
    sdp-line-rule
        name            stripAttr
        type             a
        action          delete
        comparison-type pattern-rule
        match-value     ^rtpmap:0 PCMU/
                        .+$
        new-value
    sdp-line-rule
        name            addAttr
        type             a
        action          add
        comparison-type boolean
        match-value     $change711to729.
                        $stripAttr
        new-value       rtpmap:18 G729/8000

```

### Remove Last Codec and Change Port

```

sip-manipulation
    name                removeLastCodec

```



```

description                remove the last codec
mime-sdp-rule
  name                      removeLastCodec
  msg-type                  request
  methods                   INVITE
  action                     manipulate
  comparison-type           case-sensitive
  match-value
  new-value
  sdp-media-rule
    name                    removeLast
    media-type               audio
    action                   manipulate
    comparison-type          case-sensitive
    match-value
    new-value
    sdp-line-rule
      name                   isLastCodec
      type                   m
      action                 store
      comparison-type        pattern-rule
      match-value            ^(audio )([0-9]{4,
                             5})( RTP/AVP
                             [0-9]{1-3})$
new-value
  sdp-line-rule
    name                    changePort
    type                    m
    action                   replace
    comparison-type          boolean
    match-value              $removeLastCodec.
                             $removeLast.$isLastCodec
    new-value                $removeLastCodec.
                             $removeLast.$isLastCodec.$1+$removeLastCodec.$removeLast.
                             $isLastCodec.$3

```

### Remove Codec with Dynamic Payload

```

sip-manipulation
  name                      removeAMR
  description                remove the AMR and AMR-WB dynamic
codecs
  mime-sdp-rule
    name                    sdpAMR
    msg-type                  request
    methods                   INVITE
    action                     manipulate
    comparison-type           case-sensitive
    match-value
    new-value
    sdp-media-rule
      name                   mediaAMR
      media-type              audio
      action                   manipulate
      comparison-type          case-sensitive
      match-value
      new-value
      sdp-line-rule
        name                   isAMR
        type                    a
        action                 delete
        comparison-type        pattern-rule
        match-value            ^rtptime: ([0-9]
                                {2,3}) AMR

```

```
new-value
sdp-media-rule
  name          mediaIsAMR
  media-type    audio
  action        manipulate
  comparison-type boolean
  match-value   $sdpAMR.$media
               AMR.$isAMR

new-value
sdp-line-rule
  name          delFmtpAMR
  type          a
  action        delete
  comparison-type pattern-rule
  match-value   ^fmtp:{$sdpAMR.
               $mediaAMR.
               $isAMR.$1}\b

new-value
sdp-line-rule
  name          delAMRcodec
  type          m
  action        find-replace-all
  comparison-type pattern-rule
  match-value   ^audio [0-9]+
               RTP.*( {$sdpAMR.
               $mediaAMR.$isAMR.
```

## Dialog Transparency

---

This section explains how to configure dialog transparency, which prevents the Oracle Communications Session Border Controller from generating a unique Call-ID and modifying dialog tags.

### Overview

With dialog transparency enabled, the Oracle Communications Session Border Controller is prevented from generating a unique Call-ID and from modifying the dialog tags; the Oracle Communications Session Border Controller passes what it receives. Therefore, when a call made on one Oracle Communications Session Border Controller is transferred to another UA and crosses a second Oracle Communications Session Border Controller, the second Oracle Communications Session Border Controller does not note the context of the original dialog, and the original call identifiers are preserved end to end. The signalling presented to each endpoint remains in the appropriate context regardless of how many times a call crosses through a Oracle Communications Session Border Controller or how many Oracle Communications Session Border Controllers a call crosses.

Without dialog transparency enabled, the Oracle Communications Session Border Controller's SIP B2BUA rewrites the Call-ID header and inserted dialog cookies into the From and To tags of all messages it processes. These dialog cookies are in the following format: SDxxxxNN-. Using these cookies, the Oracle Communications Session Border Controller can recognize the direction of a dialog. However, this behavior makes call transfers problematic because one Oracle Communications Session Border Controllers' Call-ID might not be properly decoded by another Oracle Communications Session Border Controller. The result is asymmetric header manipulation and failed call transfers.

### Dialog Transparency Configuration

You set one parameter in your SIP configuration to enable dialog transparency.

- For new configurations, this feature defaults to enabled

To enable SIP dialog transparency:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the **session-router** path.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-config
```

4. Use the ACLI **select** command so that you can work with the SIP configuration.

```
ACMEPACKET (sip-config) # select
```

5. **dialog-transparency**—Enter the state of SIP dialog transparency you require for your Oracle Communications Session Border Controller. The default value is **enabled**. The valid values are:

- enabled | disabled

## Route Header Removal

This section explains how to enable the Oracle Communications Session Border Controller to disregard and strip all SIP Route headers. You set an option in a SIP interface configuration to strip all Route headers for SIP requests coming from this interface.

When the Oracle Communications Session Border Controller with this option configured receives an INVITE from an interface, it removes the route headers. However, although it removes the headers, the Oracle Communications Session Border Controller maintains backward compatibility with RFC 2543 nodes. To do so, it normalizes the request to an RFC 3261 loose routing form before it removes the headers.

### Route Header Removal Configuration

The following information explains how to remove SIP route headers.

To configure SIP route header removal:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-interface) #
```

4. Type **options strip-route-headers** and press Enter. This completes the configuration of SIP route header removal.

```
ACMEPACKET (sip-interface) # options strip-route-headers
```

## SIP Via Transparency

This section explains the inbound Via header transparency feature, which enables the Oracle Communications Session Border Controller to insert its Via header on top of the top-most Via header received from user equipment (UE). It then forwards it on to the IP Multimedia Subsystem (IMS) core with the original Via header now located as the bottom-most Via header.

The Oracle Communications Session Border Controller still replaces the Contact and other header addresses with its own, and does not pass on the core's Via headers in outbound requests.

This feature is targeted for the Telecoms & Internet converged Services & Protocols for Advanced Networks (TISPAN) with SIP hosted NAT traversal support. It works with SIP NAT bridged, local-policy routed, and non-SIP NAT configurations, regardless of registration handling.

Some equipment acts as Proxy-CSCF (P-CSCF) and Serving-CSCF (S-CSCF) nodes, with the Oracle Communications Session Border Controller located between the equipment and user endpoints. The equipment needs to see the each user endpoint's original Via header in order to perform some implicit authentication, admission, and control functions in a TISPAN-compliant model.

You enable Via header transparency on the access SIP interface. Received Via headers are saved for inclusion in requests going out another interface or session agent that does not have the parameter set, in other words, the core side. For any received SIP message where the inbound previous hop interface was enabled for Via header transparency, the Oracle Communications Session Border Controller adds its own Via header as it forwards it, and it also copies the received top-most Via as the new bottom-most Via, if the outbound next hop interface/session agent is not enabled for Via header transparency. The Oracle Communications Session Border Controller also adds a `received=` parameter to the copied Via header, per the SIP RFC 3261.

Any message received from an interface without Via header transparency enabled, does not have the received Via header copied over to any other direction.

For HNT, where the original top-most (and only) Via header from a UE is a private/false address, the SD should still copy that false address into the core-side, and the `received=` parameter will contain the real Layer-3 addressing.

### SIP Via Transparency Configuration

You can configure SIP Via header transparency for the access SIP interface using the ACLI.

To configure SIP Via header transparency for an access interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the media-level configuration elements.

```
ACMEPACKET(configure)# session-router  
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface  
ACMEPACKET(sip-interface)#
```

4. You can either add support to a new SIP interface configuration or to an existing SIP interface configuration:

For a new SIP interface configuration, you can add the option by typing options, a Space, and then `via-header-transparency`.

```
ACMEPACKET(sip-interface)# options via-header-transparency
```

For an existing SIP interface configuration without options configured, select the SIP interface, type options followed by a Space, and then `via-header-transparency`.

```
ACMEPACKET(sip-interface)# select  
ACMEPACKET(sip-interface)# options via-header-transparency
```

For an existing SIP interface configuration with options configured, select the SIP interface, type options followed by a Space, the plus sign (+), and the `via-header-transparency` option.


```
ACMEPACKET(sip-interface)# select  
ACMEPACKET(sip-interface)# options +via-header-transparency
```

5. Save your work using the ACLI **save** or **done** command.

## Symmetric Latching

Symmetric latching, or forced HNT, ensures that symmetric RTP/RTCP is used for a SIP endpoint. Symmetric RTP/RTCP means that the IP address and port pair used by an outbound RTP/RTCP flow is reused for the inbound flow. The IP address and port are learned when the initial RTP/RTCP flow is received by the Oracle Communications Session Border Controller. The flow's source address and port are latched onto and used as the destination for the RTP/RTCP sourced by the other side of the call. The IP address and port in the c line and m line respectively in the SDP message are ignored.

If your network is configured with nested realms in order to separate signalling from media, make sure that the symmetric latching feature is enabled on the signaling realm.

 **Note:** This description is applicable to RTCP only when you also enable the HNT RTCP option in the media-manager configuration. Do not enable symmetric latching on core-facing interfaces.

## Symmetric Latching Configuration

To configure symmetric latching:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-level configuration elements.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

4. Select the realm where you want to apply this feature.

```
ACMEPACKET (realm-config) # select
identifier:
1: Acme_Realm <none>          0.0.0.0
2: H323REALM <none>         0.0.0.0
selection: 1
ACMEPACKET (realm-config) #
```

5. **symmetric-latching** – Enable symmetric latching on the SBC. This completes the configuration of forced HNT. The default value for this parameter is **disabled**. The valid values are:

- enabled | disabled

```
ACMEPACKET (realm-config) # symmetric-latching enabled
```

6. Save your work using the ACLI **save** or **done** command.

## Enabling RTCP Latching

To enable RTCP symmetric latching:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-level configuration elements.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **media-manager** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

## SIP Signaling Services

---

```
ACMEPACKET(media-manager) # media-manager
ACMEPACKET(media-manager-config) #
```

4. Select the media manager configuration so that you can enable HNT RTCP.

```
ACMEPACKET(media-manager-config) # select
```

5. **hnt-rtcp** – Enable support of RTCP when the SBC performs HNT. The default value is **disabled**. The valid values are:

- enabled | disabled

```
ACMEPACKET(media-manager-config) # hnt-rtcp enabled
```

6. Save your work using either the ACLI **save** or **done** command.

## SIP Number Normalization

---

This section explains the SIP number normalization feature that applies to the SIP To URI. (Currently the Oracle Communications Session Border Controller supports number normalization on From and To addresses for both inbound and outbound call legs.) Number normalization includes add, delete, and replace string functions that result in consistent number formats.

Number normalization is supported for the following call types:

- SIP to SIP
- H.323 to SIP

Number normalization applies to the SIP To URI. It occurs on ingress traffic, prior to the generation of accounting records or local policy lookups. RADIUS CDR attributes are populated with the normalized numbers. Local policy matching is based on the normalized numbers.

## Terminology

The following lists explains the terminology used later.

- X is any digit having the value 0 through 9
- N is any digit having the value 2 through 9
- 0/1 is a digit having the value of either 0 or 1
- NXX is a form of Numbering Plan Area (NPA).
- CC is a 1, 2, or 3 digit country code used in international dialing
- NN is a national number that can be a four to fourteen digit national number used in international dialing, where the combination of CC+NN is a 7 to 15 digit number.
- + symbol in E.164 indicates that an international prefix is required
- E.164 numbers are globally unique, language independent identifiers for resources on Public Telecommunication Networks that can support many different services and protocols.
- N11 number is any of the three-digit dialing codes in the form N11 used to connect users to special services, where N is a digit between 2 and 9

## Calls from IP Endpoints

The Oracle Communications Session Border Controller uses the following number normalization rules:

- North American Numbering Plan (NANP) calls: where a number with the format 1NPANXXXXXX is received, the Oracle Communications Session Border Controller adds a plus sign (+) as a prefix to the NANP number. The Oracle Communications Session Border Controller also adds the string ;user=phone after the host IP address in the SIP URI. For example:

```
sip:+1NPANXXXXXX@ipaddr;user=phone
```

- International NWZ1 calls: Oracle Communications Session Border Controller receives an international call with the format 011CCNN. The Oracle Communications Session Border Controller deletes the 011 prefix and adds a

plus sign (+) as a prefix to CC+NN; and also adds the string ;user=phone after the host IP address in the SIP URI. For example:

```
sip:+CCNN@ipaddr;user=phone
```

- Private number calls: when a private number with the format nxxxx (where n=2 through 9) is received, no number normalization is applied by the Oracle Communications Session Border Controller.
- Calls to numbers such as N11, 0-, 0+, 00-, and 01+: the Oracle Communications Session Border Controller adds ;phone-context=+1 after the number and also adds the string ;user=phone after the host IP address in the SIP URI. For example:

```
sip:N11;phone-context=+1@ipaddr;user=phone
sip:01CCNN;phone-context=+1@ipaddr;user=phone
```

- Calls with numbers that are already normalized are not modified by the Oracle Communications Session Border Controller.

## Calls from IP Peer Network

For calls received from external peer networks, the Oracle Communications Session Border Controller uses the following number normalization rules:

- Global numbers such as NANP and international E.164 numbers should have already been normalized. If not, the Oracle Communications Session Border Controller applies the same number normalization rules listed in the prior section.
- Calls to numbers such as N11, 0-, 0+, 00-, and 01+: the Oracle Communications Session Border Controller adds ;phone-context=+1 after the number and also adds the string ;user=phone (if absent) after the host IP address in the SIP URI.

## SIP Number Normalization Configuration

You can configure SIP number normalization for the realm and session agent using the ACLI.

### Realm

To configure SIP number normalization for a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-level configuration elements.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. You can either add SIP number normalization support to a new session agent configuration or to an existing session agent configuration:

- For a new realm configuration, add the option by typing **options**, a Space, and then **number-normalization**.

```
ACMEPACKET(realm-config)# options number-normalization
```

- For an existing realm configuration without any options already configured, select the realm, type **options** followed by a Space, and then **number-normalization**.

```
ACMEPACKET(realm-config)# select
ACMEPACKET(realm-config)# options number-normalization
```

- For an existing realm configuration with other options, select the realm, type **options** followed by a Space, the plus sign (+), and the **number-normalization** option.

## SIP Signaling Services

---

```
ACMEPACKET (realm-config) # select  
ACMEPACKET (realm-config) # options +number-normalization
```

5. Save your work using the ACLI **save** or **done** command.

### Session Agent

To configure SIP number normalization for a session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the media-level configuration elements.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # session-agent  
ACMEPACKET (session-agent) #
```

4. You can either add SIP number normalization support to a new session agent configuration or to an existing session agent configuration:

- For a new a session agent configuration, add the option by typing **options**, a Space, and then **number-normalization**.

```
ACMEPACKET (session-agent) # options number-normalization
```

- For an existing session agent configuration without any options already configured, select the session agent, type **options** followed by a Space, and then **number-normalization**.

```
ACMEPACKET (session-agent) # select  
ACMEPACKET (session-agent) # options number-normalization
```

- For an existing session agent configuration with other options, select the session agent, type **options** followed by a Space, the plus sign (+), and the **number-normalization** option.

```
ACMEPACKET (session-agent) # select  
ACMEPACKET (session-agent) # options +number-normalization
```

5. Save your work using the ACLI **save** or **done** command.

## SIP Port Mapping

---

This section contains information about the SIP port mapping feature. SIP port mapping lets you allocate a unique SIP signaling transport address (IP address and UDP port) on the Oracle Communications Session Border Controller in the provider network for each registered endpoint (user agent).

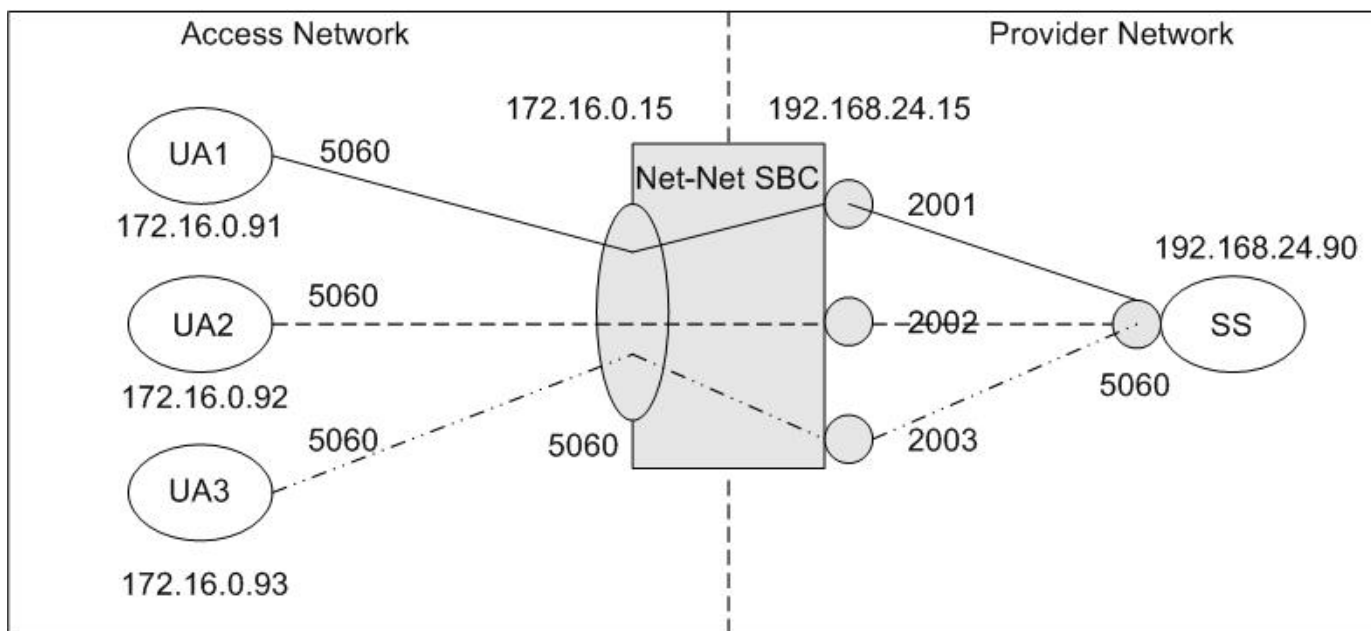
### About SIP Port Mapping

You might need to provide a unique signaling transport address for each registered endpoint for admission control, if required by your softswitch vendor. If you have questions about your softswitch, contact the vendor for assistance.

When a Oracle Communications Session Border Controller resides between the endpoints and the softswitch, the softswitch sees the same transport address (that of the Oracle Communications Session Border Controller) for all endpoints. By allocating a unique UDP port for each endpoint, the Oracle Communications Session Border Controller provides each of them a unique transport address.

The following example illustrates the SIP port mapping feature.





The diagram shows UA1, UA2, and UA3 are endpoints within the access network and that the SIP interface for the access network is 172.16.0.15:5060. On the provider network, the SIP interface is at 192.168.24.15, with the SIP port mapping feature enabled. The softswitch/registrar is also located on the provider network at 192.168.24.90:5060.

The diagram shows that port 2001 on the provider network is allocated to UA1 on the access network, port 2002 is allocated to UA2, and port 2003 is allocated to UA3. Because of this allocation, all SIP signaling messages sent from the endpoints in the access network to the softswitch on the provider network travel through an allocated signaling port. For example, all signaling messages between UA1 and the softswitch use 192.168.24.15:2001 as the transport address.

## How SIP Port Mapping Works

The Oracle Communications Session Border Controller (SBC) allocates SIP port mapping (signaling) ports during a REGISTER request that has registration caching applied. When you define a range of signaling ports for the SIP interface, you create a pool of signaling ports that can be allocated during the REGISTER request.

The SBC allocates a signaling port from the pool when it creates the registration cache entry for a Contact in a REGISTER request. It allocates a separate signaling port for each unique Contact URI from the access side. The registration cache Contact entry contains the mapping between the Contact URI in the access/endpoint realm (the UA-Contact) and the Contact URI in the registrar/softswitch realm (the SD-Contact).

The SD-Contact is the allocated signaling port. The signaling port gets returned to the pool when the Contact is removed from the registration cache. The removal can occur when the cache entry expires; or when the endpoint sends a REGISTER request to explicitly remove the Contact from the registrar. When a signaling port returns to the pool it gets placed at the end of pool list; in a least-recently-used allocation method for signaling ports.

When the SBC forwards the REGISTER request to the softswitch, it replaces the UA-Contact with SD-Contact. For example, if UA1 sends a REGISTER request with a Contact URI of sip:ua1@172.16.0.91:5060, it is replaced with sip:192.168.24.15:2001 when the REGISTER request is forwarded to the registrar.

The same translation occurs when UA1 sends that same URI in the Contact header of other SIP messages. SIP requests addressed to the allocated signaling transport address (SD-Contact) are translated and forwarded to the registered endpoint contact address (UA-Contact).

**Note:** The maximum number of registered endpoints cannot exceed the number of signaling ports available. If no signaling ports are available for a new registration, the REGISTER request receives a 503 response.

The SBC still processes requests received on the configured SIP port address. Requests sent into the registrar/softswitch realm that are not associated with a registered user will use the configured SIP port address.

## SIP Signaling Services

Using SIP port mapping with SIPconnect—where unique ports are used for each registered PBX—hinders the SBC from routing incoming calls to the corresponding PBX because the SBC uses DN for the PBX’s parent during registration, but the incoming INVITE from the softswitch contains the child DN in its Request URI. Thus the SBC cannot find a matching SBC-Contact because the username of the Request URI contains the child DN, but the username of the SBC-Contact contains the parent DN.

You can enable SIPconnect support in either the realm configuration or session agent for the SIP access network by setting the `sip-connect-pbx-reg` option. With this option set and the destination realm configured for port mapping, the SBC inserts a special search key in the registration table. Rather than adding the SD-Contact as the key as with regular (non-SIPconnect) registrations, the SBC strips user information and instead uses the host and port information as the registration key. The SBC still forwards the registration message with an intact contact username.

### SIP Port Mapping Based on IP Address

Some registrars need to know that multiple contacts represent the same endpoint. The extension to this feature answers the expectation from registrars that an endpoint registering multiple AoRs will use a single core-side mapped port to show that the AoRs really represent a single endpoint.

When you enable SIP port mapping based on IP Address, the Oracle Communications Session Border Controller supports core-side UDP port mapping based on the endpoint’s IP address. It ignores the username portion of the AoR or Contact.

The Oracle Communications Session Border Controller performs the port mapping allocation and lookup based on all requests using the via-key from the SIP Request. The via-key is a combination of Layer 3 and Layer 5 IP information in the message. The Oracle Communications Session Border Controller performs an additional lookup in the registration table to determine if a via-key already exists. If it does, then the Oracle Communications Session Border Controller uses the port already allocated and does not allocate a new one.

## About NAT Table ACL Entries

To enable SIP signaling messages to reach the host processor, the Oracle Communications Session Border Controller adds NAT table ACL entries for each SIP interface. With UDP without SIP port mapping applied, it adds a single ACL entry for each SIP port in the SIP interface configuration. For example:

```
untrusted entries:
intf:vlan source-ip/mask:port/mask dest-ip/mask:port/mask prot type index
0/0:0 0.0.0.0 172.16.1.15:5060 UDP static 10
0/3:0 0.0.0.0 192.168.24.15:5060 UDP static 16
0/1:0 0.0.0.0 192.168.50.25:5060 UDP static 17
```

### Using SIP Port Mapping


When you use SIP port mapping, one or more ACL entries are added to the NAT table to enable the range of ports defined. The NAT table does not support the specification of port ranges. However, it does support masking the port to enable ranges that fall on bit boundaries. For example, an entry for 192.168.24.15:4096/4 defines the port range of 4096 through 8191.

The algorithm for determining the set of ACLs for the port map range balances the need to represent the range as closely as possible, with the need to minimize the number of ACL entries. For example, a range of 30000 through 39999 would result in the following set of ACLs.

```
untrusted entries:
intf:vlan source-ip/mask:port/mask dest-ip/mask:port/mask prot type index
0/3:0 0.0.0.0 192.168.24.15:30000/4 UDP static 13
0/3:0 0.0.0.0 192.168.24.15:32768/4 UDP static 14
0/3:0 0.0.0.0 192.168.24.15:36864/4 UDP static 15
```

However, the first entry actually enables ports 28672 through 32767 and the last entry allows port 36864 through 40959. If SIP messages are received on ports outside the configured range (28672 through 29999 or 40000 through 40959 in this case), they are ignored.

Acme Packet recommends you use port map ranges that fall on bit boundaries to ensure the fewest possible ACL entries are created and only the configured ports are allowed by the ACLs. For example, a range of 32768 to 49151 provides for 16,384 signaling ports in a single ACL entry (192.168.24.15:32768/2).

 **Note:** If the ACLs added for the port map range do not include the SIP port configured in the SIP interface; the normal SIP ACL entry for the SIP port is also added.

## Dynamic Configuration

Dynamic configuration of SIP port mapping can cause disruption in service for existing registration cache entries; depending on the changes made to the defined port map range. If the range of mapping ports is reduced, it is possible that SIP signaling messages from the registrar/softswitch realm will no longer be sent to the host processor because of the changes in the NAT Table ACL entries.

When the range of mapping ports is changed, any signaling ports in the free signaling port pool not allocated to a registration cache entry are removed from the pool. When an allocated signaling port that is no longer part of the defined mapping port range is released, it is not returned to the pool of free steering ports.

The administrator is warned when the changed configuration is activated after the port map range of a SIP interface has been changed.

## Registration Statistics

The SIP registration cache statistics include counters for free and allocated signaling ports. You can issue a show registration command to display the statistics:

```
17:36:55-190
SIP Registrations
-- Period -- ----- Lifetime -----
Active High Total Total PerMax High
User Entries 4 4 0 7 4 4
Local Contacts 4 4 0 7 4 4
Free Map Ports 12284 12284 0 12291 12288 12288
Used Map Ports 4 4 0 7 4 4
Forwards - - 1 22 4
Refreshes - - 3 43 3
Rejects - - 0 0 0
Timeouts - - 0 1 1
Fwd Postponed - - 0 0 0
Fwd Rejected - - 0 0 0
Refr Extension 0 0 0 0 0 0
Refresh Extended - - 0 0 0
```

The labels for the first two items reflect the restructured registration cache:

- **User Entries:** counts the number of unique SIP addresses of record in the cache. Each unique address of record represents a SIP user (or subscriber). The address of record is taken from the To header in the REGISTER request. There might be one or more registered contacts for each SIP user. The contacts come from the Contact header of the REGISTER request.
- **Local Contacts:** counts the number of contact entries in the cache. Because the same user can register from multiple endpoints (user agents); the number of Local Contacts might be higher than the number of User Entries.
- **Free Map Ports:** counts the number of ports available in the free signaling port pool.
- **Used Map Ports:** counts the number of signaling ports allocated for registration cache entries. The value of Used Map Ports will equal the number of Local Contacts when the port mapping feature is used for all registrar/softswitch realms in the Oracle Communications Session Border Controller.

## SIP Port Mapping Configuration

You configure the SIP port mapping feature on a per-realm basis using the SIP interface configuration. Configure the port map range on the SIP interface for the realm where the registrar/softswitch resides. Port mapping is only applied when the access/ingress realm has registration caching and/or HNT enabled.

The range of SIP mapping ports must not overlap the following:

## SIP Signaling Services

- Configured SIP port, which might be used for signaling messages not associated with a registered endpoint.
- Port range defined for steering pool configuration using the same IP address as the SIP interface. If overlap occurs, the NAT table entry for the steering port used in a call prevents SIP messages from reaching the host processor.

To configure SIP port mapping:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-router path.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface  
ACMEPACKET(sip-interface)#
```


4. **port-map-start**—Set the starting port for the range of SIP ports available for SIP port mapping. The valid range is 1025 through 65535. The default value is 0 and when this value is set, SIP port mapping is disabled. The valid range is:

- Minimum: 0, 1025
- Maximum: 65535

```
ACMEPACKET(sip-interface)# port-map-start 32768
```

5. **port-map-end**—Set the ending port for the range of SIP ports available for SIP port mapping. The valid range is 1025 through 65535. If you set the value to the default 0, SIP port mapping is disabled. The valid range is:

- Minimum—0, 1025
- Maximum—65535

 **Note:** If not set to zero (0), the ending port must be greater than the starting port.

```
ACMEPACKET(sip-interface)# port-map-end 40959
```

6. **options**—If you want to use SIP port mapping based on IP address, set the options parameter by typing **options**, a Space, the option name **reg-via-key** with a plus sign in front of it, type the equal sign and the word **all**. Then press Enter.

```
ACMEPACKET(sip-interface)# options +reg-via-key=all
```

**If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.**

7. Save your work using the ACLI **done** command.

The following example shows SIP port mapping configured for a SIP interface:

```
sip-interface
  state          enabled
  realm-id       backbone
  sip-port
    address      192.168.24.15
    port          5060
    transport-protocol  UDP
    allow-anonymous  all
  sip-port
    address      192.168.24.15
    port          5060
    transport-protocol  TCP
    allow-anonymous  all
```

```

carriers
proxy-mode
redirect-action
contact-mode                none
nat-traversal               none
nat-interval                 30
registration-caching        enabled
min-reg-expire               120
registration-interval       3600
route-to-registrar           enabled
teluri-scheme                disabled
uri-fqdn-domain
trust-mode                   agents-only
max-nat-interval             3600
nat-int-increment            10
nat-test-increment           30
sip-dynamic-hnt              disabled
stop-recurse                 401,407
port-map-start               32768
port-map-end                  40959
last-modified-date           2005-09-23 14:32:15

```

## SIP Port Mapping for TCP and TLS

In releases prior to S-C6.2.0, the Oracle Communications Session Border Controller (SBC) supports SIP port mapping for UDP and now you can enable this feature for SIP sessions using TCP and TLS. Port mapping enables the SBC to allocate a unique port number for each endpoint registering through it by giving it a transport address (or hostport) in the registered Contact.

When you enable this feature for TCP and TLS, the SBC designates a port from a configured range for each endpoint that registers with SIP servers in the SIP interface's realm. You establish that range of ports using the **port-map-start** and **port-map-end** parameters. Unlike its behavior with UDP port mapping—where the SBC sends requests on the SIP interface from the allocated port mapping, the SBC sends all requests over an existing connection to the target next hop for TCP/TLS port mapping. If a connection does not exist, the system creates one. So for TCP/TLS port mapping, only the Contact header contains the transport address of the mapping port (i.e., the transport address of the configured SIP port). And the system refuses TCP and TLS connections on the allocated mapping port.

With TCP/TLS port mapping enabled, the SBC sends the Path header with the transport address in Register requests, unless you specify that it should not do so. Standards-conformant SIP servers (that support RFC 3327) might attempt to send requests to the allocated mapping port if the Path header is absent.



**Note:** ACL entries in the NAT table that permit TCP/TLS signaling for a SIP port configuration with TCP/TLS port mapping are the same as they would be for a TCP/TLS SIP port without port mapping enabled. Additional ACL entries that need to be set up for UDP port mapping are not required for TCP/TLS port mapping.

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### SIP Port Mapping Configuration for TCP TLS

You enable TCP/TLS port mapping in a per-realm basis using the SIP interface configuration; setting the **tcp-port-mapping** value in the **options** parameter enables the feature. Enabling this parameter turns on the port mapping feature for UDP as well.

By default, the Oracle Communications Session Border Controller includes the Path header in Register requests it sends from that SIP interface. If you do not this header to be included, however, you can set the value as **tcp-port-mapping=nopath**.

To enable TCP/TLS port mapping for a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

## SIP Signaling Services

---

```
ACMEPACKET# configure terminal  
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **sip-interface** and press Enter. If you are adding this feature to a pre-existing configuration, you will need to select and edit it.

```
ACMEPACKET (session-router) # sip-interface  
ACMEPACKET (sip-interface) #
```

4. **options**—Set the options parameter by typing options, a Space, the option name **tcp-port-mapping** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (sip-interface) # options +tcp-port-mapping
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the realm configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save your work.

## SIP Configurable Route Recursion

---

When the Oracle Communications Session Border Controller routes SIP requests from a UAC to a UAS, it might determine that there are multiple routes to try based on a matching local policy. The Oracle Communications Session Border Controller recurses through the list of routes in a specific order according to your configuration and the quality of the match. There are other scenarios when a UAS replies with a 3xx Redirect response to the Oracle Communications Session Border Controller, the 3xx response can include multiple Contacts to which the request should be forwarded in a specific order. In both cases, the Oracle Communications Session Border Controller needs to recurse through a list of targets.

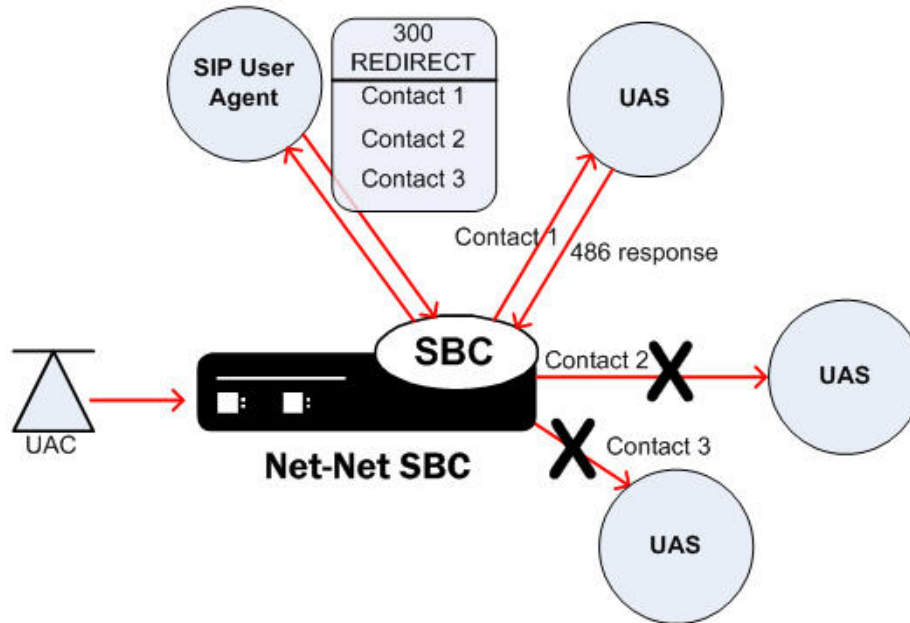
When the Oracle Communications Session Border Controller receives a non-successful (or non-6xx response) final response from the UAS, and there are multiple targets for the original request, the Oracle Communications Session Border Controller will forward the request to the next target and wait for a response. While the process of forwarding the request to multiple targets as explained in the previous paragraph is called serial forking, and the process of forwarding the request to contacts received in redirect responses is called recursion, the term recursion is used for both processes in this notice.

Use the SIP Route Recursion feature when you want the Oracle Communications Session Border Controller to forward a response to the UAC and stop recursing through the target list immediately after receiving the 3xx, 4xx, or 5xx response code that you configure. When this feature is disabled, the Oracle Communications Session Border Controller only stops recursing when it receives a message with a 401 or 407 response code. Using this feature, you can configure a specific message or range of messages to stop recursing on when received. The Oracle Communications Session Border Controller retains its default behavior to stop recursing on a 401 or 407 response code when SIP Route Recursion is configured on a SIP interface. The Oracle Communications Session Border Controller will always stop recursing when it receives a global failure (6xx); this behavior is not configurable.

You can disable response recursion for either a SIP interface or for a SIP session agent, providing you with flexibility for various network architectures. For instance, a PSTN gateway might be the only hop to reach a given endpoint, whereas several session agents might need to be contacted if multiple devices map to a contacted address of record.

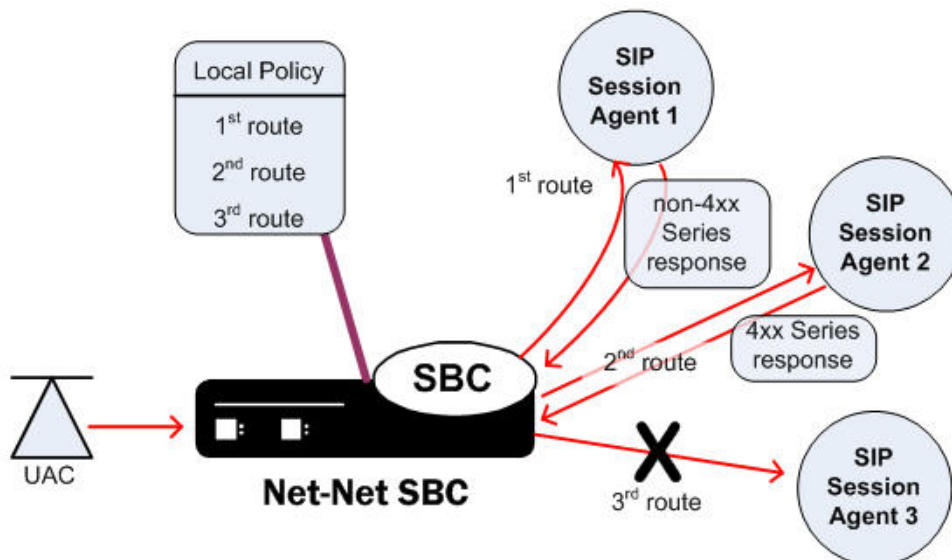
### Example 1

A more detailed example is when a softswitch might return a list of contacts for multiple PSTN gateways in a Redirect message. If the PSTN target number contacted on redirection is busy, a 486 response will be sent to the Oracle Communications Session Border Controller. Since the single target is located in the PSTN, a subsequent request through a different gateway will yield another 486 response. The Oracle Communications Session Border Controller should be configured to return the 486 response to the UAC immediately. No other SIP requests should be sent to applicable targets/contacts that were enumerated in the redirect list. See the following example:



## Example 2

The Oracle Communications Session Border Controller might determine from a local policy lookup that several routes are applicable for forwarding a SIP message. The Oracle Communications Session Border Controller will try each route in turn, but the SIP response recursion disable feature can be implemented to stop the route recursion when a configured responses message is received by the Oracle Communications Session Border Controller. See the following example:



There are a few conditions on the parameter used to configure response recursion:

- SIP Route Recursion is configurable for either the SIP interface or session agent.
- 401 and 407 are preconfigured for all configured SIP interfaces. They are not configured for session agents.
- The format is a comma-separated list of response codes or response code ranges: 404, 484-486.

- Only response codes that fall within the 3xx, 4xx, and 5xx range may be specified.

### SIP Route Recursion Configuration

You enable SIP route recursion either in the session agent or the SIP interface configuration.

#### Configuring a Session Agent for SIP Route Recursion

To configure SIP Route recursion for an existing session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the **session-router** path.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# session-agent  
ACMEPACKET(session-agent)#
```

4. Select the session agent where you want this feature.

```
ACMEPACKET(session-agent)# select  
<hostname>:  
1: asd          realm=      ip=1.0.0.0  
2: SIPSA       realm=      ip=10.10.102.1  
selection:2  
ACMEPACKET(session-agent)#
```

5. **stop-recurse**—Enter list of returned response codes that this session agent will watch for in order to stop recursion on the target's or contact's messages. This can be a comma-separated list or response code ranges.

```
ACMEPACKET(session-agent)# stop-recurse 404,484-486
```

6. Save and activate your changes.

#### Configuring a SIP Interface for SIP Route Recursion

To configure SIP route recursion for an existing SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the **session-router** path.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface  
ACMEPACKET(sip-interface)#
```

4. Select the SIP interface to which you want to apply this feature.

```
ACMEPACKET(sip-interface)# select  
<realm-id>:  
1: Acme_Realm  
selection:1  
ACMEPACKET(sip-interface)#
```

5. **stop-recurse**—Enter a list of returned response codes that this SIP interface will watch for in order to stop recursion on the target's or contact's messages. This list can be a comma-separated list of response codes or response code ranges.

```
ACMEPACKET(sip-interface)# stop-recurse 404,484-486
```

6. Save and activate your changes.



## SIP Event Package Interoperability

Service providers often deploy a Oracle Communications Session Border Controller on the border of an access network, where it sits between the SIP endpoints (user agents) and the service provider's application server. The application server and the user agents sometimes use various SIP event packages to exchange and maintain state information. The SUBSCRIBE and NOTIFY methods are used to establish subscriptions to the event packages and to report state changes to the subscribing entity.

The SIP global contact option addresses interoperability in the Dialog and Presence event packages that are used in hosted PBX and IP Centrex offerings. State information is passed in the message body of a NOTIFY request; this message body is encoded in an XML format described by the Content-Type header. The Oracle Communications Session Border Controller needs to update certain fields in the body to account for dialog mapping and SIP NAT functionality between the access and service provider realms. Often the subscriptions are established using URIs learned from Contact headers in the user agent registrations or dialog establishment (INVITE/SUBSCRIBE). For this, a Oracle Communications Session Border Controller requires a Contact URI that is usable and routable outside of an existing dialog.

The SIP global contact option enables persistent URIs in the Contact headers inserted into outgoing SIP messages. If this option is not used, URIs placed in the Contact header of outgoing messages are only valid within the context of the dialog to which the message is associated.

RFCs associated with this feature are:

- A. B. Roach, *Session Initiation Protocol (SIP)-Specific Event Notification*, RFC 3265, June 2002
- J. Rosenberg, *A Presence Event Package for the Session Initiation Protocol (SIP)*, RFC 3856, August 2004
- J. Rosenberg, et al. *Data Format for Presence Using XML*, <http://www.iptel.org/info/players/ietf/presence/outdated/draft-rosenberg-imp-pidf-00.txt>, Work In Progress (expired), June 2000
- J. Rosenberg, H. Schulzrinne, R. Mahy, *An INVITE Initiated Dialog Event Package for the Session Initiation Protocol (SIP)*, draft-ietf-sipping-dialog-package-06.txt, Work In Progress, April 2005
- H. Sugano, et al., *Presence Information Data Format (PIDF)*, RFC 3863, August 2004

## SIP Event Package Interoperability Configuration

This feature is applicable to the global SIP configuration.

To configure SIP event package interoperability:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

4. **options**—Add SIP event package interoperability support to a new SIP configuration or to an existing SIP configuration:

If you do not currently have an SIP configuration, you can add the option by typing **options**, a Space and then **global-contact**.

```
ACMEPACKET(sip-config)# options global-contact
```

Select the SIP configuration so that you can add SIP event package interoperability support to it. Then, to add this option to a list of options that you have already configured, type **options** followed by a Space, the plus sign (+), and the **global-contact** option.

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---

```
ACMEPACKET(sip-config) # select  
ACMEPACKET(sip-config) # options +global-contact
```

If you type `options global-contact` without the plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.

5. Save and activate your changes.

## SIP Proxy Subscriptions

---

When the Oracle Communications Session Border Controller operates in dialog mode (i.e., as a B2BUA), it creates and maintains dialog state for subscription dialogs created with SUBSCRIBE/NOTIFY messages and for INVITE-initiated dialogs. Since there can be a very large number of subscriptions per user in a Rich Communication Services (RCS) environment (especially for presence subscriptions), Oracle Communications Session Border Controller resources can quickly become depleted.

To alleviate this consumption of resources, you can configure your Oracle Communications Session Border Controller to operate in proxy mode for event packages that you define using the `proxy-sub-event` parameter in the global SIP configuration. When you define event packages in this list and the operation mode for the SIP configuration is dialog or session, the Oracle Communications Session Border Controller processes all SUBSCRIBE and NOTIFY requests and responses for the designated event packages in transaction stateful mode.

## Topology Hiding

So that it can perform topology hiding, the Oracle Communications Session Border Controller retains necessary routing information (such as the Contact or Record-Route header values) and it encodes certain data from the messages it receives in the messages it sends. To be more specific, the Oracle Communications Session Border Controller encodes the original URI hostport and the ingress realm name into a `gr` parameter it adds to the URI. The hostport information is replaced with the IP address and port of the SIP interface from which the message is sent (the egress interface). Without this information, subsequent in-dialog messages cannot be routed correctly because the Oracle Communications Session Border Controller does not retain dialog state (i.e., the route-set or remote-target).

For example, the URI `sip:td@192.168.24.121:5060` might be encoded as `sip:td@192.168.24.121:5060; gr=vjml9qtd175bhmhvhkqp0jov81popvbp000040`.

The Oracle Communications Session Border Controller also performs URI encoding on the message body for Content-Type `application/pdf+xml`. This contains a Presence Information Data Format document (or PIDF) in PUBLISH and NOTIFY requests so subsequent SIP requests using the URIs in the PIDF document can be routed correctly.

The Oracle Communications Session Border Controller performs URI encoding in outgoing SIP messages after SIP=NAT is applied and before outbound HMR occurs. And the system decodes URIs in SIP messages after inbound HMR takes place and before SIP-NAT is applied.


In the event a URI is encoded several times (as is the case in spiral and hairpin calls), the encoded realm+hostport values are separated by a plus sign (+), as in the following:

```
sip:td@192.168.24.121:5060; gr=vjml9qtd175bhmhvhkqp+dhfhhb0jov81popvbp
```

When the Oracle Communications Session Border Controller receives any of the following requests, it matches the contents of the request's Event header with the list you configure in the `proxy-sub-events` parameter:

- PUBLISH
- SUBSCRIBE
- NOTIFY
- REFER

This is provided the operation-mode for the SIP configuration is set to either `session` or `dialog`. If it finds a match, the Oracle Communications Session Border Controller marks the request for processing in transaction-stateful mode rather than in B2BUA mode.

 **Note:** Although PUBLISH is not a dialog-creating request, topology hiding needs to be applied to the PIDF so that subsequent NOTIFY requests containing portions of the published PIDF can be decoded properly.

When the Oracle Communications Session Border Controller forwards the request, it will have encoded all Contact and Record-Route header information using the ingress realm. The hostport value of the URIs then has egress SIP interface's IP address and port. The Via headers in the requested the Oracle Communications Session Border Controller received are not included in the outgoing request.

Then PUBLISH, SUBSCRIBE, NOTIFY, and REFER responses are compared to the request that was sent to determine if the response should receive transaction-stateful proxy treatment. The Oracle Communications Session Border Controller decodes any encoded Record-Route headers back to their original values for the outgoing response. Any Record-Route headers added downstream from the Oracle Communications Session Border Controller are encoded using the original request's egress realm (meaning the realm from which the response was received). In addition, the Contact header is encoded using the request's egress realm and ingress SIP interface and port.

### Feature Interaction

When using this feature, the Oracle Communications Session Border Controller does not keep dialog or subscription state. Therefore the Per-user SUBSCRIBE Dialog Limit feature—configured in the **enforcement-profile** configuration—will not function properly when a subscription is handled in proxy mode.

## SIP Proxy Subscription Configuration

This section shows you how to configure a list of SIP event packages to cause the Oracle Communications Session Border Controller to act in proxy mode.

 **Note:** The operation-mode parameter for the global SIP configuration must be set to either dialog or session in order for this feature to function as designed.

To configure a list of SIP event packages to enable SIP proxy subscriptions:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type `sip-config` and press Enter.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the ACLI **select** command) the configuration that you want to edit.

4. **proxy-sub-events**—Enter a list of SIP event package names that you want to enable the SIP proxy subscriptions feature. You can enter more than one value by enclosing multiple values in quotations marks, as in the following example.

```
ACMEPACKET(sip-config)# proxy-sub-events presence winfo
```

## SIP REGISTER Forwarding After Call-ID Change

This feature addresses the case when an endpoint reboots and performs a third party registration before its old registration expires. During this reregistration, the contact header is the same as it was pre-reregistration. As a consequence of the reboot, the SIP Call-ID changes. In this situation, the Oracle Communications Session Border Controller does not forward the REGISTER to the registrar, because it believes the endpoint is already registered, based on a previous registration from the same Contact: header URI.

To remedy this problem, the Oracle Communications Session Border Controller now keeps track of the Call-ID in its registration cache. The **forward-reg-callid-change** option in the global SIP configuration element forces the

## SIP Signaling Services

---

Oracle Communications Session Border Controller to forward a REGISTER message to the registrar when the Call-ID header changes in a REGISTER message received from a reregistering UAC.

### SIP REGISTER Forwarding Configuration

To configure SIP REGISTER forwarding after a Call-ID change:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure)# session-router  
ACMEPACKET (session-router)#
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router)# sip-config  
ACMEPACKET (sip-config)#
```

4. **options**—Add this feature to a new or an existing SIP configuration:

If you do not currently have a SIP configuration, you can add the option by typing **options**, a Space, and then **forward-reg-callid-change**.

```
ACMEPACKET (sip-config)# options forward-reg-callid-change
```

For an existing SIP configuration, select the SIP configuration so that you can add this feature to it. Then, to add this option to a list of options that you have already configured, type **options**, a Space, the plus sign (+), and the **forward-reg-callid-change** option.

```
ACMEPACKET (sip-config)# options +forward-reg-callid-change
```

If you type **options forward-reg-callid-change** without the plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.

5. Save and activate your changes.


## SIP Local Response Code Mapping

---

The SIP local response code mapping feature has been added as an enhancement to the SIP response code mapping. The SIP response code map feature lets you establish a table that maps SIP response-received messages (entries) to response-to-send messages (entries).

SIP local response code mapping is used with the SIP responses generated by the Oracle Communications Session Border Controller towards a specific SIP session agent. This feature lets you provision the mapping of the response codes used by the Oracle Communications Session Border Controller when it generates the responses towards a session agent.

You create the SIP local response code map using the existing mapping functionality, and then assigning that map to a session agent or to a SIP interface.

 **Note:** The configured response map is not used when the Oracle Communications Session Border Controller is acting as proxy for the responses to this session agent.

### SIP Local Response Code Mapping Configuration

The following instructions explain how to create the SIP response code map and then how to assign it to a specific session agent.

#### Creating a SIP Response Code Map

To create a SIP local response code map:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure)# session-router  
ACMEPACKET (session-router)#
```

3. Type **sip-response-map** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router)# sip-response-map  
ACMEPACKET (response-map)#
```

4. **name**—Enter the name of the SIP response map you want to configure. This value is required and must be unique.

```
ACMEPACKET (response-map)# name busy
```

5. **entries**—To configure the entries for this mapping, type **entries** and then press Enter. Typing a question mark will show you the response code entry parameters that you can configure.

```
ACMEPACKET (response-map)# entries  
ACMEPACKET (response-map-entries)#
```

**recv-code**—Enter original SIP response code for the **recv-mode** parameter. The valid range is:

- Minimum—100
- Maximum—699

```
ACMEPACKET (response-map-entries)# recv-mode 486
```

**xmit-code**—Enter the SIP response code into which you want the original response code to be translated. This valid range is:

- Minimum—100
- Maximum—699

```
ACMEPACKET (response-map-entries)# xmit-mode 600
```

**reason**—Enter a reason for the translated code into the **reason** parameter. This response comment is sent with the translated code. Make your entry in quotation marks.

```
ACMEPACKET (response-map-entries)# reason Busy Everywhere
```

The following two parameters (**method** and **register-response-expires**) enable a SIP registration response mapping feature that allows you to configure the Oracle Communications Session Border Controller to remap a SIP failure response—which it receives from another network device or that it generates locally—to a 200 OK. You might want the Oracle Communications Session Border Controller to perform this type of mapping for circumstances where non-malicious endpoints continually attempt registration, but will stop (and still not be registered) when they receive a 200 OK. This response mapping does not actually register the client with the Oracle Communications Session Border Controller, meaning that there is neither a registration cache entry or a CAM ACL for it.

For the 200 OK it generates, the Oracle Communications Session Border Controller removes any Reason or Retry-After header in the 200 OK and sets the expires time. By default, the expires time is the Retry-After time (if there is one in the response) or the expires value in the Register request (if there is no Retry-After expires time). You can also set this value using the **register-response-expires** parameter, but the value you set should never exceed the Register request's expires time.

**method**—Enter the name of the received SIP failure response message you want to map to a 200 OK. There is no default for this parameter, and leaving the parameter empty turns off the SIP registration response mapping feature.

**register-response-expires**—Enter the time you want to use for the expires time what mapping the SIP method you identified in the **method** parameter from Step 4. The maximum is 999999999. By default, the expires time is the Retry-After time (if there is one in the response) or the expires value in the Register request (if there is

no Retry-After expires time). Any value you configure in this parameter (when not using the defaults) should never exceed the Register request's expires time.

6. Note the name that you gave the SIP response code map so that you can use it when you configure a session agent to support SIP response code mapping.
7. Save and activate your changes.

### Assigning SIP Response Code Maps to Session Agents

To assign a SIP local response code map to a session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

4. **local-response-map**—Enter the name of the configured SIP response map that you want to use for this session-agent and press Enter.

```
ACMEPACKET(session-agent)# local-response-map busy
```

5. Save and activate your configuration.

### Assigning SIP Response Code Maps to SIP Interfaces

To apply SIP response codes maps to a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. **local-response-map**—Enter the name of the configured SIP response map that you want to apply to this SIP interface for locally-generated SIP responses. This parameter is blank by default.

5. Save and activate your configuration.

## Session Agent Ping Message Formatting

---

You can configure the user portions of the Request-URI and To: headers that define the destination of a session agent ping message, and the From: header that defines the source of a session agent ping message. These headers are sent to Oracle Communications Session Border Controller session agent. This feature is required for interoperability with certain E911 servers.

In the following example of a session agent ping-type message, you can set the user portion of the Request-URI (the text bob in the OPTIONS method line) and the user portion of the From: header (the text bob in the From: header) to the same new value. You can also set the user portion of the To: header (the text anna in the To: header) to its own new value.

```

OPTIONS sip:bob@sip.com SIP/2.0
From: UA1 <sip:bob@sip.com>
To: NUT <sip:anna@gw.sip.com>
Call-ID: 123abc@desk.sip.com
CSeq: 1 OPTIONS
Contact: <sip:UA1@client.sip.com>
Accept: application/sdp
Content-Length: 0

```

If you do not enable this feature, then the session agent ping-type message contains the text ping in all cases.

## Session Agent Ping Message Formatting Configuration

1. Access the **session-agent** configuration element.

```

ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)

```

2. Select the **session-agent** object to edit.

```

ACMEPACKET(session-agent)# select
<hostname>:
1: 192.168.100.101:1813

selection: 1
ACMEPACKET(session-agent)#

```

3. **ping-from-user-part**—Set the user portion of the From: header that defines the source of a session agent ping message.

```

ACMEPACKET(session-agent)# ping-from-user-part bob

```

4. **ping-to-user-part**—Set the user portions of the Request-URI and To: headers that define the destination of a session agent ping message.

```

ACMEPACKET(session-agent)# ping-to-user-part anna

```

5. Type **done** to save your configuration.

## SIP PAI Stripping

The Oracle Communications Session Border Controller now has the ability to strip P-Asserted-Identity (PAI) headers so that service providers can ensure an extra measure of security against malicious users pretending to be legitimate users. To pretend to represent another account, the malicious users simply send an INVITE with an imitation PAI. This feature allows real-time detection of such fraudulent use.

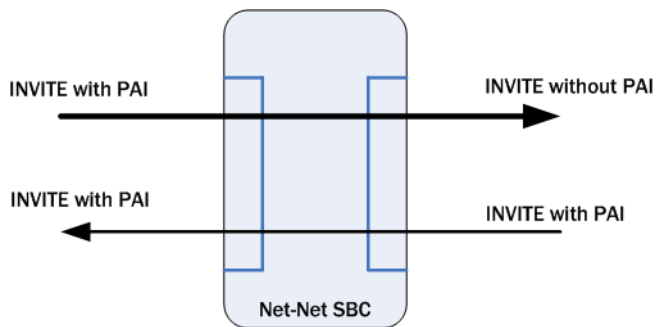
This feature uses a combination of:

- DoS protection applied on a per-realm basis
- SIP PAI header stripping

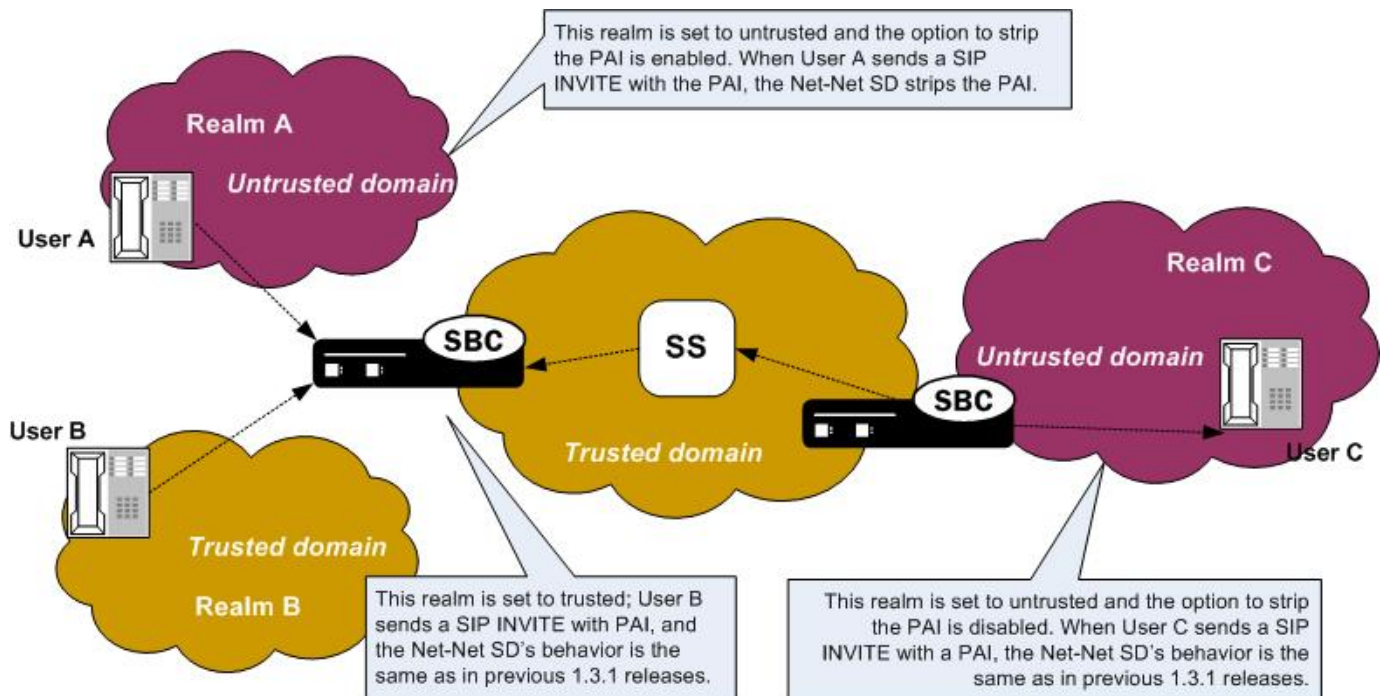
The combination of these settings can produce different results for the SIP PAI stripping feature.

- SIP PAI header stripping enabled for an untrusted realm—If the PAI stripping parameter is set to enabled in a realm that is untrusted, then the Oracle Communications Session Border Controller strips the PAI headers from SIP INVITEs that are received from the external address, regardless of the privacy type. The Oracle Communications Session Border Controller then sends the modified INVITE (without the PAI). If the INVITE comes from a trusted realm, then the Oracle Communications Session Border Controller does not strip the PAI header and the system behaves as it does when you are using previous 1.3.1 releases.

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- Multiple SIP PAIs in a SIP INVITE—The Oracle Communications Session Border Controller removes all PAIs when there are multiple PAIs set in SIP INVITES that come from untrusted realms.
- Oracle Communications Session Border Controller behavior bridging trusted and untrusted realms—The following graphics shows you how Oracle Communications Session Border Controllers can be positioned and configured to handle PAI stripping between trusted and untrusted realms.



Realm Configuration Settings	REALM A	REALM B	REALM C
Realm designation trusted or untrusted (trust-me)	Disabled	Enabled	Enabled
SIP PAI stripping (pai-strip)	Enabled	Enabled or disabled	Disabled

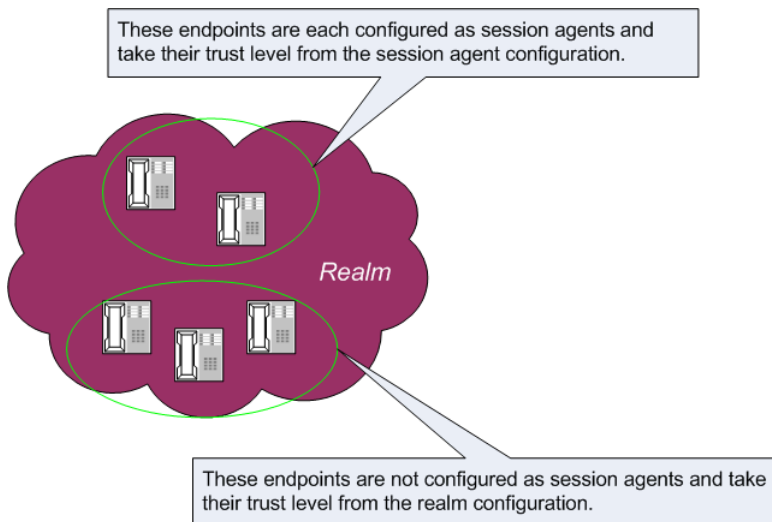


Realm Configuration Settings	REALM A	REALM B	REALM C
SBC's behavior	Strip PAI regardless of privacy type	Same as behavior for SIP privacy support in previous 1.3.1 releases	Same as behavior for SIP privacy support in previous 1.3.1 releases

## SIP PAI Stripping Configuration

When you configure this feature, please note how the Oracle Communications Session Border Controller behaves when you combine the designation of a realm as trusted/untrusted and SIP PAI stripping is enabled. Enter the choices for the ACLI **trust-me** and **pai-strip** parameters accordingly.

Be aware that trust is also established in the session agent configuration, and that the trust level set in a session agent configuration overrides the trust set in a realm configuration. For example, a realm might have several endpoints, some of which are associated with session agents and some of which are not. The endpoints that have configured session agent will take their trust level from the session agent parameters you set; the other endpoints, ones that are not associated with session agents, take their trust level from the realm parameters you set.



Take this relationship into consideration when you configure SIP PAI header stripping, or this feature will not work as designed.

For the sample configuration cited below, the desired Oracle Communications Session Border Controller behavior is to always strip the PAI regardless of privacy type.

To configure SIP PAI stripping for an existing realm using the ACLI:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-manager path.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

4. Select the realm to which you want to apply this feature.

```
ACMEPACKET (realm-config) # select
identifier:
1: acmePacket <none>          192.168.20.0/24
```

## SIP Signaling Services

---

```
2: realm1 <none> 0.0.0.0
selection:2
ACMEPACKET (realm-config) #
```

5. **pai-strip**—Enable PAI stripping. The default is **disabled**. Valid values are:

- enabled | disabled

```
ACMEPACKET (realm-config) # pai-strip enabled
```

6. Save your work using the ACLI **save** or **done** command.

## SIP Statuses to Q.850 Reasons

---

This section explains the Oracle Communications Session Border Controller's ability to map Q.850 cause values with SIP responses, a feature used in SIP calls and calls that require IWF.

RFC 3326 defines a header that might be included in any in-dialogue request. This reason header includes cause values that are defined as either a SIP response code or ITU-T Q.850 cause values. You can configure the Oracle Communications Session Border Controller to support sending and receiving RFC 3326 in SIP messages for:

- Mapping H.323 Q.850 cause values to SIP responses with reason header and cause value
- Mapping SIP response messages and RFC 3326 reason header and cause
- Locally generated SIP response with RFC 3326 reason header and cause

As specified in RFC 3326, the Oracle Communications Session Border Controller sends SIP responses to the softswitch that contain the received Q.850 cause code and the reason.

Though the Oracle Communications Session Border Controller can generate RFC 3326 headers, the default behavior for this feature is disabled. Furthermore, the Oracle Communications Session Border Controller can receive and pass SIP error messages (4xx, 5xx, and 6xx) that contain the SIP reason header with a Q.850 cause code and reason (as specified in RFC 3326). If the system receives an error message without the Reason header, then the Oracle Communications Session Border Controller is not required to insert one.

In calls that require IWF, the Q.850 cause generated in the SIP response are the same as the cause received in the following H.225 messages: Disconnect, Progress, Release, Release Complete, Resume Reject, Status, and Suspend Reject. In addition, the Q.850 cause codes that the Oracle Communications Session Border Controller receives in RFC 3326 headers are passed to the H.323 part of the call unmodified; the H.323 call leg uses this cause code for releasing the call.

## SIP-SIP Calls

The SIP Reason header might appear in any request within a dialog, in a CANCEL request, and in any response where the status code explicitly allows the presence of this header field. The syntax of the header follows the standard SIP parameter:

```
Reason: SIP;cause=200;text="completed elsewhere"
Reason: Q.850;cause=16;text="Terminated"
```

This feature attends to the following possible SIP call scenarios:

- When the Oracle Communications Session Border Controller receives a SIP request or SIP response that contains the Reason header, the Oracle Communications Session Border Controller passes it without modification.
- When it generates a SIP response, the Oracle Communications Session Border Controller includes the RFC 3326 Reason header containing a Q.850 cause code and reason. This is the case for all local conditions and for all internally generated error responses (4xx, 5xx, and 6xx) to an initial SIP INVITE.

Possible local error scenarios are:

- invalid-message
- cpu-overloaded
- media-released
- media-not-allocated

## SIP-SIP Calls Configuration

Configuring reason cause mapping for SIP-SIP calls requires that you set up the ACLI local-response-map configuration with appropriate entries; these generate the SIP response and the Q.850 cause code value to be used for particular error scenarios. If you want to add a Reason header, then you need to enable that capability in the global SIP configuration.

To configure a local response map:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **local-response-map** and press Enter.

```
ACMEPACKET(session-router)# local-response-map
ACMEPACKET(local-response-map)#
```

4. Type entries and press Enter.

```
ACMEPACKET(local-response-map)# entries
ACMEPACKET(local-response-map-entry)#
```

From here, you can view the entire menu for the local response map entries configuration by typing a **?**.

5. **local-error**—Set the local error that triggers the use of this local response map; there is no default for this parameter. Valid values are:

- **invalid-message**—Response map for invalid messages
- **cpu-overload**—Response map for CPU overload
- **enum-void-route**—Response map for when an ENUM server returns a ENUM+VOID response, or the local route table has 0.0.0.0 as the next hop
- **media-released**—Response map for media release conditions
- **media-not-allocated**—Response map for when media is not allocated

6. **sip-status**—Set the SIP response code to use. There is no default and the valid range is:

- Minimum—100
- Maximum—699

7. **sip-reason**—Set the SIP reason string you want to use for this mapping. There is no default value. If your value has spaces between characters, then your entry must be surrounded by quotation marks.

8. **q850-cause**—Set the Q.850 cause. There is no default value.

9. **q850-reason**—Set the Q.850 reason string that you want to use for this mapping. There is no default value. If your value has spaces between characters, then your entry must be surrounded by quotation marks.



### Note:

The following two parameters (method and register-response-expires) enable a SIP registration response mapping feature that allows you to configure the system to remap a SIP failure response—which it receives from another network device or that it generates locally—to a 200 OK. You might want the system to perform this type of mapping for circumstances where non-malicious endpoints continually attempt registration, but will stop (and still not be registered) when they receive a 200 OK. This response mapping does not actually register the client with the system, meaning that there is neither a registration cache entry or a CAM ACL for it.

For the 200 OK it generates, the system removes any Reason or Retry-After header in the 200 OK and sets the expires time. By default, the expires time is the Retry-After time (if there is one in the response) or the expires value in the Register request (if there is no Retry-After expires time). You can also set this value using the register-response-expires parameter, but the value you set should never exceed the Register request's expires time.

10. **method**—Enter the name of the received SIP failure response message you want to map to a 200 OK.  
There is no default for this parameter, and leaving the parameter empty turns off the SIP registration response mapping feature.
11. **register-response-expires**—Enter the time you want to use for the expires time what mapping the SIP method you identified in the method parameter from Step 4.  
The maximum is 999999999. By default, the expires time is the Retry-After time (if there is one in the response) or the expires value in the Register request (if there is no Retry-After expires time). Any value you configure in this parameter (when not using the defaults) should never exceed the Register request's expires time.
12. Repeat this process to create the number of local response map entries that you need.
13. Save and activate your configuration for changes to take effect.

### Adding the Reason Header

To enable the Oracle Communications Session Border Controller to add the Reason header:

1. In Superuser mode, type `configure terminal` and press Enter.  

```
ACMEPACKET# configure terminal
```
2. Type `session-router` and press Enter.  

```
ACMEPACKET(configure)# session-router
```
3. Type `sip-config` and press Enter.  

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```
4. **add-reason-header**—Enable this parameter to add the Reason header.  
The default value is disabled. The valid values are:
  - enabled | disabled
5. Save and activate your configuration for changes to take effect.

### Calls Requiring IWF

For interworking calls between SIP and H.323, you can configure:

- Mappings for SIP status codes to Q.850 values
- Mappings for particular Q.850 cause codes to SIP status codes

If it cannot find the appropriate mapping, then the Oracle Communications Session Border Controller uses default mappings defined in the Default Mappings table below.

The following describes how the Oracle Communications Session Border Controller handles different IWF call scenarios:

- **SIP request containing a Reason header**—When it receives a request containing a Reason header, the Oracle Communications Session Border Controller determines if the request is a SIP BYE or SIP CANCEL message. RFC 3326 states that the Reason header is mainly used for these types of requests. If there is a Reason header and it contains the Q.850 cause value, then the Oracle Communications Session Border Controller releases the call on the H.323 side using the specified cause value.
- **SIP response**—When it receives the error response to an initial SIP INVITE, the Oracle Communications Session Border Controller uses its SIP-Q.850 map to determine the Q.850 that it will use to release the call. If there is not a map entry, then the Oracle Communications Session Border Controller uses the default mappings shown in the Default Mappings table.
- **Active call released from the H.323 side**—If an active call is released from the H.323 side, the Oracle Communications Session Border Controller checks the outgoing realm (the SIP side) to see if the addition of the Reason header is enabled. If it is, then the Oracle Communications Session Border Controller adds the Reason header in the SIP BYE request with the Q.850 value it received from the H.323 side.
- **Error during setup of the call on the H.323 side**—In the event of an error during setup on the H.323 side of the call, the system needs to send:

- An error response, if this is a SIP to H.323 call
- A SIP CANCEL, if this is a H.323 to SIP call and the H.323 side hangs up before the call is answered on the SIP side

In this case, the Oracle Communications Session Border Controller checks to see if adding the Reason header is enabled in the IWF configuration. If it is, then the Oracle Communications Session Border Controller adds the Reason header with the Q.850 cause value it received from the H.323 side.

- Call released due to a Oracle Communications Session Border Controller error—If the call is released due a Oracle Communications Session Border Controller error and adding the Reason header is enabled in the IWF configuration, the error response to the initial INVITE contains the Reason header. The Oracle Communications Session Border Controller checks the SIP to Q.850 map configurations to determine whether or not the SIP error response code it is generating is configured. If it is, then the Oracle Communications Session Border Controller maps according to the configuration. If it is not, the Oracle Communications Session Border Controller derives cause mapping from the default table.

Like the configuration for SIP-only calls that enable this feature, you can set a parameter in the IWF configuration that enables adding the Reason header in the SIP requests or responses.

### Default Mappings

This table defines the default mappings the Oracle Communications Session Border Controller uses when it cannot locate an appropriate entry that you have configured.

Q.850 Cause Value		SIP Status		Comments
1	Unallocated number	404	Not found	
2	No route to specified transit network	404	Not found	
3	No route destination	404	Not found	
16	Normal calling clearing		BYE message	A call clearing BYE message containing cause value 16 normally results in the sending of a SIP BYE or CANCEL request. However, if a SIP response is to be sent to the INVITE request, the default response code should be used.
17	User busy	486	Busy here	
18	No user responding	408	Request timeout	
19	No answer from the user	480	Temporarily unavailable	
20	Subscriber absent	480	Temporarily unavailable	
21	Call rejected	603	Decline (if location filed in Cause information element indicates user; otherwise 403 Forbidden is used)	
22	Number changed	301	Moved permanently (if information in diagnostic field of Cause information element is suitable for generating SIP Contact	

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Q.850 Cause Value		SIP Status		Comments
			header, otherwise 410 Gone is used)	
23	Redirection to new destination	410	Gone	
25	Exchange routing error	483	Too many hops	
27	Destination out of order	502	Bad gateway	
28	Address incomplete	484	Address incomplete	
29	Facility rejected	501	Not implemented	
31	Normal, unspecified	480	Temporarily unavailable	
34	No circuit, channel unavailable	503	Service unavailable	
38	Network out of order	503	Service unavailable	
41	Temporary failure	503	Service unavailable	
42	Switching equipment congestion	503	Service unavailable	
47	Resource unavailable unspecified	503	Service unavailable	
55	Incoming calls barred with CUG	403	Forbidden	
57	Bearer capability not authorized	403	Forbidden	
58	Bearer capability not presently available	503	Service unavailable	
65	Bearer capability not implemented	488	Not acceptable here	
69	Requested facility not implemented	501	Not implemented	
70	Only restricted digital information available	488	Not acceptable here	
79	Service or option not implemented, unspecified	501	Not implemented	
87	User not member of CUG	403	Forbidden	
88	Incompatible destination	503	Service unavailable	
102	Recovery on timer expiry	504	Server time-out	

## SIP Status

To configure a SIP status to Q.850 Reason with cause mapping:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-q850-map** and press Enter.

```
ACMEPACKET(session-router)# sip-q850-map
ACMEPACKET(sip-q850-map)#
```

4. Type **entries** and press Enter.

```
ACMEPACKET(sip-q850-map)# entries
ACMEPACKET(sip-q850-map-entry)#
```

From here, you can view the entire menu for the SIP status to Q.850 Reason with cause mapping entries configuration by typing a **?**.

5. **sip-status**—Set the SIP response code that you want to map to a particular Q.850 cause code and reason. There is no default, and the valid range is:
  - Minimum—100
  - Maximum—699
6. **q850-cause**—Set the Q.850 cause code that you want to map to the SIP response code that you set in step 5. There is no default.
7. **q850-reason**—Set the Q.850 reason corresponding to the Q.850 cause code that you set in step 6. There is no default. If your value has spaces between characters, then your entry must be surrounded by quotation marks.
8. Repeat this process to create the number of local response map entries that you need.
9. Save and activate your configuration for changes to take effect.

To configure a Q.850 cause to a SIP status with reason mapping:

10. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

11. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

12. Type **sip-q850-map** and press Enter.

```
ACMEPACKET(session-router)# q850-sip-map
ACMEPACKET(q850-sip-map)#
```

13. Type **entries** and press Enter.

```
ACMEPACKET(q850-sip-map)# entries
ACMEPACKET(q850-sip-map-entry)#
```

From here, you can view the entire menu for the Q.850 cause to a SIP response code with reason mapping entries configuration by typing a **?**.

14. **q850-cause**—Set the Q.850 cause code that you want to map to a SIP status with reason. There is no default.
15. **sip-status**—Set the SIP response code to which you want to map the Q.850 cause that you set in step 5. There is no default, and the valid range is:
  - Minimum—100
  - Maximum—699
16. **sip-reason**—Set the reason that you want to use with the SIP response code that you specified in step 6. There is no default. If your value has spaces between characters, then your entry must be surrounded by quotation marks.
17. Repeat this process to create the number of local response map entries that you need.

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---

18. Save and activate your configuration for changes to take effect.

To enable the Oracle Communications Session Border Controller to add the Reason header for calls that require IWF:

19. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

20. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

21. Type **iwf-config** and press Enter.

```
ACMEPACKET(session-router)# iwf-config
ACMEPACKET(iwf-config)#
```

22. **add-reason-header**—Enable this parameter to add the Reason header. The default is **disabled**. The valid values are:

- enabled | disabled

23. Save and activate your configuration for changes to take effect.

## Trunk Group URIs

---

The Oracle Communications Session Border Controller's trunk group URI feature, applicable for SIP and IWF signaling services, enables the capabilities related to trunk groups that are described in this section. This implementation follows the IPTEL draft Representing Trunk Groups in Tel/SIP Uniform Resource Identifiers (URIs) (draft-ietf-iptel-trunk-group-06.txt), and also supports more customized approaches.

- For a typical access call flow scenario, when the calling party's call arrives at the Oracle Communications Session Border Controller, the Oracle Communications Session Border Controller formulates a SIP INVITE message that it sends to a softswitch. The Oracle Communications Session Border Controller now supports a new URI contact parameter in the SIP request message so that service providers need to be able to:
  - Determine from where the Oracle Communications Session Border Controller received the call
  - Signal information about the originating gateway from a Oracle Communications Session Border Controller to a softswitch (e.g., an incoming trunk group or a SIP gateway to a Oracle Communications Session Border Controller)
- This feature supports the signaling of routing information to the Oracle Communications Session Border Controller from network routing elements like softswitches. This information tells the Oracle Communications Session Border Controller what egress route (or outgoing trunk groups) it should choose for terminating next hops/gateways. For this purpose, new SIP URI parameters in the Request-URI are defined. Additional URI parameters include the network context to identify the network in which the originating or terminating gateway resides.
- Especially important for large business applications, this feature can free Oracle Communications Session Border Controller resources by reducing the number of local policy, session agent, and session agent group configurations. By enabling the trunk group URI feature, the Oracle Communications Session Border Controller instead uses a routing scheme based on signaled SIP URI information.

## Terminology

The following IPTEL terms are used in the descriptions of and instructions for how to configure this feature:

- Trunk—In a network, a communication path connecting two switching systems used in the establishment of an end-to-end connection; in selected applications, it may have both its terminations in the same switching system
- Trunk group—A set of trunks, traffic engineered as a unit, for the establishment of connections within or between switching systems in which all of the paths are interchangeable except where sub-grouped
- Trunk group name—Provides a unique identifier of the trunk group; referred to as tgrp
- Trunk group context—Imposes a namespace by specifying a domain where the trunk groups are; also referred to simply as context



## Trunk Group URI Parameters

Trunk group URI parameters identify originating and terminating trunk group information in SIP requests.

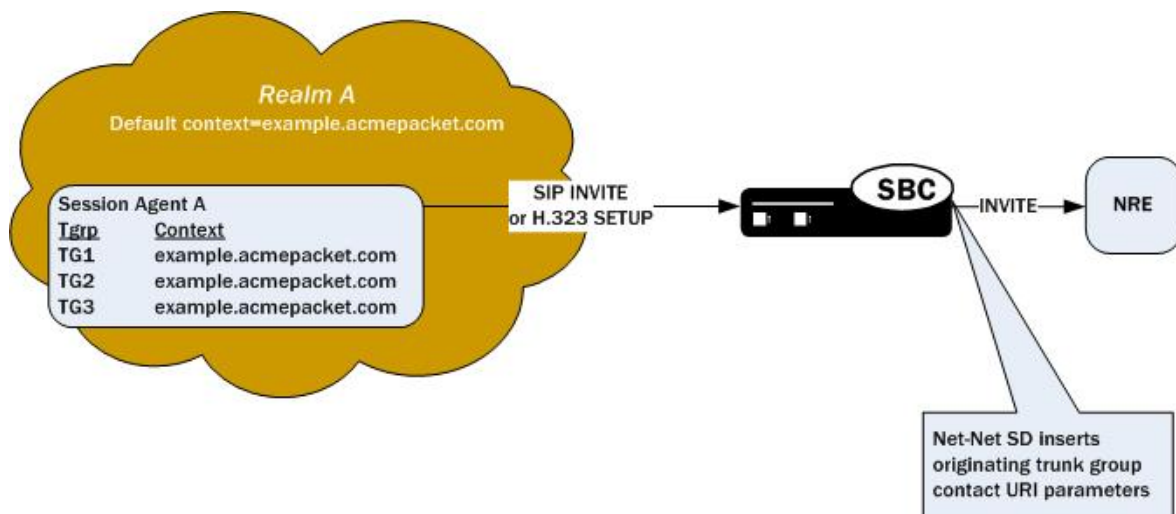
In the absence of official SIP standards for transporting trunk groups between signaling elements, the Oracle Communications Session Border Controller allows you to define URI parameters for use with originating and terminating trunk group URIs.

### Originating Trunk Group URI Parameters and Formats

You can configure session agents and session agents groups on the Oracle Communications Session Border Controller to insert trunk group URI parameters in the SIP contact header. When SIP gateways comply with the IPTEL draft, they include the originating URI parameter in the SIP contact header. For those SIP and H.323 gateways that are not compliant, the Oracle Communications Session Border Controller inserts SIP trunk group URI parameters on the gateway's behalf.

When there are no applicable session agent or session agent group configurations, the Oracle Communications Session Border Controller uses the source IP address of the endpoint or gateway as the trunk group name (tgrp) parameter in the originating trunk group URI.

The following diagram shows a scenario where the Oracle Communications Session Border Controller inserts originating trunk group URI parameters.



There are two available formats for the originating trunk group URIs:

1. In compliance with the IPTEL draft, the first format has two parameters: tgrp (identifier of the specific trunk group) and trunk-context (defines the network domain of the trunk group). These appear in the following formats:

- tgrp="trunk group name"
- trunk-context="network domain"

The URI BNF for would appear as it does in the example directly below, where the tgrp is tg55 and the trunk-context is trunk-context is telco.example.com:

```
tel:+15555551212;tgrp=tg55;trunk-context=telco.example.com
```

2. The second format is customized specifically for access URIs and contains two provisioned parameters: tgrp (or tgrname) and context (or provstring). This appears as tgrp.context, where these definitions apply:

- tgrp (tgrname)—Provisioned trunk group name for the originating session agent; this value must have at least one alphabetical character, cannot contain a period (.), and can contain a hyphen (-) but not as the first or the last character
- context (provstring)—Name of the originating trunk group context; this value must have at least one alphabetical character in the top label

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This format conforms to format for a hostname in the SIP URI as specified in RFC 3261, such that a trunk group identifier would appear as:

```
custsite2NY-00020.type2.voip.carrier.net
```

where the tgrp is custsite2NY-00020, and the context is type2.voip.carrier.net.

The BNF for an access URI conforms to the following:

```
SIP-URI = "sip:" [userinfo ] hostport uri-parameters [headers ]
```

```
uri-parameters = *( ";" uri-parameter )
```

```
uri-parameter = transport-param / user-param / method-param
```

```
/ ttl-param / maddr-param / lr-param / other-param
```

```
other-param = accessid / pname [ '=' pvalue ]
```

```
accessid = "access=" accessURI
```

```
accessURI = scheme tname [ "." provstring ]
```

```
scheme = "sip:" / token
```

```
tname = ALPHA / *(alphanumeric) ALPHA *(alphanumeric / "-") alphanumeric /
```

```
alphanumeric *(alphanumeric / "-") ALPHA *(alphanumeric) # up to 23 characters
```

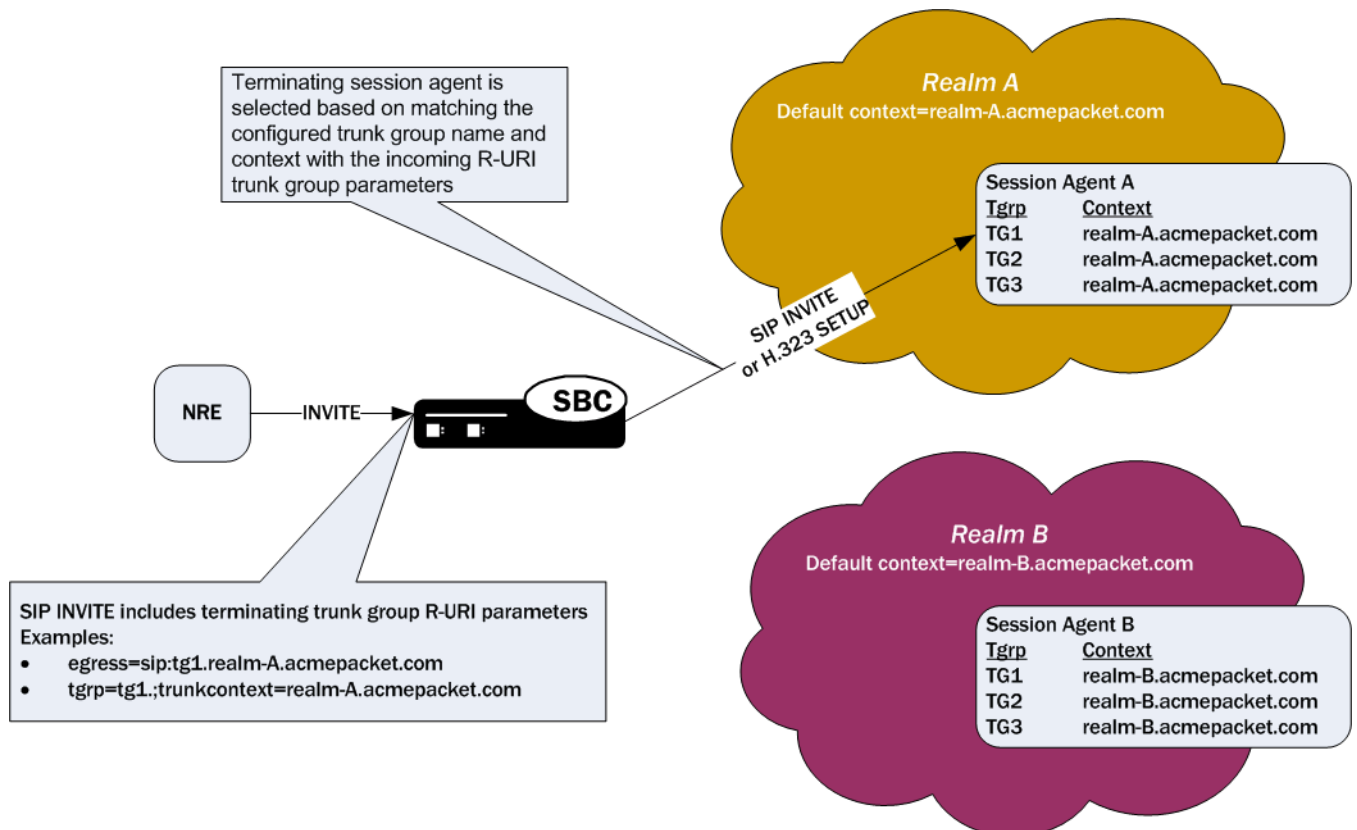
```
provstring = *(domain ".") toplevel # up to 24 characters
```

```
toplevel = ALPHA / ALPHA *( alphanumeric / "-") alphanumeric
```

```
domain = alphanumeric/ alphanumeric *( alphanumeric / "-") alphanumeric
```

### Terminating Trunk Group URI Parameters and Formats

Terminating trunk group URI parameters appear in the R-URI, and they can be included in by a network routing element to instruct the Oracle Communications Session Border Controller which egress trunk groups to use. By matching the trunk group URI parameter with configured session agents or session agent groups, the Oracle Communications Session Border Controller can locate the terminating gateway. The trunk group name can also be expressed as the IP address of the terminating gateway.



In the absence of official SIP standards for transporting trunk groups between signaling elements, the Oracle Communications Session Border Controller allows you to define the URI parameters used in terminating trunk groups.

There are two available formats for the terminating trunk group URIs:

1. In compliance with the IPTEL draft, the first format has two parameters: `tgrp` (which can be either a trunk group name or an IP address) and `trunk-context` (defines the network domain of the trunk group). These appear in the following formats:

- `tgrp="trunk group name"`
- `trunk-context="network domain"`

An example R-URI with terminating trunk group parameters appears as follows, where the `tgrp` is `TG2-1` and the context is `isp.example.net@egwy.isp.example.net`:

```
INVITE sip:+15555551212;tgrp=TG2-1;trunk-
context=isp.example.net@egwy.isp.example.net SIP/2.0
```

2. The second format is customized specifically for egress URIs and contains two provisioned parameters: `tgrp` (or `tgname`) and `context` (or `tgdomain`). This appears as `tgrp.context` (or `tgname.tgdomain`), where definitions apply:

- `tgrp (tgname)`—Provisioned trunk group name for the originating session agent; this value must have at least one alphabetical character, cannot contain a period (`.`), and can contain a hyphen (`-`) but not as the first or the last character
- `context (tgdomain)`—Name of the terminating trunk group context; this value can be up to twenty-four characters

The use of multiple terminating trunk groups is not supported.

The BNF for a single, egress URI with trunk group information conforms to:

```
SIP-URI = "sip:" [userinfo ] hostport uri-parameters [headers ]
uri-parameters = *( ";" uri-parameter )
uri-parameter = transport-param / user-param / method-param
                / ttl-param / maddr-param / lr-param / other-param
other-param = egressid / pname [ '=' pvalue ]
egressid = "egress=" egressURI
egressURI = scheme tgname [ "." tgdomain ]
scheme = "sip:" / token
tgname = ALPHA / *(alphanumeric) ALPHA *(alphanumeric / "-") alphanumeric /
        alphanumeric *(alphanumeric / "-") ALPHA *(alphanumeric) # up to 23 characters
tgdomain = *(domain ".") toplabel # up to 24 characters
toplabel = ALPHA / ALPHA *( alphanumeric / "-") alphanumeric
domain = alphanumeric / alphanumeric *( alphanumeric / "-") alphanumeric
```

For all trunk group URI support, you must set the appropriate parameters in the SIP manipulations configuration and in the session agent or session agent group configurations.

In the originating trunk group URI scenario, a call arrives at the Oracle Communications Session Border Controller from a configured session agent or session agent group. If this session agent or session agent group has the appropriate trunk group URI parameters and inbound manipulation rules configured, the Oracle Communications Session Border Controller then looks to the SIP manipulations configuration and add the trunk group URI information according to those rules. Those rules tell the Oracle Communications Session Border Controller where and how to insert the trunk group URI information, and the system forwards the call.

In the terminating trunk group scenario, a call arrives at the Oracle Communications Session Border Controller from, for instance, a call agent. This call contains information about what trunk group to use. If the information matches a session agent or session agent group that has outbound manipulation rules configured, the Oracle Communications Session Border Controller will then look up the SIP manipulations configuration and strip information according to those rules. Those rules tell the Oracle Communications Session Border Controller where and how to remove the information, and the Oracle Communications Session Border Controller forwards the call.

### SIP Header and Parameter Manipulation

SIP header and parameter manipulation is its own configuration where you can set up rules for the addition, removal, and modification of a SIP header or the elements of a SIP header. For example, you can set up the configuration to add a URI parameter to the URI in a SIP header or replace an FQDN with an IP address. For trunk group URI support, this configuration tells the Oracle Communications Session Border Controller where and how to manipulate the SIP message to use originating (access) and terminating (egress) trunk group URI parameters.

These manipulations can be applied at the realm or at the session agent level.

### Trunk Group Routing

You can configure SIP interfaces (using the ACLI **term-tgrp-mode** parameter) to perform routing based on the trunk group information received in SIP requests. There are three options: none, IPTEL, and egress URI.

- If you leave this parameter set to none (its default), the Oracle Communications Session Border Controller will not look for or route based on terminating trunk group URI parameters
- When you set this parameter to either **iptel** or **egress-uri** and the incoming request has the trunk group parameter of this type (IPTEL or egress URI), the Oracle Communications Session Border Controller will select the egress next hop by matching the “tgrp” and trunk context with a configured session agent or session agent group.

If the received terminating trunk group URI parameters include an IP address, the egress next hop is the IP address specified. The Oracle Communications Session Border Controller determines the egress realm by matching the trunk context it receives with the trunk context you configure for the realm.

- If the incoming request does not have trunk group parameters or it does not have trunk group parameters of the type that you configure, the Oracle Communications Session Border Controller uses provisioned procedures and/or local policy for egress call routing.

The Oracle Communications Session Border Controller returns errors in these cases:

- If the terminating trunk group URI parameters do not identify a local Oracle Communications Session Border Controller session agent or session agent group, then the Oracle Communications Session Border Controller returns a SIP final response of 488 Not Acceptable Here.
- If the Oracle Communications Session Border Controller receives a SIP INVITE with terminating trunk group URI parameters that do not match the specified syntax, the Oracle Communications Session Border Controller returns a 400 final response with the reason phrase Bad Egress=Parameters.

### Trunk Group URIs and SIP Registration Caching

For calls where SIP registration caching is used, you will need to set certain parameters that enable the Oracle Communications Session Border Controller to preserve trunk group URI parameters on the outgoing side.

- For SIP-SIP calls, you set the preserve-user-info **option** in the SIP interface configuration.
- For SIP-H.323 calls requiring IWF, you set the preserve-user-info-sa **option** in the session agent configuration.

## Trunk Group URI Configuration

Before you configure your Oracle Communications Session Border Controller to support trunk group URIs, you need to determine:

- How you want to manipulate SIP headers (entered in the SIP header manipulations configuration)
- For terminating trunk group routing, the trunk group mode you want to use (none, IPTEL, or egress URI); this decides routing based on trunk group information
- The trunk group name and context to use entered in a session agent or session agent group configuration
- Whether you are using originating or terminating trunk group URIs (entered in the session agent configuration)
- The trunk group context for use in a realm configuration, in case the trunk group name in the session agent or session agent group does not have a context

## Configuring SIP Manipulations

When you configure the SIP header manipulations to support trunk group URIs, take note of:

- The name of the configuration, so that you can use it when you apply the manipulations in a session agent for the inbound or outbound manipulations
- The **new-value** parameter, which specifies the trunk group and trunk group context that you want to manipulate; the possible values that apply to trunk group URI configurations are \$TRUNK\_GROUP and \$TRUNK\_GROUP\_CONTEXT

## Setting the Trunk Group URI Mode for Routing

To set the mode for routing for terminating trunk group URIs:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. **term-tgrp-mode**—Set the mode that you want to use for routing for terminating trunk group URIs. The default is **none**. Your choices are:

- **none**—Disables routing based on trunk groups
- **iptel**—Uses trunk group URI routing based on the IPTEL formats
- **egress-uri**—Uses trunk group URI routing based on the egress URI format

## Configuring a Session Agent for Trunk Group URIs

In a session agent, you can configure the outbound or inbound SIP header manipulation rules to use, as well as a list of trunk group names and contexts. For the trunk group names and contexts, you can use either the IPTEL or the custom format.

To configure a session agent for trunk group URIs:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

4. **out-manipulationid**—Enter the name of the SIP header manipulations configuration that you want to apply to the traffic exiting the Oracle Communications Session Border Controller via this session agent. There is no default.
5. **in-manipulationid**—Enter the name of the SIP header manipulations configuration that you want to apply to the traffic entering the Oracle Communications Session Border Controller via this session agent. There is no default.
6. **trunk-group**—In either IPTEL or custom format, enter the trunk group names and trunk group contexts to match. If you do not set the trunk group context, then the Oracle Communications Session Border Controller will use the one you set in the realm for this session agent.

Your ACLI entries for this list must be one of these formats: tgrp:context or tgrp.context.

To make multiple entries, surround your entries in parentheses and separate them from each other with spaces. For example:

```
ACMEPACKET(session-agent)# trunk-group (tgrp1:context1 tgrp2:context2)
```

7. **options**—If you want to configure trunk group URIs for SIP-H.323 calls that use the IWF and you are using SIP registration caching, you might need to add the `preserve-user-info-sa` to your list of session agent options.

If you are adding this option to a new session agent, you can just type **options**, a Space, and **preserve-user-info-sa**.

**If are adding this to an existing session agent, you must type a plus (+) sign before the option or you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign: options +preserve-user-info-sa.**

### Configuring a Session Agent Group for Trunk Group URIs

In a session agent group, you can configure the outbound or inbound SIP header manipulation rules to use, as well as a list of trunk group names and contexts. For the trunk group names and contexts, you can use either the IPTEL or the custom format.

To configure a session agent group for trunk group URIs:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-group** and press Enter.

```
ACMEPACKET(session-router)# session-group
ACMEPACKET(session-agent-group)#
```

4. **trunk-group**—In either IPTEL or custom format, enter the trunk group names and trunk group contexts to match. If you do not set the trunk group context, then the Oracle Communications Session Border Controller will use the one you set in the realm for this session agent group.

Your CLI entries for this list must take one of these formats: `tgrp:context` or `tgrp.context`.

To make multiple entries, surround your entries in parentheses and separate them from each other with spaces. For example:

```
ACMEPACKET(session-agent-group)# trunk-group (tgrp1:context1 tgrp2:context2)
```

### Setting a Trunk Group Context in a Realm

You can set trunk group contexts at the realm level, which will be used by all session agents and session agent groups if there is no context specified in their configurations.

The realm trunk group URI context accommodates the IPTEL and the custom format.

To configure a trunk group context for a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the session-related configurations.

```
ACMEPACKET(configure)# media-manager
```

3. Type **realm-config** and press Enter.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. **trunk-context**—Enter the trunk group context to use for this realm. There is no default.

## Using this Feature with a SIP Interface

If you are using the trunk group URIs feature with SIP interface that has registration caching enabled, then you need to configure the `preserve-user-info` option for that SIP interface.

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter to access the session-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type `session-group` and press Enter.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. **options**—Add support for trunk group URIs with SIP interface that uses registration caching.

If you are adding this option to a new SIP interface, you can just type `options`, a Space, and `preserve-user-info`.

**If are adding this to an existing SIP interface, you must type a plus (+) sign before the option or you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign: `options +preserve-user-info`.**

### Example 1 Adding Originating Trunk Group Parameters in IPTEL Format

This ACLI sample shows you how the ACLI SIP manipulations might appear in a case where you want to add originating trunk parameters in IPTEL format.

```

sip-manipulation
  name                               add_ipTEL
  header-rule
    name                               contact
    action                             manipulate
    match-value
    msg-type                           any
    element-rule
      name                               tgrp
      type                               uri-user-param
      action                             add
      match-val-type                    any
      match-value
      new-value                          $TRUNK_GROUP
element-rule
  name                               trunk-context
  type                               uri-user-param
  action                             add
  match-val-type                    any
  match-value
  new-value                          $TRUNK_GROUP_CONTEXT

```

### Example 2 Adding Originating Trunk Group Parameters in Custom Format

This ACLI sample shows you how the ACLI SIP manipulations might appear in a case where you want to add originating trunk parameters in custom format.

```

sip-manipulation
  name                               add_att
  header-rule
    name                               contact
    action                             manipulate
    match-value
    msg-type                           any
    element-rule

```

```
name egressURI
type uri-param
action add
match-val-type any
match-value
new-value "sip:"+$TRUNK_GROUP
+"."+$TRUNK_GROUP_CONTEXT
```

### Example 3 Removing IPTEL Trunk Group Names

This ACLI sample shows you how the ACLI SIP manipulations might appear in a case where you want to remove IPTEL trunk groups names.

```
sip-manipulation
  name strip_ip_tel
  header-rule
    name request-uri
    action manipulate
    match-value
    msg-type any
    element-rule
      name tgrp
      type uri-user-param
      action delete-element
      match-val-type any
      match-value
      new-value
element-rule
  name trunk-context
  type uri-user-param
  action delete-element
  match-val-type any
  match-value
  new-value
```

### Example 4 Removing Custom Trunk Group Names

This ACLI sample shows you how the ACLI SIP manipulations might appear in a case where you want to remove custom trunk groups names.

```
sip-manipulation
  name strip_egress
  header-rule
    name request-uri
    action manipulate
    match-value
    msg-type any
    element-rule
      name egressURI
      type uri-param
      action delete-element
      match-val-type any
      match-value
      new-value
```

---

## Emergency Session Handling

The Oracle Communications Session Border Controller provides a mechanism to handle emergency sessions from non-allowed endpoints. An endpoint is designated as non-allowed if it fails the admission control criteria specified by the allow-anonymous parameter in the SIP Ports configuration element.

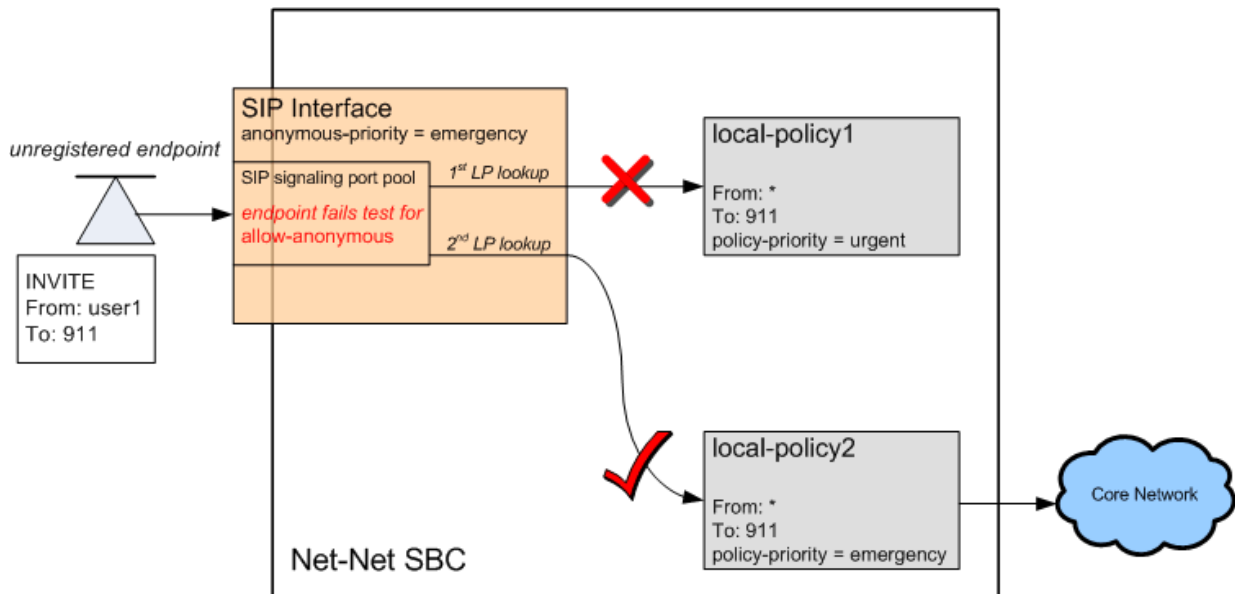


When the Oracle Communications Session Border Controller receives a non-allowed emergency request, it performs a local policy lookup for a matching local policy. An emergency local policy could be configured to match if the To: header in a SIP message was addressed to 911.

An emergency policy priority selection criteria has been added to both the SIP interface and the local policy configuration elements. In the SIP interface, the parameter is called anonymous-priority. In the local policy, the parameter is called policy-priority.

For the Oracle Communications Session Border Controller to choose a local policy to route an emergency call, the emergency policy priority value on the local policy must be equal to or greater than the emergency policy priority value on the SIP interface where the emergency message was received. In this scheme, an emergency policy priority value of none is the lowest value and an emergency policy priority value of emergency is the highest.

When a match is made between all existing local policy criteria and the emergency policy priority, the emergency call will be sent to the core network according to the chosen local policy. In addition, the policy priority value of the chosen local policy is inserted into the Priority header of the core-bound SIP message..



## Emergency Session Handling Configuration Procedures

Note the value of the allow-anonymous parameter in the SIP interface's SIP Ports for the incoming interface you are configuring. When an incoming emergency call from an unregistered endpoint can not be characterized by this setting, the Oracle Communications Session Border Controller will use the following means to route the call.

Set the anonymous-priority parameter in the incoming SIP interface. This parameter specifies that for an INVITE received from an anonymous endpoint, the Oracle Communications Session Border Controller will choose a local policy of equal or greater policy priority for outbound routing.

Next, set the policy-priority parameter located in the local-policy configuration element. Most likely, this local policy will route messages to SIP devices that act on emergency calls. The local policy is selected when its value (or above) matches the anonymous-priority parameter in the sip-interface that receives the incoming phone call from an unregistered endpoint.

The enumerated values for both the anonymous-priority and policy-priority are: none, normal, non-urgent, urgent, emergency.

## Emergency Session Handling Configuration

To set the anonymous priority for a message received in a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

## SIP Signaling Services

---

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-level configuration elements.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-interface  
ACMEPACKET (sip-interface) #
```

4. Type **select** and the number of the SIP interface you want to configure.

```
ACMEPACKET (sip-interface) # select 1
```

5. **anonymous-priority**—Set the policy priority for this SIP interface. It is used to facilitate emergency sessions from unregistered endpoints. This value is compared against the policy-priority parameter in the local-policy configuration element. The default is none. The valid values are:

- none | normal | non-urgent | urgent | emergency

This completes the configuration.

```
ACMEPACKET (sip-interface) # anonymous-priority emergency
```

6. Save your work using the ACLI **done** command.

### Setting Policy Priority

To set the policy priority for a local policy:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-level configuration elements.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **local-policy** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # local-policy  
ACMEPACKET (local-policy) #
```

4. Type **select** and the number of the local policy you want to configure.

```
ACMEPACKET (local-policy) # select 1
```

5. **policy-priority**—Enter the policy priority for this local policy. It is used to facilitate emergency sessions from unregistered endpoints. This value is compared against the anonymous-priority parameter in the sip-interface configuration element. The default is none. The valid values are:

- none | normal | non-urgent | urgent | emergency

This completes the configuration.

```
ACMEPACKET (local-policy) # anonymous-priority emergency
```

6. Save your work using the ACLI **done** command.

## Fraud Prevention

---

The Oracle Communications Session Border Controller can constrain outgoing SIP messages to a maximum size in bytes in order to support fraud prevention techniques. If a message does exceed the configured size, it is dropped. A SIP message can be constrained from 0 to 65535 bytes, with a default value of 4096 bytes.

## Fraud Prevention Configuration

To set a maximum SIP message size:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

4. Type **select** to configure the existing sip config.

```
ACMEPACKET(sip-config)# select
```

5. **sip-message-len**—Set the size constraint in bytes of a SIP message. The default is **4096**. The valid range is:

- Minimum—0
- Maximum—65535

This completes the configuration.

```
ACMEPACKET(sip-config)# sip-message-len 5000
```


6. Save your work using the ACLI **done** command.

## SIP Early Media Suppression

This section explains how to configure SIP early media suppression, which lets you determine who can send early media and in what direction. Early media are the RTP/RTCP packets sent from the called party to the caller, or vice versa, before a session is fully established (before a 200 OK is received). When the Oracle Communications Session Border Controller receives an INVITE message with SDP, it can forward media packets to the calling endpoint as soon as it forwards the INVITE to the next hop. It can also forward media packets received from the calling endpoint to the called endpoint as soon as the Oracle Communications Session Border Controller receives SDP in a SIP response to the INVITE, usually a provisional message. This allows for any early media to be played, such as remote ringback or announcement.

Early media can be unidirectional or bidirectional, and can be generated by the caller, the callee, or both.

With early media suppression, you can block early media until the call is established. You can define which outbound realms or next hop session agents are allowed to send or receive early media. early media suppression only applies to RTP packets. RTCP packets received by Oracle Communications Session Border Controller are still forwarded to their destination in both directions, unless an endpoint is behind a NAT and the media manager has not been enabled for RTCP forwarding.

 **Note:** To use early media suppression, you cannot configure media release of any kind: same-realm, same-network, or multiple-system media release.

With the SIP-based addressing, early media suppression is based on the outbound SIP interface realms and the value of their early-media-allow parameter. When the Oracle Communications Session Border Controller forwards a SIP Invite out a SIP interface, the outbound realm is chosen based on the SIP layer information, such as the session agent for the next-hop or the address prefix of the next-hop SIP device. The matching realm's early-media-allow parameter value then applies to either allow all, block all, or block one-way early media until a 200 ok is received. At that point bidirectional media is allowed. The decision is based on SIP-layer addressing of next-hops.

You configure a rule for a realm or a session agent to use early media suppression. An early media suppression rule specifies whether you want to prevent early media in any direction, allow early media going to the calling endpoint in

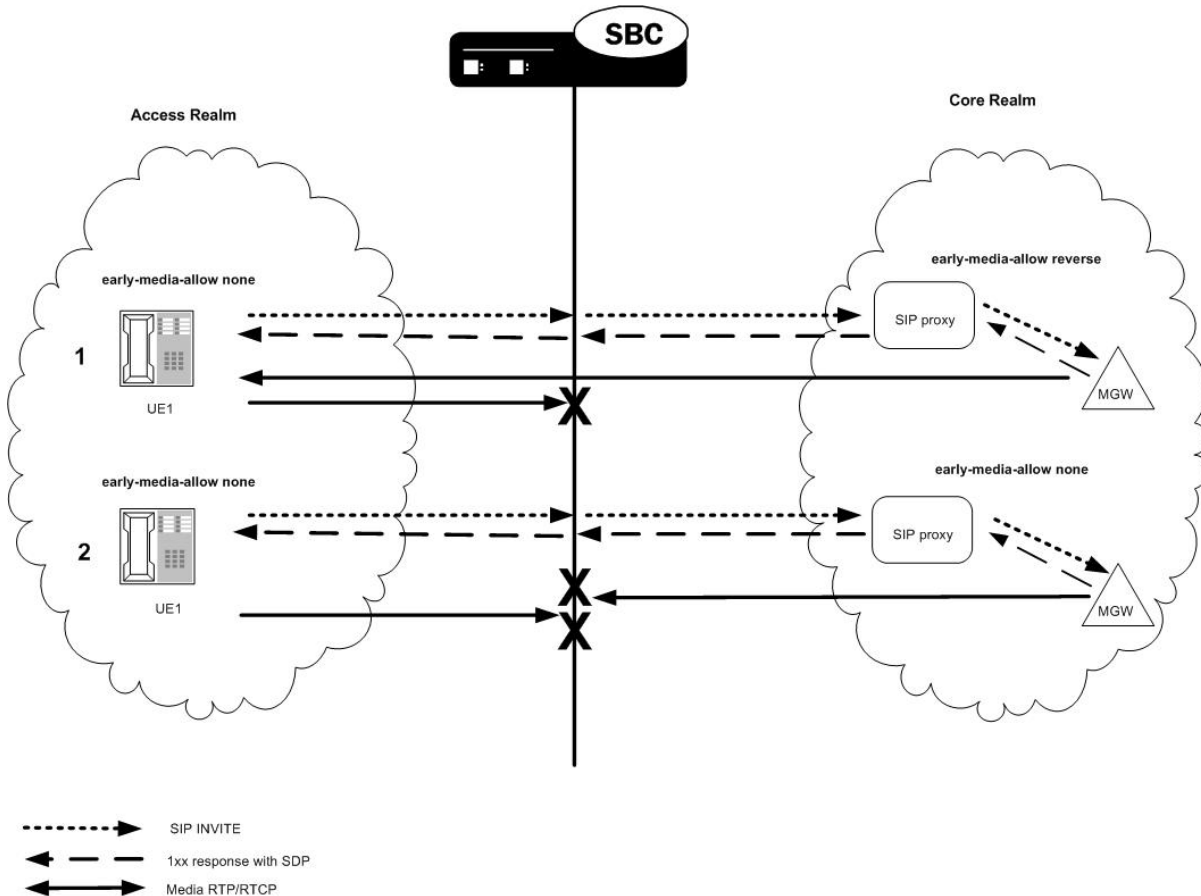
## SIP Signaling Services

the reverse direction, or allow early media in both directions. The forward direction is when the packets flow from the caller to the called party. The reverse direction is when the packets flow from the called party to the caller.

The early media suppression rule is applied to a session. When the Oracle Communications Session Border Controller initiates a new session, it first checks whether the next hop is a session agent and if so, whether an early media suppression rule has been configured it. If an early media suppression rule is found, the Oracle Communications Session Border Controller enforces it. If the next hop is not a session agent or no early media suppression rule is configured, the Oracle Communications Session Border Controller checks whether an early media suppression rule has been configured for the outbound realm. If it finds one, it enforces it.

### Example

The following illustration shows two examples of early media suppression.



1. Caller UE1 makes a call to the PSTN media gateway (MGW). The INVITE traverses from UE1 to the Oracle Communications Session Border Controller through the softswitch to the MGW. The Oracle Communications Session Border Controller allows early media from the core to reach UE1.
2. The PSTN MGW makes a call to UE1. The INVITE traverses to the Oracle Communications Session Border Controller and to UE1. The Oracle Communications Session Border Controller blocks all early media to and from UE1 until a 200 OK is received.

### Early Media Suppression Support

The Oracle Communications Session Border Controller supports suppressing early media in the following directions no matter which side makes the SDP offer, until it receives 200 OK for an INVITE:

- Forward direction based on the outbound realm or next-hop session agent
- Forward and reverse directions based on the outbound realm or next-hop session agent.

---

The Oracle Communications Session Border Controller allows all media when a 200 OK response is received for the INVITE, regardless of whether the 200 OK response contains SDP.

## Call Signaling

The Oracle Communications Session Border Controller media manager performs early media suppression according to an early media suppression rule. No change has been made to call signaling. For SIP, the Oracle Communications Session Border Controller still forwards SDP received in an INVITE request or response after performing a NAT to the media connection address. After which, the Oracle Communications Session Border Controller is ready to receive media packets from the endpoints. If an early media suppression rule has been configured, the Oracle Communications Session Border Controller drops the packets going in the direction being specified by the rule.

For a H.323 to SIP call, early media suppression rule does not change how the Oracle Communications Session Border Controller performs H.225/Q.931 call signaling and starts the H.245 procedure (if required) to establish logical channels for early media on the H.323 leg of the call.

## Suppression Duration

When early media suppression is enabled in a session, the block lasts until the session is established. For a SIP to SIP call or an H.323 to SIP call, a session is established when the system receives a 200 OK response to the INVITE. A 200 OK response to the INVITE terminates early media suppression, even when it does not contain a SDP. (A 200 OK response to a PRACK or an UPDATE request does not terminate early media suppression.) After a session is established, the Oracle Communications Session Border Controller can receive a change in media session (for example, a re-INVITE with a new SDP) without an early media suppression rule blocking the media.

## About the Early Media Suppression Rule

An early media suppression rule is configured in the form of a permission. It specifies whether early media is allowed in both directions, the reverse direction only or not at all. Reverse direction media is media sent in the upstream direction towards the calling endpoint.

### Session Agent Rule

The next-hop session agent's early media suppression rule is applied regardless of whether the media packet's source or destination address is the same as the session agent's address. For example, if the session's next hop session agent is 10.10.10.5 but the SDP in a 183 response specifies 10.10.10.6 as its connection address.

### Rule Resolution

When the call's next hop is a session agent and both the outbound realm of the call and the session agent have an early media suppression rule, the session agent's early media suppression rule takes precedence. If the session agent's early media suppression rule has not been configured, the outbound realm's early media suppression rule is used, if configured.

## Selective Early Media Suppression


Normally, the Oracle Communications Session Border Controller performs early media blocking based on destination realm. Calls to such realms are prohibited from sending and receiving RTP until a SIP 200 OK response is received, and you can set the direction of the blocked media.

While decisions to block early media are customarily based on SIP-layer addressing, there are cases when the Oracle Communications Session Border Controller can reject early media based on the SDP address in the SDP answer for a 1XX or 2XX response. By comparing the SDP address with the realm prefix or additional prefix address, it can block early media for matching realms. For these cases, you define global or signaling realms—ones that are not tied to SIP interfaces, but which establish additional address prefixes and rules for blocking early media.

This way, the Oracle Communications Session Border Controller blocks all early media for SIP interface realms, but can accept it for global realms that reference media or PSTN gateways. This configuration allows early media for calls destined for the PSTN, and blocks it for user-to-user and PSTN-to-user calls.

Selective early media suppression addresses the fact that some service providers need to allow early media for certain user-to-user and PSTN-to-user calls to support, for example, custom ringback tones. The enhancements also address the fact that Oracle Communications Session Border Controllers can themselves lose the ability to decide whether or not early media should be blocked when confronted with hairpinned call flows, or with traffic that traverses multiple Oracle Communications Session Border Controllers.

To address this need, you can configure realm groups. Realm groups are sets of source and destination realms that allow early media to flow in the direction you configure. For example, you can set up realm groups to allow media from PSTN realms to user realms so that users can listen to PSTN announcements, but prohibit early media from user realms to PSTN realms.

 **Note:** The source and destination realms you add to your lists need to be a global signaling realm matching the caller's SDP address prefix or a SIP realm.

### Configuring the Realm

To configure the realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# media-manager
```

3. Type **realm** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# realm
ACMEPACKET(realm)#
```

4. If configuring an existing realm, enter the select command to select the realm.

5. **early-media-allow**—Enter the early media suppression rule for the realm. The valid values are:

- **none**—No early media is allowed in either direction
- **both**—Early media is allowed in both directions
- **reverse**—Early media received by Oracle Communications Session Border Controller in the reverse direction is allowed

There is no default value. If you leave this parameter blank, early media is allowed in either direction. You can use the following command to clear this parameter:

```
early-media-allow ()
```

6. Save and activate your configuration.

For example:

```
realm-config
  identifier                access1
  addr-prefix               192.168.1.0/24
  network-interfaces
  mm-in-realm               media:0
  mm-in-network             enabled
  msm-release               disabled
  qos-enable                disabled
  max-bandwidth             0
  max-latency               0
  max-jitter                0
  max-packet-loss           0
  observ-window-size        0
  parent-realm
  dns-realm
  media-policy
  in-translationid
```

```

out-translationid
class-profile
average-rate-limit          0
access-control-trust-level  none
invalid-signal-threshold   0
maximum-signal-threshold   0
deny-period                 30
early-media-allow          none
last-modified-date         2006-02-06 13:09:20

```

## Configuring Session Agents

If you do not configure early media suppression for a session agent, the early media suppression for the outbound realm is used, if configured.

To configure session agents:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

4. If configuring an existing session agent, enter the select command to select the session agent.
5. **early-media-allow**—Enter the early media suppression rule for the session agent. The valid values are:

- **none**—No early media is allowed in either direction
- **both**—Early media is allowed in both directions
- **reverse**—Early media received by Oracle Communications Session Border Controller in the reverse direction is allowed

There is no default value. If you leave this parameter blank, early media is allowed in either direction. You can use the following command to clear this parameter:

```
early-media-allow ()
```

6. Save and activate your configuration.

For example:

```

session-agent
  hostname          cust1
  ip-address        192.168.1.24
  port              5060
  state             enabled
  app-protocol      SIP
  app-type
  transport-method  UDP
  realm-id          access1
  description
  carriers
  allow-next-hop-lp enabled
  constraints        disabled
  max-sessions       0
  max-outbound-sessions 0
  max-burst-rate     0
  max-sustain-rate   0
  time-to-resume     0
  ttr-no-response    0
  in-service-period  0

```

```
burst-rate-window          0
sustain-rate-window        0
req-uri-carrier-mode       None
proxy-mode
redirect-action
loose-routing              enabled
send-media-session         enabled
response-map
ping-method
ping-interval              0
media-profiles
in-translationid
out-translationid
trust-me                   disabled
early-media-allow          reverse
last-modified-date         2006-05-06 13:26:34
```

### Configuring Realm Groups

To configure a realm group for selective early media suppression:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **realm-group** and press Enter.

```
ACMEPACKET(media-manager)# realm-group
ACMEPACKET(realm-group)#
```

4. **name**—Enter the name of the realm group.

5. **source-realm**—Enter the list of one or more global/SIP realms that you want to designate as source realms for the purpose of blocking early media; this is the realm identifier value for the realms you want on the list. Values in this list refer to calling SDP realms; this parameter has no default. To enter more than one realm in the list, list all items separated by a comma and enclose the entire entry in quotation marks:

```
ACMEPACKET(realm-group)# source-realm Private, Public
```

To add a realm to the list, use the plus sign (+) in front of each new entry.

```
ACMEPACKET(realm-group)# source-realm +Private
```

You can also remove single items in the list by using the minus sign (-) directly in front of the realm identifier.

```
ACMEPACKET(realm-group)# source-realm -Private
```

6. **destination-realm**—Enter the list of one or more global/SIP realms that you want to designate as destination realms for the purpose of blocking early media; this is the realm identifier value for the realms you want on the list. Values in this list refer to called SDP realms; this parameter has no default. To enter more than one realm in the list, list all items separated by a comma and enclose the entire entry in quotation marks:

7. **ACMEPACKET(realm-group)# source-realm Private, Public**

To add a realm to the list, use the plus sign (+) in front of each new entry.

```
ACMEPACKET(realm-group)# destination-realm +Private
```

You can also remove single items in the list by using the minus sign (-) directly in front of the realm identifier.

```
ACMEPACKET(realm-group)# destination-realm -Private
```

8. **early-media-allow-direction**—Set the direction for which early media is allowed for this realm group. Valid values are:

- **none**—Turns off the feature for this realm group by blocking early media



- reverse—Allows early media to flow from called to caller
  - both (default)—Allows early media to flow to/from called and caller
9. Save and activate your configuration.

## SDP-Response Early Media Suppression

This section explains how to configure SDP-response early media suppression, which can be used when the Oracle Communications Session Border Controller is deployed after a softswitch or proxy in the signaling path. In this deployment, user endpoints and gateways communicate directly with the softswitch or proxy, which in turn sends call signaling to the Oracle Communications Session Border Controller. The call signaling gets sent back to the same or different softswitch or proxy. Because the Oracle Communications Session Border Controller does not communicate with the endpoints or gateways that are the media terminators, early media suppression for this deployment must use SDP-based addressing rather than the SIP-based addressing (described in the SIP Early Media Suppression section in this technical notice).

Using this feature lets you configure specific IP addresses for which early media should not be suppressed, based on SDP addressing. The Oracle Communications Session Border Controller checks the SDP addresses in SIP responses against these IP address or address ranges to determine on which media gateway a call terminates.

### SIP-Based Addressing

With the SIP-based addressing described in the SIP Early Media Suppression section, early media suppression is based on the outbound SIP interface realms and the value of their early-media-allow parameter. When the Oracle Communications Session Border Controller forwards a SIP Invite out a SIP interface, the outbound realm is chosen based on the SIP layer information, such as the session agent for the next-hop or the address prefix of the next-hop SIP device. The matching realm's early-media-allow parameter value then applies to either allow all, block all, or block one-way early media until a 200 ok is received. At that point bidirectional media is allowed. The decision is based on SIP-layer addressing of next-hops.

### SDP-Based Addressing

SDP-response early media suppression follows the same sequence described for SIP-based addressing with one exception. A provisional response with SDP media can make the Oracle Communications Session Border Controller select a new early-media-allow rule from another realm, based on the addressing inside the responding SDP.

When the SDP-response early media suppression feature is enabled, the Oracle Communications Session Border Controller searches the outbound SIP interface's realms for a matching address prefix with the connection address in the responding SDP. If it finds a match, it uses the early-media-allow parameter value of that realm until the 200 OK message is received, then bidirectional media is allowed regardless. If the Oracle Communications Session Border Controller does not find a match, it searches all of the global realms for one. If it finds a match, the Oracle Communications Session Border Controller uses that realm's early-media-allow parameter value. If it does not find a match in the global realm(s), the Oracle Communications Session Border Controller continues to use the previous early-media-allow parameter value.

#### Global Realms

Global realms are realms that are not parents or children of any other realms, do not have defined SIP interfaces and ports (or any signaling interface or stack), and are configured to use the network interface lo0:0. They are special realms, applicable system-wide, and are currently only used for this feature. The only global realm configuration parameters applicable to early media suppression are:

- addr-prefix
- additional-prefixes
- early-media-allow
- network-interface (which must be set to lo0:0)

### Additional Prefixes

You can specify additional prefixes in addition to that of the `addr-prefix` parameter you configure for a realm. For example, you can configure a global realm with additional address prefixes to specify the IP addresses (or ranges of addresses) of the media gateways that are allowed to send and receive early media. This overrides the SIP interface realm's early media blocking settings.

You can also enter additional prefixes in non-global realms. These additional prefixes function the same as would multiple values in the `addr-prefix` parameter (which only takes one value), except addresses in additional-prefixes are not used for SIP NATs.

### Using the SDP-Response Early Media Suppression Rule

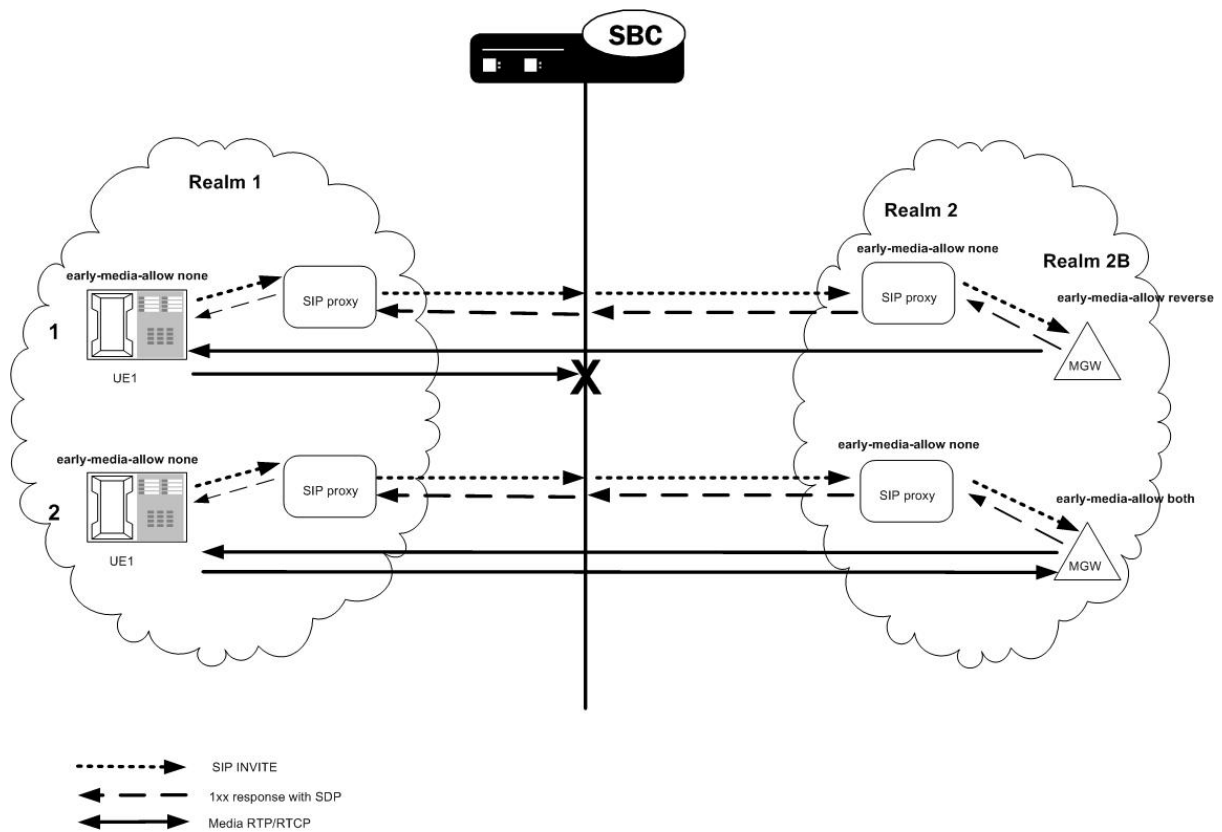
To use SDP-response early media suppression, you must add the `early-media-sdp-realms` option to the SIP interface configuration that interfaces with the next-hop device, such as the supported softswitch.

When the Oracle Communications Session Border Controller receives a provisional response that includes SDP from the called endpoint, and the `early-media-sdp-realms` option is active in the outgoing SIP interface of the call, it first searches the realms that apply to the outgoing SIP interface. If it does not find a realm, the Oracle Communications Session Border Controller searches the global realms. If the search yields a new realm that is not the SIP interface realm, its early media suppression rule (if any) replaces the existing one. Only the early media suppression rule of the new realm is applied to the call. Other realm properties from the outbound realm remain applicable to the call. If no new realm is found, the early media policy of the outgoing SIP interface realm is applied.

The Oracle Communications Session Border Controller allows media when the SDP media connect address in a response matches one of a configured list of IP address ranges defined in a realm and the realm has early media allowed. You need to configure specific IP address or address range to specify which media gateways should not be suppressed based on SDP media addresses. The IP addresses are checked against the SDP being received. The decision for suppression is based on whether the matching realm allows early media. The early media will be suppressed if the matching realm does not allow early media or if there is no match and the outbound SIP interface realm does not allow early media.

### Example

The following illustration shows two examples of SDP-response early media suppression.



## Configuring SDP-Response Early Media Suppression

To configure SDP-response early media suppression:

1. Add the `early-media-sdp-realms` option to the SIP interface that interfaces with the softswitch.
2. Configure the SIP interface realm with an early media suppression rule that blocks all early media.
3. Configure either or both of the following:
  - One or more of the SIP realm's child realms, each with an early media suppression rule that allows all or reverse direction early media and a list of additional prefixes that specifies the IP addresses of the media gateways, or a range of IP addresses that includes the media gateways. Early media is allowed from these gateways only for calls that signals through this SIP interface.
  - One or more realms that has the network interface equal to `lo0:0`, an early media suppression rule that allows all or reverse direction early media and a list of additional prefixes that specifies the IP addresses of the media gateways, or a range of IP addresses that includes the media gateways. Early media is allowed from these gateways regardless of interface.

## Configuring the SIP Interface

To configure a SIP interface:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type `sip-interface` and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

## SIP Signaling Services

From this point, you can configure SIP interface parameters. To view all sip-interface parameters, enter a ? at the system prompt.

4. If configuring an existing interface, enter the select command to select the interface.
5. **options**—Enter early-media-sdp-realms as the option. If adding to an existing list of options, use a preceding plus (+) sign.

```
options +early-media-sdp-realms
```

6. Continue to the next section to configure the outbound realm.

For example:

```
ACMEPACKET(session-router) # sip-interface
ACMEPACKET(sip-interface) # options + early-media-sdp-realms
ACMEPACKET(sip-interface) # done
sip-interface
  state                enabled
  realm-id             access1
  sip-port
    address            192.168.1.30
    port               5060
    transport-protocol UDP
    allow-anonymous    all
  carriers
  proxy-mode          Proxy
  redirect-action
  contact-mode        maddr
  nat-traversal       none
  nat-interval        30
  registration-caching disabled
  min-reg-expire      300
  registration-interval 3600
  route-to-registrar  disabled
  teluri-scheme       disabled
  uri-fqdn-domain
  options             early-media-sdp-realms
  trust-mode          all
  last-modified-date  2006-05-10 18:27:31
```

### Configuring a Realm

To configure a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure) # media-manager
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager) # realm-config
ACMEPACKET(realm-config) #
```

4. If configuring an existing realm, enter the select command to select the realm.
5. **early-media-allow**—Enter the early media suppression rule for the realm. The valid values are:
  - **both**—Early media is allowed in both directions
  - **reverse**—Early media received by Oracle Communications Session Border Controller in the reverse direction is allowed
  - **none**—Early media is blocked

6. **additional-prefixes**—Enter a single or a comma-delimited list of IP address prefixes to use in addition to the value of the **addr-prefix** parameter.

```
<IPv4> [ /<number of bits> ]
```

<IPv4> is a valid IPv4 address and <number of bits> is the number of bits to use to match an IP address with the address prefix. Not specifying <number of bits> implies that all 32 bits are used for matching.

Enclose the list between quotes if there is any space between a comma and the next address prefix.

You can add and remove address prefixes to and from the list:

- **add-additional-prefixes** adds one or more additional prefixes

```
add-additional-prefixes 192.168.201.69
```

- **remove-additional-prefixes** removes one or more additional prefixes

```
remove-additional-prefixes 192.168.201.69
```

If using multiple address prefixes, enter a comma-delimited list.

7. Save and activate your configuration.

For example:

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)# additional-prefixes
192.168.200.0/24,192.168.201.68
ACMEPACKET(realm-config)# done
realm-config
    identifier                    early-media
    addr-prefix                    0.0.0.0
    network-interfaces
    media2:0
    mm-in-realm                    disabled
    mm-in-network                  enabled
    msm-release                    disabled
    qos-enable                     disabled
    max-bandwidth                  0
    max-latency                    0
    max-jitter                     0
    max-packet-loss                0
    observ-window-size             0
    parent-realm
    dns-realm
    media-policy
    in-translationid
    out-translationid
    in-manipulationid
    out-manipulationid
    class-profile
    average-rate-limit             0
    access-control-trust-level     none
    invalid-signal-threshold       0
    maximum-signal-threshold       0
    untrusted-signal-threshold     0
    deny-period                    30
    symmetric-latching             disabled
    pai-strip                      disabled
    trunk-context
    early-media-allow              reverse
    additional-prefixes            192.168.200.0/24
192.168.201.69
    last-modified-date             2006-05-11 06:47:31
```

### SIP Duplicate SDP Suppression

---

Using the **strip-dup-sdp** option in the SIP configuration, you can enable your Oracle Communications Session Border Controller to suppress a duplicate SDP answer in the reliable responses (1xx and 2xx) in an INVITE transaction.

During INVITE transactions in certain networks, SDP answers in reliable provisional responses (1xx) can cause interoperability issues. This issue occurs when the UAS includes the SDP answer in subsequent 1xx responses or in the final 2xx response, and the UAC views that inclusion as a protocol violation. The UAC does so based on the fact that the SDP answer was reliably delivered to it in a 1xx response, and so it views additional SDP information as unnecessary in and ensuing reliable 1xx or 200 OK responses.

RFCs 3216 and 3262 do not specifically call out the UAS's including SDP information in this way as a protocol violation. Still, the system allows you set enable the **strip-dup-sdp** option as a means of preventing the UAC from terminating sessions. With this option enabled, the Oracle Communications Session Border Controller removes the SDP answer in subsequent reliable provisional or final 200 OK responses if it is identical to the SDP answer previously received.

### SIP Duplicate SDP Suppression Configuration

To enable duplicate SDP suppression for SIP sessions:

1. In Superuser mode, type configure terminal and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router  
ACMEPACKET(session-router)#
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET(session-router)# sip-config  
ACMEPACKET(sip-config)#
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the CLI **select** command) the configuration that you want to edit.

4. **options**—Set the options parameter by typing options, a Space, and then the option name **strip-dup-sdp**. Then press Enter.

```
ACMEPACKET(sip-config)# options +strip-dup-sdp
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

### SIP SDP Address Correlation

---

SIP SDP address correlation ensures that when the Oracle Communications Session Border Controller receives a request containing SDP, the L3 source address of the request is compared against the address in the c-line of the SDP. When the addresses match, the session proceeds as it normally would. If there is a mismatch, the call is rejected with the default 488 status code. You can also configure the code you want to use instead of 488.

This functionality works only with non-HNT users. The value c=0.0.0.0 is an exception and is always processed.

### SIP SDP Address Correlation Configuration Address Checking

The **sdp-address-check**, in the **enforcement-profile** element can be set to enable the SDP address correlation.

To enable SDP address checking:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **enforcement-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # enforcement-profile
ACMEPACKET (enforcement-profile) #
```

4. Use the ACLI **select** command so that you can work with the enforcement profile configuration to which you want to add this parameter.

```
ACMEPACKET (enforcement-profile) select
```

5. **sdp-address-check**—Enable or disable SDP address checking on the Oracle Communications Session Border Controller. The default for this parameter is **disabled**.

```
ACMEPACKET (enforcement-profile) # sdp-address-check enabled
```

6. Save and activate your configuration.

If a mismatch occurs and you want to reject the call with a status code other than 488, you set the code you want to use in the local response code map entries.

## SIP SDP Address Correlation Configuration Mismatch Status Code

To apply a new status code to a SDP address correlation mismatch:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **local-response-map** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # local-response-map
ACMEPACKET (local-response-map) #
```

4. Type **entries** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (local-response-map) # entries
ACMEPACKET (local-response-map-entry) #
```

From here, you can view the entire menu for the local response map entries configuration by typing a **?**.

5. **local-error**—Enter sdp-address-mismatch for which to apply the new status code.

6. **sip-status**—Set the SIP response code to use.

7. **sip-reason**—Set the SIP reason string you want to use for this mapping.

```
ACMEPACKET (local-response-map-entry) # local-error sdp-addressmismatch
ACMEPACKET (local-response-map-entry) # sip-status 403
ACMEPACKET (local-response-map-entry) # sip-reason sdp address mismatch
```

8. Save and activate your configuration.

## SIP SDP Address Correlation Configuration Enforcement Profile

To apply an enforcement profile to a realm:

## SIP Signaling Services

---

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. **enforcement-profile**—Enter the name of the enforcement profile you want to apply to this realm.

```
ACMEPACKET(realm-config)# enforcement-profile profile1
```

5. Save and activate your configuration.

## SDP Insertion for (Re)INVITES

---

If your network contains some SIP endpoints that do not send SDP in ReINVITES but also contains others that refuse INVITES without SDP, this feature can facilitate communication between the two types. The Oracle Communications Session Border Controller can insert SDP into outgoing INVITE messages when the corresponding, incoming INVITE does not contain SDP.

You can also use this feature when the network devices used in H.323-SIP interworking do not include SDP in the INVITES sent to SIP endpoints. In this case, the Oracle Communications Session Border Controller can insert SDP in the outgoing INVITE messages it forwards to the next hop.

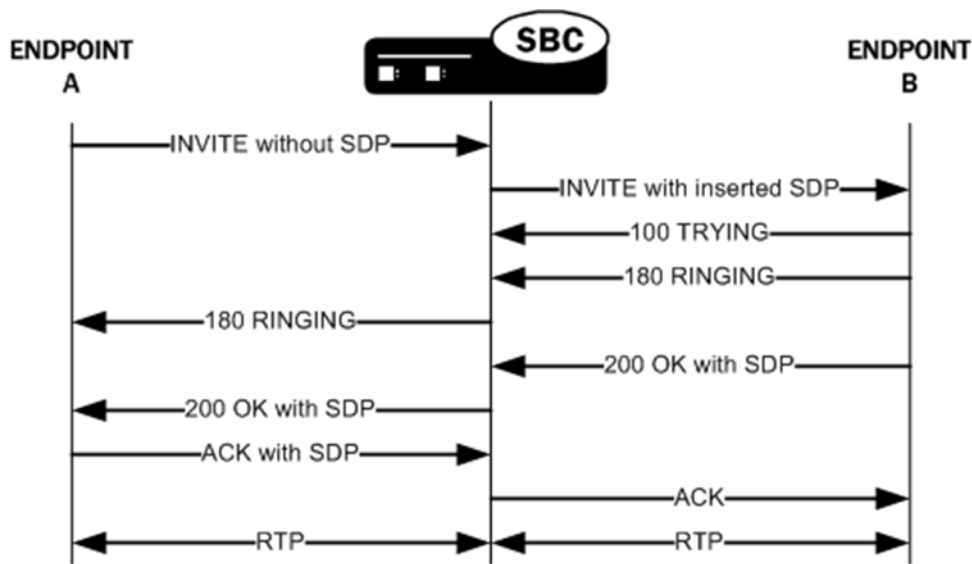
This feature works for both INVITES and ReINVITES.

This section explains how the SDP insertion feature works for INVITES and ReINVITES. The examples used in this section are both pure SIP calls. Even when you want to use this feature for IWF calls, though, you configure it for the SIP side.

## SDP Insertion for SIP INVITES

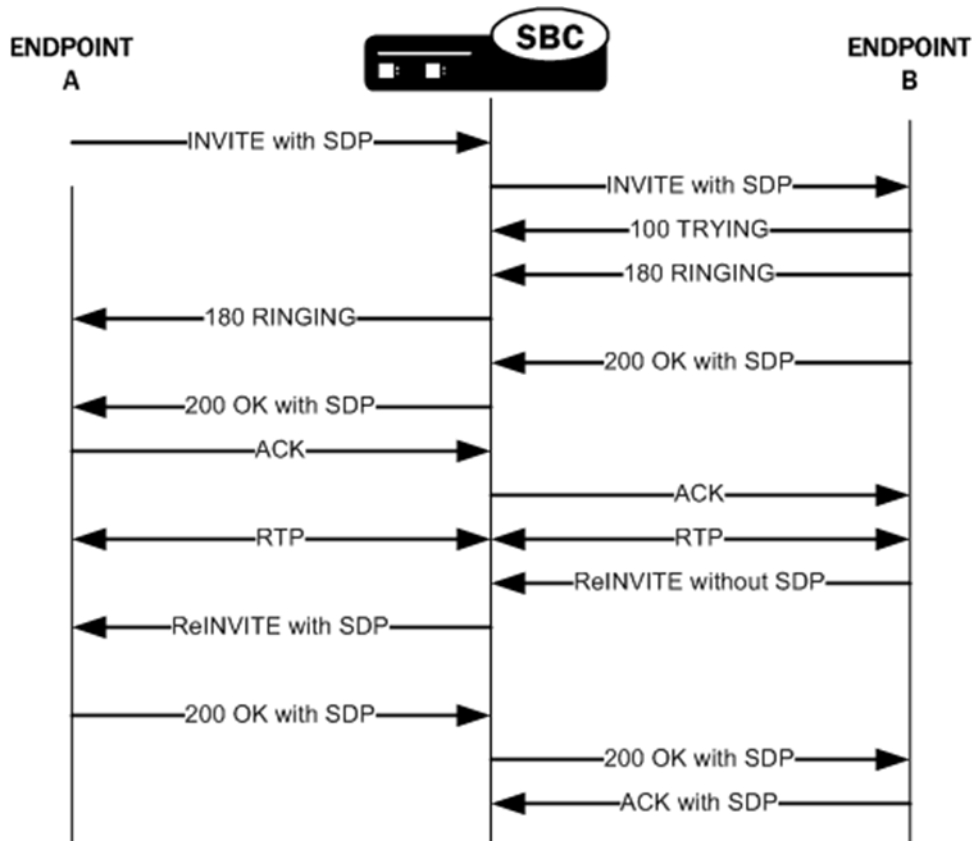
With the parameters mentioned above appropriately configured, the Oracle Communications Session Border Controller inserts SDP into an outgoing INVITE when the corresponding incoming INVITE has none. Because no SDP information is available for the session, the Oracle Communications Session Border Controller uses a media profile from a list of them you configure and then apply for SDP insertion.





### SDP Insertion for SIP ReINVITES

The section explains SDP insertion for ReINVITES, using a case where SIP session has been established with an initial INVITE containing SDP. In the diagram below, you can see the initial INVITE results in a negotiated media stream. But after the media stream is established, Endpoint B sends a ReINVITE without SDP to the Oracle Communications Session Border Controller. In this case, the Oracle Communications Session Border Controller inserts the negotiated media information from the initial INVITE as the ReINVITE's SDP offer. For subsequent ReINVITES with no SDP, the Oracle Communications Session Border Controller inserts the negotiated media information from the last successful negotiation as the ReINVITE's SDP offer. It then sends this ReINVITE with inserted SDP to the next hop signaling entity.



### SDP Insertion Configuration

This section shows you how to configure SDP insertion for the calls cases described above.

#### Configuring SDP Insertion for SIP INVITES

To work properly, SDP insertion for SIP invites requires you to set a valid media profile configuration.

To enable SDP insertion for INVITES:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal  
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router) # sip-interface  
ACMEPACKET (sip-config) #
```

4. **add-sdp-invite**—Change this parameter from disabled (default), and set it to **invite**.
5. **add-sdp-profile**—Enter a list of one or more media profile configurations you want to use when the system inserts SDP into incoming INVITES that have no SDP. The media profile contains media information the Oracle Communications Session Border Controller inserts in outgoing INVITE.

This parameter is empty by default.

6. Save and activate your configuration.

#### Configuring SDP Insertion for SIP ReINVITES

In this scenario, the Oracle Communications Session Border Controller uses the media information negotiated early in the session to insert after it receives an incoming ReINVITE without SDP. The Oracle Communications Session Border Controller then sends the ReINVITE with inserted SDP to the next hop signaling entity. You do not need the media profiles setting for ReINVITES.

To enable SDP insertion for ReINVITES:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal  
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router) # sip-interface  
ACMEPACKET (sip-config) #
```

4. **add-sdp-invite**—Change this parameter from disabled (default), and set it to **reinvite**.
5. Save and activate your configuration.

### SDP Version Change without SDP Body Change


---

When an SRTP call is made through the Oracle Communications Session Border Controller and a UE sends a reINVITE, there may be no change in the SDP contents. When the other UE responds, some devices will increment the `o=` line's session version, despite no SDP change. Normally the change in SDP version indicates that the SDP has changed, which would otherwise require the Oracle Communications Session Border Controller to modify the media

flow set-up. In order to leave the media flows unchanged when session version changes but the rest of the SDP does not, an option is configured in the media-manager-config.

If the Oracle Communications Session Border Controller attempts to modify these media flows when unnecessary, calls could be dropped by the system disrupting media packet sequence number synchronization.

The ignore-reinv-session-ver option is set to handle this situation. When configured, the Oracle Communications Session Border Controller compares the current SDP received from a UE against the previously-received SDP from the same UE. If the SDP in the newer and older messages is the same, the Oracle Communications Session Border Controller ignores any changes to the session-version and will not modify the media flow portion of the call.

 **Note:** When the SDP change is only a reordering of SDP lines without any other change, the option has no effect.

This option is used only to mitigate compatibility issues when running SRTP calls over any of Acme Packet's IPsec accelerated NIUs for the Net-Net 4500 or 3820. NOT for ETC NIUs.

## SDP Version Change Configuration

To configure media over TCP:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-level configuration elements.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **media-manager** and press Enter to begin configuring media over TCP.

```
ACMEPACKET(media-manager)# media-manager
ACMEPACKET(media-manager-config)#
```

4. **options** — Set the options parameter by typing **options**, a Space, the option name **ignore-reinv-session-ver** with a “plus” sign in front of it, and then press Enter.

```
ACMEPACKET(media-manager-config)# options +ignore-reinv-session-ver
```

If you type the option without the “plus” sign, you will overwrite any previously configured options. In order to append the new options to the realm configuration's options list, you must prepend the new option with a “plus” sign as shown in the previous example.

Save and activate your configuration.

## Restricted Media Latching

The restricted media latching feature lets the Oracle Communications Session Border Controller latch only to media from a known source IP address, in order to learn and latch the dynamic UDP port number. The restricting IP address's origin can be either the SDP information or the SIP message's Layer 3 (L3) IP address, depending on the configuration.

### About Latching

Latching is when the Oracle Communications Session Border Controller listens for the first RTP packet from any source address/port for the destination address/port of the Oracle Communications Session Border Controller. The destination address/port is allocated dynamically and sent in the SDP. After it receives a RTP packet for that allocated destination address/port, the Oracle Communications Session Border Controller only allows subsequent RTP packets from that same source address/port for that particular Oracle Communications Session Border Controller destination address/port. Latching does not imply that the latched source address/port is used for the destination of the reverse direction RTP packet flow (it does not imply the Oracle Communications Session Border Controller will perform symmetric RTP).

### Restricted Latching

The Oracle Communications Session Border Controller restricts latching of RTP/RTCP media for all calls within a realm. It latches to media based on one of the following:

- SDP: the IP address and address range based on the received SDP c= connect address line in the offer and answer.
- Layer 3: the IP address and address range based on the received L3 IP address of the offer or answer. This option is for access registered HNT endpoints. If the L3 IP address is locally known and cached by the Oracle Communications Session Border Controller as the public SIP contact address, that information could be used instead of waiting for a response. The Oracle Communications Session Border Controller might use the L3 IP address restriction method for all calls regardless of whether the endpoint is behind a NAT or not, for the same realms.

### Symmetric Latching

A mode where a device's source address/ports for the RTP/RTCP it sends to the Oracle Communications Session Border Controller (SBC) that are latched, are then used for the destination of RTP/RTCP sent to the device.

After allocating the media session in SIP, the SBC sets the restriction mode and the restriction mask for the calling side as well as for the called side. It sets the source address and address prefix bits in the flow. It also parses and loads the source flow address into the MIBOCO messages. After receiving the calling SDP, the SBC sets the source address (address and address prefix) in the appropriate flow (the flow going from calling side to the called side). After receiving the SDP from the called side, the SBC sets the source address in the flow going from the called side to the calling side.

The SBC uses either the address provided in the SDP or the layer 3 signaling address for latching. You also configure the SBC to enable latching so that when it receives the source flow address, it sets the address and prefix in the NAT flow. When the NAT entry is installed, all the values are set correctly. In addition, sipd sends the information for both the incoming and outgoing flows. After receiving SDP from the called side sipd, the SBC sends information for both flows to the MBCD so that the correct NAT entries are installed.

Enabling restricted latching may make the SBC wait for a SIP/SDP response before latching, if the answerer is in a restricted latching realm. This is necessary because the SBC does not usually know what to restrict latching to until the media endpoint is reached. The only exception could be when the endpoint's contact/IP is cached.

### Relationship to Symmetric Latching

The current forced HNT symmetric latching feature lets the Oracle Communications Session Border Controller assume devices are behind NATs, regardless of their signaled IP/SIP/SDP layer addresses. The Oracle Communications Session Border Controller latches on any received RTP destined for the specific IP address/port of the Oracle Communications Session Border Controller for the call, and uses the latched source address/port for the reverse flow destination information.

If both restricted latching and symmetric latching are enabled, the Oracle Communications Session Border Controller only latches if the source matches the restriction, and the reverse flow will only go to the address/port latched to, and thus the reverse flow will only go to an address of the same restriction.

- Symmetric latching is enabled.

If symmetric latching is enabled, the Oracle Communications Session Border Controller sends the media in the opposite direction to the same IP and port, after it latches to the source address of the media packet.

- Symmetric latching is disabled.

If symmetric latching is disabled, the Oracle Communications Session Border Controller only latches the incoming source. The destination of the media in the reverse direction is controlled by the SDP address.

### Example 1

A typical example is when the Oracle Communications Session Border Controller performs HNT and non-HNT registration access for endpoints. Possibly the SDP might not be correct, specifically if the device is behind a NAT. Therefore the Oracle Communications Session Border Controller needs to learn the address for which to restrict the media latching, based on the L3 IP address. If the endpoint is not behind a NAT, then the SDP could be used instead if

preferred. However, one can make some assumptions that access-type cases will require registration caching, and the cached fixed contact (the public FW address) could be used instead of waiting for any SDP response.

### Example 2

Another example is when a VoIP service is provided using symmetric-latching. A B2BUA/proxy sits between HNT endpoints and the Oracle Communications Session Border Controller, and calls do not appear to be behind NATs from the Oracle Communications Session Border Controller's perspective. The Oracle Communications Session Border Controller's primary role, other than securing softswitches and media gateways, is to provide symmetric latching so that HNT media will work from the endpoints.

To ensure the Oracle Communications Session Border Controller's latching mechanism is restricted to the media from the endpoints when the SIP Via and Contact headers are the B2BUA/proxy addresses and not the endpoints', the endpoint's real (public) IP address in the SDP of the offer/answer is used. The B2BUA/proxy corrects the c= line of SDP to that of the endpoints' public FW address.

The Oracle Communications Session Border Controller would then restrict the latching to the address in the SDP of the offer from the access realm (for inbound calls) or the SDP answer (for outbound calls).

## Restricted Latching Configuration

To configure restricted latching:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-level configuration elements.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

4. Select the realm where you want to apply this feature.

```
ACMEPACKET (realm-config) # select
identifier:
1: Acme_Realm <none>          0.0.0.0
2: H323REALM <none>         0.0.0.0
selection:1
ACMEPACKET (realm-config) #
```

5. **restricted-latching**— Enter the restricted latching mode. The default is **none**. The valid values are:
  - **none**—No restricted-latching used
  - **sdp**—Use the address provided in the SDP for latching
  - **peer-ip**—Use the layer 3 signaling address for latching
6. **restriction-mask**— Enter the number of address bits you want used for the source latched address. This field will be used only if the restricted-latching is used. The default is **32**. When this value is used, the complete IP address is matched for IPv4 addresses. The valid range is:
  - Minimum—1
  - Maximum—128

## Enhanced SIP Port Mapping

This section explains how to configure SIP port mapping feature to support:

- Anonymous requests from endpoints

- Cases where endpoints dynamically change transport protocols between UDP and TCP

### Anonymous Requests

If a SIP endpoint sends an INVITE message with a From header that is anonymous, the Oracle Communications Session Border Controller can find the registration cache entry by using the Contact and Via headers. In cases such as instant messaging (IM), where there is no Contact header, the Oracle Communications Session Border Controller can use the Via header.

The Oracle Communications Session Border Controller's checks whether the reg-via-key option is configured for the access-side SIP interface where a REGISTER is received. If the option is enabled, the Oracle Communications Session Border Controller makes the via-key by adding the IP address from the Via header to the firewall address (if there is a firewall present between the Oracle Communications Session Border Controller and the endpoint).

When an INVITE arrives at a SIP interface where this option is enabled, the Oracle Communications Session Border Controller determines whether the From header is anonymous or not. If it is anonymous, then the Oracle Communications Session Border Controller uses the Via-key to find the registration entry.

### Anonymous SIP Requests Configuration

To enable support for anonymous SIP requests:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-interface  
ACMEPACKET (sip-interface) #
```

4. Type **options +reg-via-key** and press Enter.

```
ACMEPACKET (sip-interface) # options +reg-via-key
```

If you type **options reg-via-key** without the plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.

5. Save and activate your configuration.

## SIP Registration Via Proxy

---

The Oracle Communications Session Border Controller supports a number of features that require it to cache registration information for UAs (endpoints) registering and receiving requests through it. For those features to operate correctly, the Oracle Communications Session Border Controller must act as the outbound proxy through which these endpoints register.

In order to support deployments where a proxy sits between the Oracle Communications Session Border Controller and the endpoints, the Oracle Communications Session Border Controller must consider the bottom Via header it receives from endpoints when constructing and matching cache registration entries. And when you use SIP port mapping, the system must use the bottom Via header as a way to determine the endpoint uniquely so that it can have a unique mapping port when the SIP interface is configured with the reg-via-key=all option.

Using the reg-via-proxy option, you can enable your Oracle Communications Session Border Controller to support endpoints that register using an intervening proxy. You can set this option either for a realm or for a SIP interface. If you use it for a SIP interface, add to the SIP interface pointing toward the proxy and endpoints—the access side.

## Considerations for Reg-Via-Key and Port Mapping

When you set the **reg-via-proxy** option, the Oracle Communications Session Border Controller includes the bottom Via header from received requests in the registration cache Via Key. The system also uses it for determining whether or not the request matches a registration cache entry. Each unique bottom Via received a unique mapping port when you turn SIP port mapping on and set the SIP interface with the **reg-via-key=all** option.

## Request Routing

So that requests addressed to the corresponding registered contact are routed to the proxy, the Oracle Communications Session Border Controller includes the intervening proxy (i.e., the top Via) in the routing information for the registration cache when you set **reg-via-proxy**. To carry out this routing scheme, the system adds a Path header (if none is present) to the REGISTER. But it removes the Path header prior to sending the REGISTER to the registrar.

Note that when the received REGISTER contains a Path header, the Oracle Communications Session Border Controller uses it for routing requests toward the endpoint and includes it in the forwarded REGISTER request—as is the case when you do not enable SIP registration via proxy.

## SIP Registration Via Proxy Configuration

To configure SIP registration via proxy:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. Type **options +reg-via-proxy** and press Enter.

```
ACMEPACKET(sip-interface)# options +reg-via-proxy
```

**If you type options reg-via-proxy** without the plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.

5. Save and activate your configuration.

## Dynamic Transport Protocol Change

The Oracle Communications Session Border Controller also uses the IP address and port in the Contact and Via headers. This is useful for cases when endpoints dynamically change transport protocols (TCP/UDP), and the port number used for sending an INVITE might not be the same one used to send a Register message.

If you do not enable this feature, when an endpoint registered with the Oracle Communications Session Border Controller used UDP for its transport protocol, a call fails if that endpoint subsequently initiates the call using TCP. The Oracle Communications Session Border Controller checks for the Layer 3 IP address and port, and it rejects the call if the port is changed.

With the new option **reg-no-port-match** added to the SIP interface configuration, the Oracle Communications Session Border Controller will not check the Layer 3 port in the INVITE and REGISTER messages.

## Dynamic Transport Protocol Change Configuration

To enable dynamic transport protocol change:

## SIP Signaling Services

---

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure)# session-router  
ACMEPACKET (session-router)#
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router)# sip-interface  
ACMEPACKET (sip-interface)#
```

4. Type **options +reg-no-port-match** and press Enter.

```
ACMEPACKET (sip-interface)# options +reg-no-port-match
```

**If you type options reg-no-port-match without the plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.**

5. Save and activate your configuration.

## SIP Privacy Extensions

---

This section explains how you can configure privacy services to be applied only when the source is trusted and the destination is considered untrusted. (Prior to this release, the Oracle Communications Session Border Controller always applied the privacy services, unless the source and the destination were both trusted.)

The Oracle Communications Session Border Controller considers all user endpoints and nodes outside the core as untrusted.

The Oracle Communications Session Border Controller acts as the boundary device between the trusted platform and the untrusted Internet, to implement privacy requirements. When it receives a message, the Oracle Communications Session Border Controller checks whether the source is trusted. It evaluates the level of privacy requested in a Privacy header, if present.

Depending on whether the source is trusted or untrusted, the Oracle Communications Session Border Controller can do different things when passing the message to the outgoing side. It also checks whether the destination is trusted.

### Privacy Types Supported

The Oracle Communications Session Border Controller supports the following Privacy types:

- **user**: user-level privacy function provided. Any non-essential informational headers are removed, including the Subject, Call-Info, Organization, User-Agent, Reply-To, and In-Reply-To. Possibly the original value of the From header is changed to anonymous.
- **header**: headers that cannot be set arbitrarily by the user (Contact/Via) are modified. No unnecessary headers that might reveal personal information about the originator of the request are added. (The values modified must be recoverable when further messages in the dialog need to be routed to the originator.)
- **id**: third-party asserted identity kept private with respect to SIP entities outside the trust domain with which the user authenticated.

The following SIP headers can directly or indirectly reveal identity information about the originator of a message: From, Contact, Reply-To, Via, Call-Info, User-Agent, Organization, Server, Subject, Call-ID, In-Reply-To and Warning.

#### **user**

The Oracle Communications Session Border Controller supports the Privacy type user. It can remove non-essential information headers that reveal user information by:



- Setting the SIP From header and display information to anonymous
- Removing the Privacy header
- Removing Proxy-Require option tag = privacy (if present)
- Removing the following headers:

Subject

Call-Info

Organization

User-Agent

Reply-To

In-Reply-To

### header

The Oracle Communications Session Border Controller also supports the Privacy type header. It modifies SIP headers that might reveal the user identity by:

- Stripping the Via header
- Replacing the Contact header
- Stripping Record-Route
- Removing the Privacy header
- Removing Proxy-Require option tag = privacy (if present)

In general, the B2BUA behavior of the Oracle Communications Session Border Controller by default provides header privacy for all sessions.

### id

The Oracle Communications Session Border Controller also supports the Privacy type id. It keeps the Network Asserted Identity private from SIP entities outside the trusted domain by:

- Stripping only P-Asserted-Identity
- Removing the Privacy header and Proxy-Require option-tag = privacy
- Setting the From header to anonymous (for the backward compatibility)

## Examples

The following examples show the actions the Oracle Communications Session Border Controller performs depending on the source and target of the calls.

### Calls from Untrusted Source to Trusted Target

When calls are from an untrusted source to a trusted target and PPI is included in the INVITE sent to IP border elements, the Oracle Communications Session Border Controller maps the PPI information to PAI in the outgoing INVITE to the trusted side (even if the Privacy header is set to id or to none). The Privacy and From headers get passed on unchanged.

IP border elements must pass PAI (if received in the ingress INVITE) and the From and Privacy headers to the egress side just as they were received on the ingress side.

The Oracle Communications Session Border Controller maps the PPI to PAI by default, if the outgoing side is trusted. To change this behavior, you need to configure the `disable-ppi-to-pai` option.

### Calls from Trusted to Untrusted

When calls are from a trusted source to an untrusted target, and the Privacy header is set to id, the Oracle Communications Session Border Controller strips PAI, makes the From header anonymous, and strips the Privacy header.

## SIP Signaling Services

---

If the Privacy header is set to none, the Oracle Communications Session Border Controller does not change the From header and passes on the Privacy header, if there is one.

### Calls from Trusted to Trusted

When calls are going from trusted source to trusted target acting as a peer network border element and PPI is included, the Oracle Communications Session Border Controller maps PPI to PAI. The Privacy header remains the same as signaled and the Oracle Communications Session Border Controller passes the From header and the PAI without changes.

## Configuring SIP Privacy Extensions

Prior to this release the session agent's trust mode provided this functionality. Now you configure SIP interface's trust-mode as none, which means nothing is trusted for this SIP interface.

You also configure the disable-ppi-to-pai parameter disable the changing of the P-Preferred header to the P-Asserted-Identity header, if the outgoing side is trusted.

### Trust Mode

To configure the trust mode:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-interface  
ACMEPACKET (sip-interface) #
```

From this point, you can configure SIP interface parameters. To view all sip-interface parameters, enter a ? at the system prompt.

4. If configuring an existing interface, enter the select command to select the interface.
5. **trust-mode**—Select the trust mode for this SIP interface. The default value is **all**. The valid values are:
  - **all**—Trust all previous and next hops except untrusted session agents
  - **agents-only**—Trust only trusted session agents
  - **realm-prefix**—Trusted only trusted session agents or address matching realm prefix
  - **registered**—Trust only trusted session agents or registered endpoints
  - **none**—Trust nothing
6. Save and activate your configuration.

The following example shows the trust-mode set to none. The remaining SIP interface options are omitted for brevity.

```
sip-interface  
state                enabled  
realm-id             access1  
sip-port  
    address          192.168.1.30  
    port              5060  
    transport-protocol UDP  
    allow-anonymous  all  
carriers  
proxy-mode           Proxy  
redirect-action  
contact-mode         maddr  
nat-traversal        none
```

```

nat-interval          30
registration-caching disabled
min-reg-expire        300
registration-interval 3600
route-to-registrar    disabled
teluri-scheme         disabled
uri-fqdn-domain
options
trust-mode            none

```

### Disabling the PPI to PAI Change

To disable the changing of PPI to PAI:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-config** and press Enter. The system prompt changes.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

From this point, you can configure SIP configuration parameters. To view all sip-config parameters, enter a ? at the system prompt.

4. If configuring an existing SIP configuration, enter the select command to select it.
5. **options**—Enter **disable-ppi-to-pai**. If adding to an existing list of options, use a preceding plus (+) sign.

```
options +disable-ppi-to-pai
```

6. Save and activate your configuration.

## SIP Registration Cache Limiting

Using SIP registration cache limiting for SIP endpoint access deployments, you can restrict the size of the SIP registration cache for the global SIP configuration.

You can implement this feature if you have been seeing issues where, either due to network failure scenarios or incorrect sizing of system capabilities, the Oracle Communications Session Border Controller and/or the SIP registrar cannot support the number of registering endpoints. Although the Oracle Communications Session Border Controller protects itself and the registrar against SIP REGISTER floods, conditions can still occur where too many legitimate endpoints attempt to register with the registrar via the Oracle Communications Session Border Controller.

By enabling SIP registration cache limiting, you restrict the number of legitimate endpoints that can register. The Oracle Communications Session Border Controller rejects any endpoints beyond the limit you set. If you do not want to use this feature, simply leave the reg-cache-limit parameter set to its default of 0, meaning there is no limit to the entries in the SIP registration cache.

When you limit the number of registered endpoints allowed in the Oracle Communications Session Border Controller's registration cache, the Oracle Communications Session Border Controller analyzes each registration before starting to process it. First, the Oracle Communications Session Border Controller checks the contact header to determine if it is already in the list of contacts for the user. If it finds the contact in its cache list, the Oracle Communications Session Border Controller treats the registration as a refresh; it treats any other headers as new. Note that the Oracle Communications Session Border Controller checks the message prior to making any changes to the cache because it must either accept or reject the message as a whole.

The Oracle Communications Session Border Controller adds the number of new contacts to the number already present in the cache, and rejects any registration with a contact that would cause it to exceed its limit. Rejection causes the Oracle Communications Session Border Controller to send a response communicating that its registration

cache is full. The default response is the 503 Registration DB-Full message, but you can use the SIP response mapping feature to use another message if required.

You can set an option in the global SIP configuration that defines the value in the Retry-After header. The Oracle Communications Session Border Controller sends this header as part of its rejection response when the registration cache is full. Another option sets the percentage of the registration cache size which, if exceeded, causes the Oracle Communications Session Border Controller to send an alarm.

### About Registration Cache Additions Modifications and Removals

When it receives a REGISTER message with new contact information for a user, the Oracle Communications Session Border Controller considers it an addition to the cache and augments the number of registration cache entries. Then the Oracle Communications Session Border Controller forwards the message to the registrar, and—when and only when the registrar returns both the original and new contacts in the 200 OK—the registration cache count stays the same. However, if the registrar returns only the new contact (making this a case of modification), then the Oracle Communications Session Border Controller removes the old contact information and subtracts accordingly from the number of registration cache entries.

Thus the Oracle Communications Session Border Controller does not know whether a REGISTER might result in an addition or a modification until it receives a response from the registrar. For this reason, the Oracle Communications Session Border Controller first assumes it is to make an addition, and then updates the registration cache and count when it has the necessary information from the registrar.

The registration cache count does not reflect removals during the rejection check because the Oracle Communications Session Border Controller ignores registration messages or expires headers with their expires values set to zero when it counts new entries. The fact that removals take place after additions and modifications means that messages which remove one contact while adding another might be rejected. That is, the addition might exceed the registration cache limit before any removal can take place to make room for it.

### Registration Cache Alarm Threshold

A percentage of the registration cache limit, the registration cache alarm threshold is a configurable value you can set to trigger an alarm when the registration cache is reaching its limit. When exceeded, this threshold triggers the generation of an alarm and SNMP trap. When registrations fall back beneath the threshold, the Oracle Communications Session Border Controller clears the alarm and sends a clear trap.

This alarm is Major in severity, and its text reads as follows:

```
Number of contacts <registration count> has exceeded the registration cache  
threshold <threshold %> of <registration cache limit value>.
```

### Notes on Surrogate Registration

The Oracle Communications Session Border Controller does not, under any circumstances, reject surrogate registrations on the basis of the registration cache limit. However, surrogate registrations generate contacts, and so they do add to the global registration count. In the case where the surrogate registrations add to the registration count to the extent the count exceeds the limit you configure, you will have more registrations in the cache than the configured limit.

### Monitoring Information

You can monitor how many entries are in the SIP registration cache using the ACLI `show registration` command and referring to the Local Contacts statistics.

### SIP Registration Cache Limiting Configuration

This section shows you how to configure the registration cache limit, and how to set the options controlling retry times and thresholds for alarm purposes.

To configure SIP registration cache limiting:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) #
```

If you are adding this feature to an existing configuration, you need to select the configuration (using the **ACLI select** command) before making your changes.

4. **registration-cache-limit**—Set the registration cache limit, or the maximum number of SIP registrations that you want to keep in the registration cache. The minimum and default value for this parameter is 0, and you can set it to a maximum value of 999999999. Leaving this parameter set to 0 means there is no limit on the registration cache (and therefore leaves this feature disabled).
5. **options**—Set the options parameter by typing options, a Space, the option name **reg-cache-lim-retry-after=X** (where X is the value added to the Retry-After header) with a plus sign in front of it. This option defaults to 1800, and you can enter values from 0 to 999999999.

You can configure the alarm threshold option the same way, substituting the option name **reg-cache-alarm-thresh=X** (where X is the percentage of registration cache limit that triggers an alarm). This option defaults to 95, and you can enter value from 0 to 100.

```
ACMEPACKET (sip-config) # options +reg-cache-lim-retry-after=2500
ACMEPACKET (sip-config) # options +reg-cache-alarm-thresh=90
```

If you type options and then the option value for either of these entries without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

6. Save and activate your configuration.

## SIP Registration Overload Protection

You can configure your Oracle Communications Session Border Controller for SIP Registration overload protection, which augments the Oracle Communications Session Border Controller's protection methods. Working with the Oracle Communications Session Border Controller's access control and registration caching functions, this new feature guards against benign avalanche restarts. The avalanche is caused by events where many endpoints lose power or connectivity at once, are restored to service, and then flood the Oracle Communications Session Border Controller as they attempt to register again.

Normally, the Oracle Communications Session Border Controller handles SIP registration by creating a temporary registration cache for the endpoint's address of record (AoR) and forwards the REGISTER request to the registrar. To challenge the endpoint's registration, the registrar sends back either a 401 Unauthorized or 407 Proxy Authorization Required response. When it receives the 401 or 407, the Oracle Communications Session Border Controller saves the challenge context in anticipation of receiving a second REGISTER with the endpoint's authentication credentials. The Oracle Communications Session Border Controller forwards the second REGISTER (with authentication credentials) to the registrar, and then the registrar confirms registration with a 200 OK. Both REGISTER requests are subject to the system's access control rules, set either for the ingress realm or the ingress session agent. The Oracle Communications Session Border Controller also honors the maximum registration sustain rate constraint for session agents; this applies when the incoming REGISTER is from a session agent and the outgoing REGISTER is sent to a session agent.

When you enable SIP Registration overload protection, the Oracle Communications Session Border Controller temporarily promotes the endpoint to the trusted level when it receives the 401 or 407 response (to the first REGISTER) from the registrar. This ensures that the second REGISTER (containing authentication credentials) can

reach the Oracle Communications Session Border Controller. Temporary promotion lasts only for the amount of time remaining before the REGISTER server transaction expires plus the time allotted in the transaction expiration parameter in the SIP configuration. Before the temporary promotion expires, there is enough time for any necessary retransmissions of the first REGISTER and for the second REGISTER to take place. The following situations might also occur:

- If the Oracle Communications Session Border Controller receives a 401 or 407 to the second REGISTER request, it resets its access control level for the endpoint's address to the default level; it then treats additional REGISTER requests from the same context at the default access control level.
- If the Oracle Communications Session Border Controller receives a 200 OK response to the REGISTER message, it extends the promotion time to the expiration period for the registration cache.

If the Oracle Communications Session Border Controller is able to find the temporary registration cache and the saved challenge context when the second REGISTER arrives, it forwards the REGISTER without checking the maximum registration sustain rate constraint for ingress and egress session agents—thereby ensuring that the REGISTER with authentication credentials is sent to the registrar. So when you use this feature, you should set the maximum registration sustain rate constraint of the session agent (representing the registrar) at half the registrar's maximum registration sustain rate. Additional REGISTER requests with the same challenge context are subject to the maximum registration sustain rate constraint.

## SIP Registration Overload Protection Configuration

When you configure this feature, be sure to set the `reg-overload-protect` option in your global SIP configuration:

To enable SIP Registration overload protection on your Oracle Communications Session Border Controller:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type `sip-config` and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

4. `options`—Set the options parameter by typing `options`, a Space, the option name preceded by a plus sign (+) (`reg-overload-protect`), and then press Enter.

```
ACMEPACKET(sip-config)# options +reg-overload-protect
```

**If you type either of these options without the plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.**



**Note:** Note that the sip-config option "cache-challenges" (enabled by default) must not have been disabled for SIP Registration Overload Protection to work properly. If you have disabled cache-challenges, re-evaluate the reason you disabled it. If registration overload protection supersedes your reason for disabling cache-challenges, re-enable the option as shown below.

```
ACMEPACKET(sip-config)# options +cache-challenges=yes
```

Note that the configuration syntax above is equivalent to the following, which uses the "-" character to remove the option.

```
ACMEPACKET(sip-config)# options -cache-challenges
```

5. Save and activate your configuration.

## SIP Instance ID in Registration Cache

As defined in RFC 5626, Managing Client-Initiated Connections in the Session Initiation Protocol (SIP), the +sip.instance uniquely identifies a specific user agent instance. The instance-id does not change even when the User Agent is rebooted or power cycled. Using the instance-id from the +sip-instance parameter allows the SBC to locate a registered contact even when the IP address of the UA has changed.

The +sip.instance is a Contact header field parameter that contains a globally unique identifier, generally a Universally Unique Identifier (UUID) Uniform Resource Name (URN) that identifies a specific user agent client.

As defined in RFC 5626, Managing Client-Initiated Connections in the Session Initiation Protocol (SIP), the parameter contains an "instance-id" that uniquely identifies a specific user Agent instance. The instance-id does not change even when the User Agent is rebooted or power cycled (see Section 3.1 of RFC 5626). Using the instance-id from the +sip-instance parameter allows the SBC to locate a registered contact even when the IP address of the UA has changed.

An example REGISTER message containing a +sip-instance parameter and assigned value is shown below.

```
REGISTER sip:example.com SIP/2.0
Via: SIP/2.0/TCP 192.0.2.2;branch=z9hG4bK-bad0ce-11-1036
Max-Forwards: 70
From: Bob <sip:bob@example.com>;tag=d879h76
To: Bob <sip:bob@example.com>
Call-ID: 8921348ju72je840.204
CSeq: 1 REGISTER
Supported: path, outbound
Contact: <sip:line1@192.0.2.2;transport=tcp>; reg-id=1;
+sip.instance="<urn:uuid:00000000-0000-1000-8000-000A95A0E128>"
Content-Length: 0
```

The user agent calculates a +sip.instance generally using a time-stamp, a unique string -- such as a MAC address, and/or a randomly generated number. After calculating the value, the user agent saves it to persistent storage, thus ensuring that the value is unchanged by subsequent power cycles.

## SIP Instance ID and the Registration Cache

Every assignment of a new IP address to a mobile user agent client results in a new REGISTER request. The receipt of the REGISTER request, in turn, generates an additional entry in the registration cache. The implementation can be simplified by including the +sip.instance value in the criteria used to match incoming REGISTER requests with existing sessions maintained in the registration cache. When a match is found, the SBC re-uses the existing registration cache rather than adding a new one.

If such a +sip.instance match is found and the contact address has changed, the SBC determines if digest authentication has been used in previous associated REGISTER requests. If so, the SBCs forwards the REGISTER request to the registrar for authentication of the new location.

## SIP Instance ID Configuration

Because this configuration is dynamically performed at the **sip-config** level, it applies to all SIP interfaces and takes effect immediately

1. Access the **sip-config** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

2. Select the **sip-interface** object to edit.

```
ACMEPACKET(sip-interface)# select
<RealmID>:
1: realm01 172.172.30.31:5060
```

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---

```
selection: 1
ACMEPACKET(sip-interface) #
```

3. **match-sip-instance**—Set this parameter to enabled to use the +sip-instance-id when matching incoming calls with the registration cache. Valid values are:

- **disabled**—(the default) disables the use of the +sip-instance-id when matching incoming calls with the registration cache
- **enabled**—enables the use of the +sip-instance-id when matching incoming calls with the registration cache

```
ACMEPACKET(sip-config) # match-sip-instance enabled
ACMEPACKET(sip-config) #
```

4. Type **done** to save your configuration.

## SIP Request Method Throttling

---

You can configure throttling mechanisms for SIP INVITEs and REGISTERs using session agent constraints. However, you might want to throttle other types of SIP methods, and for those methods you should use the rate constraints configuration available both in the session constraints (which you then apply to a SIP interface or a realm) and the session agent configurations.

Oracle recommends you use session agent constraints for session-rate INVITE throttling and registration-rate for REGISTER throttling.

For SIP access deployments, you can configure rate constraints for individual method types along with a set of burst and sustain rates. These constraints can help to avoid overloading the core network. In addition, they restrain the load non-INVITE messages use, thus reserving capacity for INVITE-based sessions and Registrations

When you configure SIP request method throttling, you must exercise care because it is possible to reject in-dialog requests. Therefore, Oracle recommends you do NOT configure constraints—although the configuration allows you to and will not produce error messages or warnings if you set them—for the following SIP method types:

- ACK
- PRACK
- BYE
- INFO
- REFER

However, the Oracle Communications Session Border Controller is likely to throttle NOTIFY requests despite their being part of a Subscribe dialog.

Therefore, the methods you will most likely configure for throttling are:

- NOTIFY
- OPTIONS
- MESSAGE
- PUBLISH
- REGISTER

The Oracle Communications Session Border Controller counts Re-INVITEs and challenged responses against the throttle limit, but does not check to determine if the constraints have been exceeded for either.

You can configure separate constraints—inbound and outbound values for burst and sustain rates—for each different method type you configure. Although you should use session agent constraints (and not rate constraints) for INVITEs, if you also set up rate constraints for INVITEs, then the smallest configured value takes precedence.

## About Counters and Statistics

Each rate constraint you configure for a SIP method tracks its own counters. For example, if you configure a rate constraint for the PUBLISH method, the burst and sustain rates you set for it apply only to the PUBLISH method and



not to any other methods for which you might set up rate constraints. You can, however, set the burst rate window in the session constraints configuration that will apply to all methods configured as rate constraints.

The Oracle Communications Session Border Controller captures statistics for SIP methods throttled by rate constraints for SIP interfaces and session agents; it does not capture these statistics for the global SIP configuration.

## SIP Request Method Throttling Configuration

This section shows you how to set up rate constraints for session constraints (which are then applied to SIP interfaces) and session agents.

To use this feature, you must enable the **extra-method-stats** parameter in the global SIP configuration.

To set the **extra-method-stats** parameter in the global SIP configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) #
```

If you are adding this feature to an existing configuration, you need to select the configuration (using the ACLI **select** command) before making your changes.

4. **extra-method-stats**—Set this parameter to **enabled**.

5. Save and activate your configuration.

### Rate Constraints for SIP Interfaces

To apply rate constraints to SIP interfaces, you need to configure rate constraints in the session constraints configuration and then apply the session constraints to the SIP interface where you want them used.

Note that you need to set up the parent **session-constraint** configuration to save any rate constraints you configure.

To configure rate constraints:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **session-constraints** and press Enter.

```
ACMEPACKET (session-router) # session-constraints
ACMEPACKET (session-constraints) #
```

If you are adding rate constraints to an existing configuration, then you will need to select the configuration you want to edit.

4. Type **rate-constraints** and press Enter.

```
ACMEPACKET (session-constraints) # rate-constraints
ACMEPACKET (rate-constraints) #
```

5. **method**—Enter the SIP method name for the method you want to throttle. Although the parameter accepts other values, your entries should come only from the following list for the feature to function properly:

- NOTIFY
  - OPTIONS
  - MESSAGE
  - PUBLISH
  - REGISTER
6. **max-inbound-burst-rate**—For the SIP method you set in the methods parameter, enter the number to restrict the inbound burst rate on the SIP interface where you apply these constraints. The default and minimum value is 0, and the maximum is 999999999.
  7. **max-outbound-burst-rate**—For the SIP method you set in the methods parameter, enter the number to restrict the outbound burst rate on the SIP interface where you apply these constraints. The default and minimum value is 0, and the maximum is 999999999.
  8. **max-inbound-sustain-rate**—For the SIP method you set in the methods parameter, enter the number to restrict the inbound sustain rate on the SIP interface where you apply these constraints. The default and minimum value is 0, and the maximum is 999999999.
  9. **max-outbound-sustain-rate**—For the SIP method you set in the methods parameter, enter the number to restrict the outbound sustain rate on the SIP interface where you apply these constraints. The default and minimum value is 0, and the maximum is 999999999.
  10. Save your changes and apply this session constraint and its rate constraint(s) to SIP interfaces.

### Applying Session and Rate Constraints to a SIP Interface

You need the name of the session constraints configuration to apply the restrictions you set up to a SIP interface.

To apply session and rate constraints to a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET(session-router) # sip-interface
ACMEPACKET(sip-interface) #
```

If you are adding this feature to an existing configuration, then you will need to select the configuration you want to edit.

4. **constraint-name**—Enter the name of the session constraint configuration where you have set up rate constraints to apply them to this SIP interface. This parameter has no default, and must be the valid name of a session constraint configuration.
5. Save and activate your configuration.

### Configuring Rate Constraints for Session Agents

You can also use this feature for individual SIP session agents.

To configure rate constraints for a SIP session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET(session-router) # session-agent
ACMEPACKET(session-agent) #
```

If you are adding rate constraints to an existing configuration, then you will need to select the configuration you want to edit.

4. Type **rate-constraints** and press Enter.

```
ACMEPACKET(session-agent) # rate-constraints
ACMEPACKET(rate-constraints) #
```

5. **method**—Enter the SIP method name for the method you want to throttle. Your entries should come only from the following list:
  - NOTIFY
  - OPTIONS
  - MESSAGE
  - PUBLISH
  - REGISTER
6. **max-inbound-burst-rate**—For the SIP method you set in the methods parameter, enter the number to restrict the inbound burst rate on the SIP interface where you apply these constraints. The default and minimum value is 0, and the maximum is 999999999.
7. **max-outbound-burst-rate**—For the SIP method you set in the methods parameter, enter the number to restrict the outbound burst rate on the SIP interface where you apply these constraints. The default and minimum value is 0, and the maximum is 999999999.
8. **max-inbound-sustain-rate**—For the SIP method you set in the methods parameter, enter the number to restrict the inbound sustain rate on the SIP interface where you apply these constraints. The default and minimum value is 0, and the maximum is 999999999.
9. **max-outbound-sustain-rate**—For the SIP method you set in the methods parameter, enter the number to restrict the outbound sustain rate on the SIP interface where you apply these constraints. The default and minimum value is 0, and the maximum is 999999999.
10. Save and activate your configuration.

## SIP Delayed Media Update

The Oracle Communications Session Border Controller supports SIP delayed media update. When enabled, this feature keeps the Oracle Communications Session Border Controller from updating its media flow information for flows established after an offer-answer exchange. The Oracle Communications Session Border Controller does not update the flow information until a new offer and answer arrive for a specific set of media flows.

The (subsequent) offer does not have to be for the same session; rather, it can appear as a new SIP INVITE that uses the same SDP.

### Delayed Media Update Disabled

When this feature is disabled (which is the default behavior), the Oracle Communications Session Border Controller updates media flow entries in its CAM based on signaled SDP when it processes the SDP. If it processes an SDP offer, Oracle Communications Session Border Controller allocates steering port resources; the Oracle Communications Session Border Controller updates any missing elements for the flow when the answer is returned.

In cases when a secondary offer arrives (either a reINVITE, an UPDATE, or the original INVITE is hairpinned back through the Oracle Communications Session Border Controller), the Oracle Communications Session Border Controller updates the following media flow information at the time of the offer

- Destination IP address
- Destination port
- Realm for the media flows
- Media release settings

## SIP Signaling Services

---

This behavior affects specific applications that are better served by the Oracle Communications Session Border Controller waiting to update media flow information until it receives the answer to the second offer.

### Delayed Media Update Enabled

When you enable the SIP delayed media update feature, the Oracle Communications Session Border Controller:

- Delays changing the active media flow CAM entry for a new offer if a previous offer and answer have been received for the same media flows; it encodes new SDP information in an outgoing offer, but does not change the CAM entry until the answer is received
- Delays changing the active media flow CAM entry even when the new offer is for a new session
- Supports media release when performing delayed media update changes
- Offers per-realm configuration

This section describes how the delayed media update feature works for hairpinned call flows and for an SDP offer arriving for installed flows.

- Hairpinned call flows—In this type of call flow, the application server (AS) sends an INVITE back to the Oracle Communications Session Border Controller and that INVITE needs to be forwarded to another user (user B). When it receives the offer in this INVITE and delayed media update is disabled, the Oracle Communications Session Border Controller determines that the call is hairpinned and deletes the CAM entry for the flow for user A, who has sent the initial INVITE. The Oracle Communications Session Border Controller deletes the CAM entry for the flow from the AS to user A.

With delayed media update enabled, the CAM entry for the flow from the AS to user A is not deleted. Instead, the Oracle Communications Session Border Controller waits until it has an answer from user B, and then performs the necessary updates and deletions.

- SDP offer for installed media flows—With delayed media update enabled, if it has received an offer and answer and a new offer arrives for the same flow, the Oracle Communications Session Border Controller delays updating the CAM entries until an answer is received for the new offer.

### SIP Delayed Media Update Configuration

You enable this feature on a per-realm basis by setting one parameter.

To enable SIP delayed media update:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the signaling-related configurations.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm-config** and press Enter.

```
ACMEPACKET (media-manager) # realm-config
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI **select** command) the realm that you want to edit.

4. **delay-media-update**—Enable keeping the Oracle Communications Session Border Controller from updating its media flow information for flows established after an offer/answer exchange. The default is **disabled**. The valid values are:
  - enabled | disabled
5. Save and activate your configuration.

### Expedited Call Leg Release for Preempted Hairpin Calls

---

When hairpinned calls are ended because of signaling failures (such as a SIP mid-dialog signaling timeout, or an H.323 TCP keepalive failure) on one call leg, the Oracle Communications Session Border Controller deletes both legs'

media flows simultaneously by default. In addition, when the first hairpinned call leg is torn down, the second call leg is gracefully released immediately by the Oracle Communications Session Border Controller creating and sending an appropriate signaling message (e.g., BYE for a SIP call or ReleaseComplete for an H.323 call) to the endpoint.

You can override this behavior by configuring the **dont-terminate-assoc-legs** option in the media manager. When configured, the orphaned call leg in the hairpin scenario will be torn down after the **initial guard timer expires**. The disconnect times of the two call legs, as recorded by the accounting application, will be significantly different, due to the initial guard time for the second call leg.

## Accounting Considerations

To indicate cases like this, where the second leg of the hairpinned call was preempted, Oracle Communications Session Border Controller includes the following combination of release/termination causes in the CDR:

```
VSA 49: Acct-Terminate-Cause = NAS_REQUEST
VSA 62: Acme-Disconnect-Cause = 8
```

## SIPconnect

The Oracle Communications Session Border Controller supports the SIPconnect model, wherein PBXs register themselves so that service providers do not need to know IP addresses or locations in advance for static configurations. This is particularly helpful when the PBX is behind a NAT.

In the PBX registration process, the PBX creates a binding between one of its phone numbers as the address of record (AoR) and Contact-URI in the REGISTER message. The registrar knows that the single AoR actually represents many addresses, and so it registers them implicitly. However, the registrar does not return the implicit AoR number in P-Associated-URIs.

The SIPconnect feature resolves the following issues that arise from using this model:

- SIP INVITEs sent to the PBX from the Registrar through the Oracle Communications Session Border Controller have the Request-URI of registered contact. Because it typically ignores the To-URI, the PBX needs the Request-URI username portion to be the specific extension number being called.

With the SIP connect feature enabled, the Oracle Communications Session Border Controller overwrites the Request-URI username with the To-URI username.

- SIP INVITEs from the PBX have the From AoR and Contact-URI usernames of specific phones rather than of the registered AoR and Contact-URI. For the Oracle Communications Session Border Controller, this means that it cannot use the **allow-anonymous** parameter value of register; there would be no registered user matches, and the Oracle Communications Session Border Controller would reject them (with a 403 Forbidden).

With the SIP connect feature enabled, the Oracle Communications Session Border Controller performs allow-anonymous checking based on the registered Via address, which is the same for all requests for the same PBX.

## Modifications to Registration Caching Behavior

With the SIP connect feature enabled, Oracle Communications Session Border Controller registration caching works the same way that it does with the feature disabled, with the following exceptions:

The Oracle Communications Session Border Controller determines whether the destination realm has the sip-connect-pbx-reg option configured, and then:

- If it is configured, the Oracle Communications Session Border Controller replaces the user part of the Request-URI with the user part of the To header. When the INVITE contains a P-Called-Party-ID header, the Oracle Communications Session Border Controller uses the user part of the P-Called-Party-ID header (instead of the To header).
- If it is not configured, the Oracle Communications Session Border Controller determines if the destination address is for a session agent and whether that session agent has sip-connect-pbx-reg option configured. When it is configured, the Oracle Communications Session Border Controller performs the same replacements described in

the bullet directly above. When it is not configured, the Oracle Communications Session Border Controller does not make any replacements.

When it receives an INVITE request, the Oracle Communications Session Border Controller checks the incoming realm for the sip-connect-pbx-reg option.

- If it is configured, the Oracle Communications Session Border Controller uses the INVITE's source address (instead of the AoR and Contact-URI) to search the registration cache for a matched registration entry.
- If it is not configured, the Oracle Communications Session Border Controller determines if the INVITE's source address is for a session agent and whether that session agent has sip-connect-pbx-reg option configured.

When it is configured, the Oracle Communications Session Border Controller replaces the user part of the Request-URI with the user part of the To header. When the INVITE contains a P-Called-Party-ID header, the Oracle Communications Session Border Controller uses the user part of the P-Called-Party-ID header (instead of the To header).

When it is not configured, the Oracle Communications Session Border Controller does not make any replacements.

## Configuring SIP Connect Support

You configure this feature by adding the sip-connect-pbx-reg option to the realm configuration. In addition, though this feature requires that your configuration also be set up as outlined in this section. The first two items are required, and Acme Packet recommends that you also implement the suggested additional configuration.

### Required Configuration

- Registration caching is enabled.
- For the realm from which registrations come, the options list must include sip-connect-pbx-reg; this is new configuration introduced to support this feature. The presence of this option instructs the Oracle Communications Session Border Controller to skip matching the Contact header in the INVITE request with the registered Contact of the registration entry. The Oracle Communications Session Border Controller finds a registration using only the INVITE's source address.

Alternatively, you can configure the sip-connect-pbx-reg option in the options list for a session agent. When the realm where an INVITE comes from does not have this option set, the Oracle Communications Session Border Controller determines whether or not the INVITE came from a session agent. You might choose to configure session agents with this option if you do not want it applied to an entire realm. If the PBX is behind a NAT device, the session agent's IP address for the PBX (if statically configured) must be the IP address of the NAT device. And if DNS is used, the session agent's hostname must resolve to the NAT device's IP address.

### Suggested Additional Configuration

- In the SIP ports configuration (accessed through the SIP interface configuration), the **allow-anonymous** parameter must be set to registered. This setting allows the Oracle Communications Session Border Controller to accept SIP requests from session agents and registered endpoints only, but to accept REGISTER requests from any endpoint.
- For the SIP interface that accepts registrations, the **options** parameter must be set to reg-via-key. This setting allows the Oracle Communications Session Border Controller to use the source address of an INVITE as the key to find a registration entry in the registration cache. When the INVITE's Contact header matches the registered Contact in the registration entry, the Oracle Communications Session Border Controller accepts the INVITE request.

## SIP Connect Configuration

To set the SIP connect option for a realm configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the signaling-related configurations.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm-config** and press Enter.

```
ACMEPACKET (media-manager) # realm-config
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI **select** command) the realm that you want to edit.

4. **options**—Set the options parameter by typing **options**, a Space, the option name **sip-connect-pbx-reg** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (realm-config) # options +sip-connect-pbx-reg
```

**If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the realm configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.**

5. Save and activate your configuration.

To set the SIP connect option for a SIP session agent configuration:

6. In Superuser mode, type **configure terminal** and press Enter.

```
Oracle Communications Session Border Controller# configure terminal
```

7. Type **session-router** and press Enter to access the signaling-related configurations.

```
Oracle Communications Session Border Controller(configure) # session-router
```

8. Type **session-agent** and press Enter.

```
ACMEPACKET (session-router) # session-agent
```

If you are adding support for this feature to a pre-existing session agent, then you must select (using the ACLI **select** command) the session agent that you want to edit.

9. **options**—Set the options parameter by typing **options**, a Space, the option name **sip-connect-pbx-reg** with a plus sign in front of it, and then press Enter.

```
Oracle Communications Session Border Controller(session-agent) # options +sip-connect-pbx-reg
```

**If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the session agent's configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.**

10. Save and activate your configuration.

## SIP Registration Event Package Support

Certain endpoints subscribe to the Registration Event Package, RFC 3680, which defines how SIP user agents can request and obtain notifications about registration events. Previously, the Oracle Communications Session Border Controller passed the Subscribe and Notify messages of this package transparently, without modifying the XML bodies of either. However, in many cases the XML body can contain IP addresses, contact URIs, and expires times that the Oracle Communications Session Border Controller needs to modify for proper operation. This new feature enables the Oracle Communications Session Border Controller to modify correctly the XML body for the Registration Event Package.

In addition to resolving this type of issue, enabling registration event package support on your system provides the functions described below:

- The Oracle Communications Session Border Controller performs NAT on all contacts in the reginfo, regardless of their state.
- The Oracle Communications Session Border Controller performs NAT on the address of record (AoR) attribute of the Registration element when it matches an existing cache entry. When either the Contact-URI or the AoR does

not match a cache entry and the host part of the URI is an IP address, the Oracle Communications Session Border Controller will NAT the host part using the applicable SIP NAT configuration

- Contacts are found in the XML URI element for the contact. But if there is no URI element, then the Oracle Communications Session Border Controller uses the Contact element information for the contact.
- If the expires attribute in the Contact element is a value other than zero, the Oracle Communications Session Border Controller uses (inserts) the expires values from the registration cache.
- This feature also introduces delayed deletion from the registry cache. When a 200 OK comes back in response to a REGISTER message and the 200 OK does not include all previously registered contacts, the missing contacts are deleted. If the global SIP configuration option `contact_cache_linger=XX` (where XX is the number of seconds to wait before deleting), then the contacts to be deleted remain for the specified number of seconds before they in fact are deleted.

## Updating Expiration Values

This feature also supports updating the expiration values for the registration cache when a Contact element has the expires attribute. For this support, the following apply:

- If the value of the expires attribute is greater than the expiration value for the access-side registration cache entry, the Oracle Communications Session Border Controller replaces the XML expires attribute value with the cached one from the access side.
- If the value of the XML expires attribute is less than the core-side expiration value for the core-side registration cache entry, the Oracle Communications Session Border Controller updates the core-side expiration value with the value from the expires attribute. Further, the Oracle Communications Session Border Controller adjusts the access-side expiration value of the registration cache in these ways:
  - If the value of the XML expires attribute is less than the current access-side expiration value for the registration cache entry, the Oracle Communications Session Border Controller sets the access-side expiration value to be equal to the value in the expires attribute.
  - Otherwise, the Oracle Communications Session Border Controller leaves the expires value for the access-side expiration value for the registration cache entry unchanged. If this happens, the Oracle Communications Session Border Controller replaces the value of the XML expires attribute with the adjusted access-side expiration value.
- If the expires attribute from a Contact element is 0 (meaning that the core is removing the registration), the Oracle Communications Session Border Controller removes that Contact-URI from its registration cache. And if the registration cache entry has no remaining Contact-URIs, the Oracle Communications Session Border Controller deletes the registration cache entry altogether.

## Contact Cache Linger Configuration

You enable this feature as part of the global SIP configuration, using that configuration's **options** parameter. You can optionally configure the number of seconds you want to keep a contact in the registration cache before it is deleted. This is the option:

- **contact-cache-linger=XX**—Number of seconds to wait before a contact is deleted from the cache (where XX is the number of seconds)

To enable SIP Registration overload protection on your Oracle Communications Session Border Controller:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router  
ACMEPACKET(session-router)#
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-config  
ACMEPACKET(sip-config)#
```



4. **options**—Set the options parameter by typing **options**, a Space, the option name preceded by a plus sign (+) (**contact-cache-linger=XX**) where XX is the number of seconds to keep a contact in the cache before deleting it.

```
ACMEPACKET(sip-config)# options +contact-cache-linger=5
```

If you type either of these options without the plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.

5. Save and activate your configuration.

## SIP Event Package for Registrations

This feature enables the Oracle Communications Session Border Controller, acting as a Proxy Call Session Control Function (P-CSCF) to initiate subscription to the SIP Event Package for Registrations. Support for the SIP Event Package for Registrations requires no special license.

### Applicable Standards

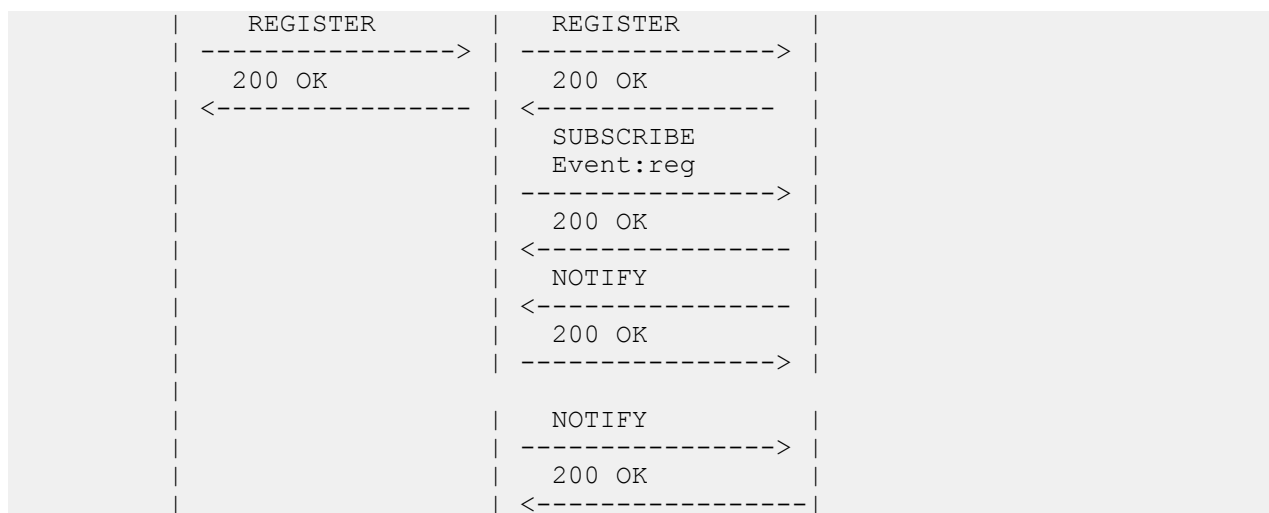
RFC 3265, *Session Initiation Protocol (SIP)-Specific Event Notification*, outlines a framework within which a SIP node, designated as the subscriber, can request automatic, asynchronous notification of certain events from a remote peer, designated as the notifier. RFC 3265 also defines an Event Package as a set of state information to be reported by a notifier to a subscriber, and mandates that such Event Packages be described in a specific RFC explicitly identifying the state information to be reported by the notifier and defining required syntax and semantics to support the subscription/notification exchange.

RFC 3680, *A Session Initiation Protocol (SIP) Event Package for Registrations*, fulfills this requirement by defining a method that enables SIP user agents to request a defined set of state information from a SIP Registrar.

Section 5.2.3 of the 3GPP (Third Generation Partnership Project) 24.229, *IP Multimedia Call Control Protocol Based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP)*; Stage 3, mandates that a P-CSCF subscribe to the SIP Event Package for Registrations as defined in RFC 3680.

### Call Flow

An example call flow between a SIP endpoint, the Oracle Communications Session Border Controller, acting as a P-CSCF, and a SIP Registrar (S-CSCF) illustrates the Subscription/Notification process.



The first two messages (the REGISTER request and the 200 REGISTER response) accomplish the successful registration of the SIP endpoint.

```
REGISTER sip:example.com SIP/2.0
Via: SIP/2.0/UDP pc34.example.com;branch=z9hG4bKnaaff
From: sip:joe@example.com;tag=99a8s
To: sip:joe@example.com
Call-ID: 88askjda9@pc34.example.com
CSeq: 9976 REGISTER
Contact: sip:joe@pc34.example.com
```

Immediately after processing the initial 200 OK from the SIP Registrar, the Oracle Communications Session Border Controller sends a SUBSCRIBE Request to the S-CSCF.

```
SUBSCRIBE sip:joe@example.com SIP/2.0
Via: SIP/2.0/UDP app.example.com;branch=z9hG4bKnashds7
From: sip:sd.example.com;tag=123aa9
To: sip:joe@example.com
Call-ID: 9987@app.example.com
CSeq: 9887 SUBSCRIBE
Contact: sip:joe@pc34.example.com
P-Asserted-Identity: <sip:sd@example.com>
Event: reg
Max-Forwards: 70
Accept: application/reginfo+xml
```

The Request URI and To header contain the address-of-record (sip:joe@example.com) of the subscription subject. This value was previously contained in the From and To headers of the original REGISTER request. These fields are always identical in REGISTER requests, except in the case of third-party registration.

The From and P-Asserted-Identity headers contain the identity of the subscription requester.

The Contact header contains an IP address or FQDN at which the subscription subject can be reached. This field duplicates the value of the Contact header in the original REGISTER request. Multiple contacts can be registered for a single address-of-record.

The Event header contains the required value, reg, which identifies the requested Event Package subscription. reg specifies the SIP Event Package for Registrations.

The Accept header contains the required, default value, application/reginfo+xml, indicating syntactical support for Registration notifications and attached XML notification bodies.

Assuming the S-CSCF accepts the subscription, it responds with a 200 SUBSCRIBE response.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP app.example.com;branch=z9hG4bKnashds7
;received=192.0.2.1
From: sip:sd.example.com;tag=123aa9
To: sip:joe@example.com;tag=xyzygg
Call-ID: 9987@app.example.com
CSeq: 9987 SUBSCRIBE
Contact: sip:joe@pc34.example.com
Expires: 3600
```

The From header contains the identity of the subscription requester.

The To header contains the address-of-record of the subscription subject.

The Contact header contains an IP address or FQDN at which the subscription subject can be reached.

The Expires header contains the subscription duration in seconds. Upon receipt of a 2xx response to the SUBSCRIBE request, the Oracle Communications Session Border Controller stores the information for the established dialog and the expiration time. If continued subscription is required, the Oracle Communications Session Border Controller automatically refreshes the subscription to the SIP Event Package for Registrations, either 600 seconds before the expiration time if the initial subscription was for greater than 1200 seconds, or when half of the time has expired if the initial subscription was for 1200 seconds or less.

Following the 200 SUBSCRIBE response, the S-CSCF generates an initial notification, with Event Package state information contained in an XML body.

```

NOTIFY sip:app.example.com SIP/2.0
Via: SIP/2.0/UDP server19.example.com;branch=z9hG4bKnasaij
From: sip:app@example.com;tag=xyzygg
To: sip:sd.example.com;tag=123aa9
Call-ID: 9987@app.example.com
CSeq: 1289 NOTIFY
Contact: sip:server19.example.com
Event: reg
Max-Forwards: 70
Content-Type: application/reginfo+xml
Content-Length: ...

<?xml version="1.0"?>
<reginfo xmlns="urn:ietf:params:xml:ns:reginfo" version="1"
state="partial">
  <registration aor="sip:joe@example.com" id="a7" state="active">
    <contact id="76" state="init" event="registered"
duration-registered="0">
      <uri>sip:joe@pc34.example.com</uri>
    </contact>
  </registration>
</reginfo>

```

## Notification Bodies

Registration state changes are reported in XML attachments to NOTIFICATIONS generated by the S-CSCF. As shown above, the XML consists of one or more registration elements that report the state of a specific address-of-record. Attributes supported by the registration element are as follows:

aor contains the address-of-record

id identifies this specific registration

state reports the Registration state — init, active, or terminated

init — address-of-record not yet cached

active — address-of-record maintained in current cache

terminated — address-of-record removed from cache, not currently valid

registration elements, in turn, contain one or more child contact elements. Attributes supported by the contact element are as follows>

id identifies this specific contact

state reports the Contact state — active or terminated

event reports the event that generated the last state change — registered, created, refreshed, shortened, expired, deactivated, probation, unregistered, or rejected

duration-registered reports the length (in seconds) of the current registration

contact elements, contain a single child uri element that identifies the contact address of FQDN.

## SIP Event Package for Registrations Configuration

Subscription to the SIP Event Package for Registrations is enabled at the SIP interface level.

1. Use the following command sequence to move to **sip-interface** Configuration Mode.

```
ACMEPACKET# configure terminal
```

```
ACMEPACKET(configure)# session-router
```

```
ACMEPACKET(session-router)# sip-interface
```

## SIP Signaling Services

---

```
ACMEPACKET(sip-interface)#
```

2. Use the **subscribe-reg-event** parameter to subscribe to the SIP Event Package for Registrations.

By default, subscription is disabled.

```
ACMEPACKET(sip-interface)# subscribe-reg-event enabled
```

```
ACMEPACKET(sip-interface)#
```

3. Use **done**, **exit**, and **verify-config** to complete enabling the Event Package subscription.

## SIP Transport Selection

---

With this feature enabled, when the Oracle Communications Session Border Controller forwards a message larger than the value specified in the maximum UDP length parameter, it attempts to open an outgoing TCP connection to do so. This connection might fail for a number of reasons; for example, an endpoint might not support UDP, or it might be behind a firewall. The UDP fallback option addresses this condition. If it is configured in SIP interfaces associated with an outgoing message and a TCP session cannot be established, the Oracle Communications Session Border Controller falls back to UDP and transmits the message. When the option is not present, the Oracle Communications Session Border Controller's default behavior is to return the SIP status message 513 Message too Large.

### SIP Transport Selection Configuration

You enable this feature per SIP interface by setting options that control the maximum UDP length and allow UDP fallback:

- **max-udp-length=X** (where X is the maximum length)—Sets the largest UDP packets that the Oracle Communications Session Border Controller will pass. Packets exceeding this length trigger the establishment of an outgoing TCP session to deliver the packet; this margin is defined in RFC 3261. The system default for the maximum UDP packet length is 1500.

You can set the global SIP configuration's **max-udp-length=X** option for global use in your SIP configuration, or you can override it on a per-interface basis by configuring this option in a SIP interface configuration.

- **udp-fallback**—When a request needs to be sent out on the SIP interface for which you have configured this option, the Oracle Communications Session Border Controller first tries to send it over TCP. If the SIP endpoint does not support TCP, however, then the Oracle Communications Session Border Controller falls back to UDP and tries the request again.

To enable SIP Transport Selection:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the **session-router** path.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface
```

4. **options**—Set the options parameter by typing **options**, a Space, the option name **max-udp-length=X** (where X is the maximum UDP length you want to set), and then press Enter.

```
ACMEPACKET(sip-interface)# options +max-udp-length=900
```

**If you type options max-udp-length=X, you will overwrite any previously configured options. In order to append the new option to the sip-interface's options list, you must prepend the new option with a plus sign as shown in the previous example.**

5. **options**—Set the options parameter by typing **options**, a Space, the option name **udp-fallback**, and then press Enter.

```
ACMEPACKET(sip-interface)# options +udp-fallback
```

If you type **options udp-fallback**, you will overwrite any previously configured options. In order to append the new option to the **sip-interface**'s options list, you must prepend the new option with a plus sign as shown in the previous example.

6. Save and activate your configuration.

## uaCSTA NAT Support

The Oracle Communications Session Border Controller offers User Agent Computer Supported Telecommunications Application (uaCSTA) support, which allows for the network address translation (NAT) of a key XML element in SIP INFO messages to use a phone's real contact URI.

### Overview

Certain customers who use a uaCSTA for third party call control have encountered difficulties with the XML in their SIP messages used to support business applications. In these cases, the XML—specifically the <deviceID> XML tag—carries encoded IP addresses that need to be changed as they traverse the Oracle Communications Session Border Controller.

The SIP business application allows users to click-to-dial another party using e-mail application clients. The user's click triggers the application server to send a uaCSTA SIP INFO message through the system to the UA/phone. These SIP INFO messages contain XML with the user's Contact-URI. But the server is only aware of the Oracle Communications Session Border Controller's NAT'd Contact-URI and not the user's, so the XML in the SIP INFO is carrying incorrect information.

The XML element, then, needs to be NAT'd to the phone's real Contact-URI. This is especially important because of the broad use of SIP INFO messages, which instruct a phone to:

- Answer a call
- Hold a call
- Retrieve a call

All of these functions are available via a clickable interface on the e-mail application.

The Oracle Communications Session Border Controller performs the NAT to the <deviceID> XML tag only if it is configured to perform registration caching.

When the Oracle Communications Session Border Controller receives a SIP message from the core side and the request has:

- A Content-Type of application/csta+xml
- A Content-Length greater than 0

it parses the message's message body into an XML document. Should parsing fail, then the Oracle Communications Session Border Controller will forward the SIP INFO request without modification to the XML message body. Otherwise, the Oracle Communications Session Border Controller searches for the <deviceID> subelement within the XML document. If it finds the <deviceID> subelement, the Oracle Communications Session Border Controller searches through its registration cache for a registered Contact that matches the value of the <deviceID>. If it finds a match, the Oracle Communications Session Border Controller replaces the value of the <deviceID> with that of the corresponding registered Contact. If the value of the <deviceID> is a Contact that the Oracle Communications Session Border Controller generates for a registered UA, the corresponding contact from the look-up would be the Contact of the registered UA.

These functions performed, the Oracle Communications Session Border Controller then reformats the SIP INFO request with the modified XML message body before sending it to the next hop. If there is no match found, then the

## SIP Signaling Services

Oracle Communications Session Border Controller forwards the SIP INFO request without modifying the XML message body.

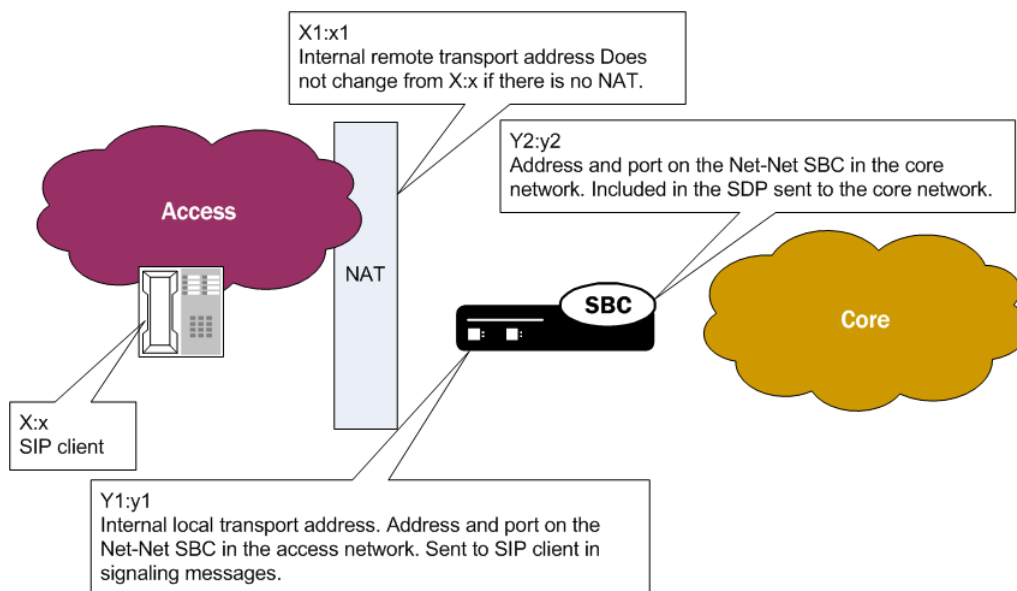
Other than ensuring your Oracle Communications Session Border Controller is configured to perform registration caching, you do not need take any further steps.

## SIP Packet Cable Multi-Media

As a packet cable multi-media (PCMM) enhancement for SIP sessions key to next generation architectures, the Oracle Communications Session Border Controller can now include certain SDP attributes specifying media flow addresses in outgoing SIP messages. Previously, these address were hidden by the Oracle Communications Session Border Controller. Since SIP proxies and application servers in the core network, however, need to know these addresses to guarantee QoS for media flows in packet cable networks.

Certain options in the SIP interface configuration enable the Oracle Communications Session Border Controller to reveal address information on the core side.

When a SIP client in the access network sends and receives RTP media, the Oracle Communications Session Border Controller uses the SIP client's IP address and port (X:x) as its own internal remote transport address. The Oracle Communications Session Border Controller adds this information to outgoing SDP that it sends to the core side, and removes it from incoming SDP. If the SIP client sits behind a NAT, then the Oracle Communications Session Border Controller uses the IP address and port produced from the NAT (X1:x1) process for insertion and removal. The SIP client sends RTP to an IP address and port (Y1:y1) on the Oracle Communications Session Border Controller, referred to as the internal local transport address; this information is included in SDP (included in SIP messages) sent to the SIP client. Meanwhile, the Oracle Communications Session Border Controller also has an IP address and port (Y2:y2) in the core network. The far-end SIP UA sends RTP to this IP address and port, which are also included in SDP the Oracle Communications Session Border Controller sends to the core side.



To enforce QoS properly on the access side, the flow between the SIP client (or the SIP client's post-NAT IP address and port) and the internal local address must be revealed on the core side using SIP signaling messages.

## Details

To enable this enhancement, you set three parameters in the SIP interface configuration:

- **sdp-internals**—Establishes that local and remote transport addresses need to be added. This option must be enabled on the access-side SIP interface, which is where the Oracle Communications Session Border Controller receives SDP.

- **sdp-local=<name>**—Sets a name for the internal local transport port address that the Oracle Communications Session Border Controller inserts into outgoing SDP. This option is configured on the core-side SIP interface. This address is removed from incoming SDP from the core side to prevent attributes from being sent back to the core in a hairpinned call.
- **sdp-remote=<name>**—Sets a name for the internal remote transport address that the Oracle Communications Session Border Controller inserts into outgoing SDP. This option is also configured on the core-side SIP interface. This address is also removed from incoming SDP from the core side to prevent attributes from being sent back to the core in a hairpinned call.

Further, the Oracle Communications Session Border Controller determines whether or not to insert the SDP attributes based on a call's ingress and egress signaling realms:

Address Information	Calling-Side SDP	Called-Side SDP
Internal local transport address	Added to SDP when: The ingress signaling realm's SIP interface has the <b>sdp-internals</b> option configured The egress signaling realm's SIP interface has a defined <b>sdp-local</b> option	Added to SDP when: The egress signaling realm's SIP interface has the <b>sdp-internals</b> option configured The ingress signaling realm's SIP interface has a defined <b>sdp-local</b> option
Internal remote transport address	Added to SDP when: The ingress signaling realm's SIP interface has the <b>sdp-internals</b> option configured The egress signaling realm's SIP interface has a defined <b>sdp-remote</b> option	Added to SDP when: The egress signaling realm's SIP interface has the <b>sdp-internals</b> option configured The ingress signaling realm's SIP interface has a defined <b>sdp-remote</b> option

## Core-Side SDP Insertion Configuration

In a typical configuration intended to send SDP to the core side with the inserted attributes, the access SIP interfaces have the **sdp-internals** option enabled, and the core SIP interfaces have the **sdp-local** and **sdp-remote** values configured.

To set the access SIP interface for SDP insertion on the core side:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-interface)#
```

If you are adding support for this feature to a pre-existing SIP configuration, then you must select (using the CLI **select** command) the configuration that you want to edit.

4. **options**—Set the options parameter by typing **options**, a Space, the option name **sdp-internals** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(sip-interface)# options +sdp-internals
```

**If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.**

5. Save and activate your configuration.

## SIP Signaling Services

---

To set the local and remote transport addresses for a core SIP interface:

6. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

7. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

8. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-interface)#
```

9. **options**—Set the options parameter by typing **options**, a Space, the option name preceded by a plus sign (+) (**sdp-local=<name>**, where the name is attribute name for the SDP), and then press Enter. Follow the same steps to add the **sdp-remote** option.

```
ACMEPACKET(sip-interface)# options +sdp-local=Local_Turn
ACMEPACKET(sip-interface)# options +sdp-remote=PCMM_USERADD
```

If you type either of these options without the plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.

10. Save and activate your configuration.

## SIP Method-Transaction Statistic Enhancements

---

In prior releases, the Oracle Communications Session Border Controller tracks SIP session agents, SIP interfaces and SIP realms on a global level. Only counters that are related to session rates and constraints are displayed.

You can now enable your Oracle Communications Session Border Controller to track transaction messages for specific SIP session agents, SIP realms, and SIP interfaces.

The following SIP methods are tracked for Recent, Total, and Period Max values:

- INVITE | ACK | BYE | REGISTER | CANCEL | PRACK | OPTIONS | INFO | SUBSCRIBE | NOTIFY | REFER | UPDATE | MESSAGE | PUBLISH | other (unknown)

With this new tracking enhancement, the **show sdp** command has been updated with a new method argument which allows you to query statistics for a particular method for a given SIP agent, SIP interface, or SIP realm.

## SIP Method Tracking Enhancements Configuration

To enable or disable the expanded SIP Method statistics tracking:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET(session-router)# sip-config
```

4. **extra-method-stats**—Enable this parameter if you want to use the expanded SIP Method tracking feature. The default is **disabled**. The valid values are:

- enabled | disabled

5. Save and activate your configuration.



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## National Security and Emergency Preparedness for SIP

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The Oracle Communications Session Border Controller supports Emergency Telecommunications Service (ETS), which gives priority treatment of National Security and Emergency Preparedness (NSEP) communications for IP network infrastructures. ETS can increase the likelihood that calls, sessions, and other communications will be successfully completed when they are initiated by government-authorized users over the public network infrastructure. Legacy circuit-switched services such as Government Emergency Telecommunications Service (GETS) and Wireless Priority Service (WPS) also fall under the ETS rubric, and are now also supported on the Oracle Communications Session Border Controller.

To provide this support, you can enable the Oracle Communications Session Border Controller to act on SIP calls that contain an ETS dial number (DN) and/or the SIP Resource-Priority header that carries ETS resource values.

The Oracle Communications Session Border Controller identifies ETS calls by using the system's pre-existing network management controls (NMC) functionality. With NMC and Resource-Priority header (RPH) support enabled on your system, the Oracle Communications Session Border Controller detects ETS calls and provides the appropriate treatment for them.

The Oracle Communications Session Border Controller supports this feature by treating ETS calls based on the r-value parameter in the Resource-Priority header. The r-value is a key piece of information because it defines the resource priority that the call originator requests. The r-value parameter provides namespaces and priorities that the Oracle Communications Session Border Controller can manipulate in outgoing traffic.

In addition to a new RPH profile configuration containing information about how to treat RPHs, new parameters in the global SIP configuration and NMC configuration have been added. The RPH profile is applied to an NMC rule, where they determine r-values, a media policy to use, and what type of call treatment to apply. Also applies to an NMC rule, the new RPH policy configuration provides information about which r-values to insert and which to override.

### Licensing

To enable NSEP for SIP on your Oracle Communications Session Border Controller, you must obtain and install a new license. If properly installed on your system, it appears as NSEP RPH in the display issued when you use the CLI **show** command in the license configuration.

For information about how to obtain an NSEP RPH license, contact your Oracle sales representative.

### Matching by NMC and by RPH

When a Oracle Communications Session Border Controller has been enabled to act on RPH, it checks incoming requests for RPH, tries to parse that RPH, and then rejects requests in the circumstances listed below. For all of these rejections, the Oracle Communications Session Border Controller logs the error at the TRACE level.

- Request with multiple instances of the same namespace in the RPH—The Oracle Communications Session Border Controller sends out a 400 Bad Request response with the Invalid RPH - Namespace repeated header showing that there are multiple instances of the same namespace in the RPH.
- Request with invalid resource priority for a namespace—The Oracle Communications Session Border Controller sends out a 400 Bad Request response with the Invalid RPH - Invalid rvalue: x showing that there is an invalid resource value (where x is the invalid value).
- Request with WPS namespace, but without ETS namespace—The Oracle Communications Session Border Controller sends out a 400 Bad Request response with the Invalid RPH - No ETS value header showing that there is no ETS namespace.

If the Oracle Communications Session Border Controller successfully parses the RPH, it identifies the ETS call by checking the Request-URI of the incoming request against destination identifiers that you configure in the NMC rules. If there is a match between the request's ETS DN and the destination value identifier in the NMC rules, the Oracle Communications Session Border Controller tags the call; note that NMC rules need to be configured with the **rph-feature** parameter set to enabled to identify an ETS call properly. If there is no match to an NMC rule, then

## SIP Signaling Services

the Oracle Communications Session Border Controller performs matching based on RPH by comparing resource values (r-values) in the RPH with values you set in the RPH profile configuration.

For an ETS call that matches by ETS DN and NMC rule, the system checks the NMC rule to determine if it has an RPH profile (with r-values) assigned to it. If so, the Oracle Communications Session Border Controller continues by comparing the RPH profile's r-values against those in the request's RPH. In cases where the RPH does not contain a recognized value r-value, the Oracle Communications Session Border Controller:

- Processes the call as it normally would (as a non-ETS call) without changing the RPH if the resource-priority option tag is not present in the Required header (for an INVITE only and not any other requests or response from which RPH would be deleted)
- Rejects the Request when the Require header has the resource-priority header; or, inserts an Accept-Resource-Priority header (ARPH) in the response if the **insert-arp-header** parameter option is enabled

However, the call goes through the Oracle Communications Session Border Controller as an ETS call when it is matched by ETS DN and the applicable NMC does not have an RPH profile assigned. According to the settings in the NMC rule, the Oracle Communications Session Border Controller either diverts or rejects such a call. And when the call matches by RPH rather than ETS DN, the Oracle Communications Session Border Controller applies the configured RPH profile from the relevant NMC rule.

It can be the case that non-ETS calls have RPH in their requests. Here, the Oracle Communications Session Border Controller call treatment is performed according to the settings in the matching RPH profile when there is no matching NMC rule. When you configure treatment as "reject," then the Oracle Communications Session Border Controller rejects the call with a 417 Unknown-Resource Priority status code. When you set the treatment to either "accept" or priority, the Oracle Communications Session Border Controller allows the call to proceed as a non-ETS call or as a priority call.

The ETS r-value can appear in ACK, BYE, INFO, PRACK, REFER and UPDATE requests. In cases when it does and the session with which the request is associated is a non-ETS call, the Oracle Communications Session Border Controller removes the RPH from the request before forwarding it and logs a TRACE-level error. The Oracle Communications Session Border Controller also removes RPH from responses before forwarding them and logs a TRACE-level error when responses contain RPH headers with ETS values for non-ETS sessions.

## Call Treatment

This section describes how ETS calls are treated as they traverse the Oracle Communications Session Border Controller.

Call Treatment	Description
Routing	ETS calls are routed the same way as any other calls are, except when the applicable NMC rule's treatment type is divert, and rule defines the next hop. This route takes precedence over other normal routes.
Local NMC	ETS calls are exempt from the local NMC, including: session agent constraints, bandwidth constraints (e.g., per-realm bandwidth), per-user CAC, and CPU constraints. However, the call is subject to the ETS congestions control threshold. Licensing session constraints apply.
ETS Call Congestion Control	ETS calls are subject to congestion control constraints that you configure specifically for this type of traffic. In the global SIP configuration, you set up one option that defines a load limit (greater than that set for normal calls).
ETS CAC	Although the Oracle Communications Session Border Controller uses the call rate control value in the applicable NMC rule, you can also enforce call rate on a per-user basis for ETS calls.

When the Oracle Communications Session Border Controller receives a SIP INVITE with an RPH matching an NMC with an ETS DN, but whose r-values do not match the NMC's rph-profile, the Oracle Communications Session Border Controller behaves as follows:

- If the INVITE does not have the resource-priority option tag and:

- If the matching NMS is set to **PRIORITY**, the call will be treated as an NSEP call. If there is an rph-profile matching the r-value (not necessarily the one in the NMC), the Oracle Communications Session Border Controller uses the media-policy from that rph-profile for the call. The rph-policy from the NMC (if present) also applies to the call.
- If the matching NMC is not set to **PRIORITY**, the Oracle Communications Session Border Controller will treat the call as a normal one.

If the INVITE contains the resource-priority option tag, the Oracle Communications Session Border Controller will reject the call with the 417 Unknown Resource-Priority message.

## Generating Egress RPH

For each ETS call, the Oracle Communications Session Border Controller generates RPH for the outgoing request. It forms this RPH according to the information in the NMC rule. The outgoing request types are INVITE, ACL, BYE, CANCEL, INFO, PRACK, REFER, and UPDATE.

Request RPH Status	Generated Egress RPH
Incoming request without RPH (matched by ETS DN)	Outgoing RPH value becomes the r-value set in the insert-r-value parameter in the RPH policy applied to the NMC rule.
Incoming request without RPH (matched by ETS DN)	If the insert-r-value parameter is empty in the RPH policy applied to the NMC rule or there is no RPH policy applied to the NMC rule, then the egress RPH will also not have RPH.
Incoming request has RPH	Egress RPH is the same as the ingress if the NMC rule has an RPH policy applied but the override-r-value for the policy is empty or if there is not RPH policy applied to the NMC rule.  If the override-r-value for the policy is set, then the egress RPH is set to that value.

For example, given an incoming request with the resource priority ets.0, dsn.flash and an RPH policy with an override value of wps.1,ets.1, the egress request would be sent with a resource-priority of wps.1,ets.1,dsn.flash.

The Oracle Communications Session Border Controller also includes RPH in the following series of responses, even when the downstream SIP entity does not respond with an RPH: 1xx, 2xx, 3xx, 4xx, 5xx, and 6xx. The 401 Unauthorized response is an exception.

## Media Treatment

If the RPH profile set in an NMC names a media policy, then the Oracle Communications Session Border Controller implements it for the ETS call. This media policy overrides any media policy set in the realm configuration.

The possible Differentiated Services Code Point (DSCP) values for an ETS call are:

- Audio—Applied to the respective media for an ETS call
- Video—Applied to the respective media for an ETS call
- SIP—Applied to the ETS calls' SIP signaling messages, only for the egress call leg for the ETS session

## RPH Configuration

This section shows you how to configure RPH profiles and policies that enable the Oracle Communications Session Border Controller to act on SIP calls that have an ETS DN and/or an RPH carrying ETS resources values. There are also settings for the global SIP configuration and for the NMC rule configuration that support this feature.

In addition, note that:

- You must set a media policy for the RPH profile to use. Check your system configuration and note the name of the media policy that best suits your needs.
- Valid values for the parameters that take r-values are wps.x and ets.x, where x is 0 through 4.

Remember to save and activate your configuration after you have completed the processes detailed in this section.

### Setting Up and Applying RPH Policy

The RPH policy is a configuration on the Oracle Communications Session Border Controller that you apply to NMC rules. It designates the following for ETS/WPS namespaces:

- An override resource value—Resource value used to override the incoming RPH’s resource value
- An insert resource value—Resource value inserted when the Oracle Communications Session Border Controller does not recognize the RPH, the incoming request has no RPH, or the call is H.323 and matches an NMC rule based on the ETS DN

Note that RPH policies do not apply for DSN, DRSN, Q.735, or any other type of namespace; these remain untouched in outgoing requests.

To configure an RPH policy:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **rph-policy** and press Enter. From here, you can configure the individual parameters for the RPH policy.

```
ACMEPACKET (session-router) # rph-policy  
ACMEPACKET (rph-policy) #
```

4. **name**—Enter the name that uniquely identifies this RPH policy. This is the value you use to apply the policy in the NMC rules configuration. There is no default for this parameter, and you are required to set it.

5. **override-r-value**—Enter the value that the system uses to override r-values in the original RPH.

```
ACMEPACKET (rph-policy) # override-r-value ets.1
```

6. **insert-r-value**—Enter the value that the Oracle Communications Session Border Controller inserts into the RPH.

```
ACMEPACKET (rph-policy) # insert-r-value wps.1
```

7. **rph-policy**—Enter the name of the RPH policy that you want to apply for this NMC rule. This parameter is empty by default; if you do not set an RPH policy, none will be applied.

### Setting Up and Applying RPH Profile

The RPH profile contains information about how the system should act on the namespace(s) present in a Resource-Priority header (if any). The list of resource values in this configuration calls out the resource values (or r-values) recognizable to the Oracle Communications Session Border Controller; the ETS and WPS namespaces are supported.

You also set a media policy for the RPH profile to use; it defines the Differentiated Services Code Point (DSCP) that the Oracle Communications Session Border Controller uses for media or signaling packets belonging to the egress call leg for the ETS session.

The call treatment parameter tells the Oracle Communications Session Border Controller what to do with a non-ETS call that has RPH in its request; the call can be allowed, rejected, or treated as a priority call.

To configure an RPH profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **rph-profile** and press Enter. From here, you can configure the individual parameters for the RPH policy.

```
ACMEPACKET(session-router)# rph-profile
ACMEACMEPACKETPACKET(rph-profile)#
```

4. **name**—Enter the name that uniquely identifies this RPH profile. This is the value you use to apply the profile in the NMC rules configuration. There is no default for this parameter, and you are required to set it.
5. **r-values**—Enter one or more r-values that the Oracle Communications Session Border Controller is to recognize for matching purposes. When you enter more than one value in the list, you type the name of the parameter followed by a Space, open quotation mark, the values for the list separated by spaces, a closed quotation mark. Then press Enter.

You must enter them in the order reflected below (a WPS and then an ETS value). A WPS call always has to have an ETS namespace.

```
ACMEPACKET(rph-profile)# r-values "wps.0 ets.2"
```

6. **media-policy**—Enter the name of a media policy configuration that you want applied for this RPH profile. The Oracle Communications Session Border Controller implements this media policy for the ETS call, and this media policy overrides any media policy set in the realm configuration.
7. **call-treatment**—Enter the call treatment method for a non-ETS call that contains RPH matching it to this profile. The default is accept. The valid values are:
  - **accept**—The call proceeds as it normally would
  - **reject**—The Oracle Communications Session Border Controller rejects the call with the 417 Unknown-Resource Priority status code
  - **priority**—The Oracle Communications Session Border Controller treats the call as a priority call

### Enabling NSEP for an NMC Rule

In addition to the RPH policy and RPH profile you can set for an NMC rule, you also need to set the state of this feature for the NMC rule.

To enable NSEP for an NMC rule:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **net-management-control** and press Enter.

```
ACMEPACKET(session-router)# net-management-control
ACMEPACKET(net-management-control)#
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the ACLI **select** command) the configuration that you want to edit.

4. **rph-feature**—Enable this parameter if you want to turn the NSEP feature on for this NMC rule. The default is **disabled**. The valid values are:
  - enabled | disabled

### Global SIP Configuration Settings Enabling NSEP

For the global SIP configuration, you can turn the NSEP feature on, and you can also set parameters that support call admission and congestion control.

In addition, you can enable the insertion of the ARPH header in a response when the resource-priority tag is present in the Require header and the Oracle Communications Session Border Controller rejects the request with a 417 Unknown Resource-Priority response. The ARPH value is the list of r-values you set in the RPH profile.

To enable NSEP for the global SIP configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the ACLI **select** command) the configuration that you want to edit.

4. **rph-feature**—Enable this parameter if you want to turn the NSEP feature on for the global SIP configuration. The default is **disabled**. The valid values are:
  - enabled | disabled

### Global SIP Configuration Settings Enabling CAC and Congestion Control

To set call admission and congestion control parameters for NSEP:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the ACLI **select** command) the configuration that you want to edit.

4. **nsep-user-sessions-rate**—Enter the maximum INVITEs per second to admit for ETS calls on a per-user basis. To enable NSEP call admission control (CAC), you must change the parameter value from 0; if you leave this parameter set to 0, then it is the same as disabling CAC for ETS calls. The default is **50**. The valid range is:
  - Minimum—0
  - Maximum—999999999

5. **options**—To enable congestion control for ETS calls, you configure an option that sets the CPU threshold. If this threshold is exceeded, the Oracle Communications Session Border Controller rejects new ETS calls with the 503 Service Unavailable response. The value you set here should be larger than the load limit value for normal calls; ETS calls are allowed even when the load limit threshold for normal calls is exceeded.

The threshold value can be between 0 and 100. Using a value of 0 or 100 for this parameter disables ETS call congestion control.

Set the options parameter by typing **options**, a Space, the option name **nsep-load-limit** with a plus sign in front of it, then the equal sign and the ETS call threshold you want to set. Then press Enter.

```
ACMEPACKET(sip-config)# options +nsep-load-limit=50
```

**If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.**

## Global SIP Configuration Settings Enabling ARPH Insertion

To enable ARPH insertion in responses:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the CLI **select** command) the configuration that you want to edit.

4. **options**—To enable ARPH insertion in responses type **options**, a Space, the option name **insert-arp-header** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(sip-config)# options +insert-arp-header
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

## Setting Up NSEP for Session Agents

In earlier releases, the Oracle Communications Session Border Controller supports NSEP-related CAC for users and for NMC. You can now configure a sessions-per-second rate for session agents. Set in the global SIP configuration, this rate applies to all SIP session agents. When session exceed the limit, the Oracle Communications Session Border Controller rejects them with a 503 Service Unavailable message.

To configure NSEP limits for SIP session agents:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

4. **nsep-sa-sessions-rate**—Enter maximum acceptable number of SIP INVITES (NSEP sessions) per second to allow for SIP session agents. This parameter defaults to 0, meaning there is no limit.
5. Save and activate your configuration.

## E-CSCF Emergency Setting Precedence for NMC

When the Oracle Communications Session Border Controller acts as an E-CSCF, it can route emergency or priority calls (i.e., 112, 911, 999 calls) to the corresponding ECS/PSAP based on the calling party's information. However, for registered users, this ability mixes with the Oracle Communications Session Border Controller's NMC function so that the Service Routes takes precedence over the NMC. So rather than routing the emergency call to the ECS/PSAP, the call ends up at the S-CSCF of the Service Route.

For non-registered users where there no Service Route exists, this is not an issue.

## SIP Signaling Services

---

Adding the **apply-local-policy** value to the options parameter in the network management controls configuration allows the NMC to take precedence over cached Service Route when a session matches an NMC rule. Instead, it locates a matching local policy.

Without the NMC configured to take precedence, the Oracle Communications Session Border Controller does not function optimally as an E-CSCF for registered users.

Consider the following scenario in which a UE registers as a P-CSCF to the S-CSCF via the Oracle Communications Session Border Controller. In this case, the registrar returns a Service Route header in a 200 OK response to the REGISTER message to the Oracle Communications Session Border Controller. Or, if an implicit service route is enabled, the Oracle Communications Session Border Controller generates a Service Route that it saves in the register cache before forwarding the 200 OK on to the UA. Then when a new INVITE comes from the UA, the Oracle Communications Session Border Controller checks its register cache and uses the Service Route as the next hop—without taking local policy into consideration.

Using the **apply-local-policy** value requires local policy's consideration in deciding the next hop. If you configure this option and the Oracle Communications Session Border Controller receives a new INVITE, it will decide whether the session is priority, registered, or with Service Route. Then it will determine if the call is E-CSCF. If so and the first matching local policy has E-CSCF enabled, the priority local policy is applied over the Service Route. The Oracle Communications Session Border Controller stays with the Service Route if not.

## E-CSCF Emergency Configuration

This section shows you how to configure a network management control rule to take precedence over a Service Route.

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

3. Type **net-management-control** and press Enter.

```
ACMEPACKET(session-router) # net-management-control
ACMEPACKET(net-management-control) #
```

4. **options**—Follow your entry with this value:

- +disable-temp-file

```
ACMEPACKET(net-management-control) # options +apply-local-policy
```

**This value allows a network management control rule to take precedence over a service route for an emergency or priority call.**

5. Type **done** and continue.

## SIP TCP Connection Reuse

---

You can configure your Oracle Communications Session Border Controller to reuse TCP connections created by SIP peering devices for outgoing SIP in-dialog and out-of-dialog request transactions.

The SIP draft draft-ietf-sip-connect-reuse-07.txt describes a way for SIP UAs to reuse connections created by a remote endpoint for outgoing requests for TLS. The Oracle Communications Session Border Controller does not support the model connection reuse signalled by a parameter; rather, it is provisioned on a per-session-agent basis.

You enable SIP TCP connection reuse on a per-session-agent basis. The Oracle Communications Session Border Controller checks incoming TCP connection request to determine if they are from session agent that has this feature turned on. When it is, the Oracle Communications Session Border Controller adds the connection's source address to its list of alias connections. This is a list of connections that the Oracle Communications Session Border Controller



can use for outgoing requests rather than creating its own connection (as it does when this feature is not enabled). So if a preferred connection fails, the Oracle Communications Session Border Controller can refer to this list and use the alias connection.

The presence of an alias parameter in the Via header is just one mechanism that will call the Oracle Communications Session Border Controller to use the inbound TCP/TLS connection for outbound requests. The Oracle Communications Session Border Controller will automatically add an alias for the inbound connections in the following circumstances:

- The other end of the connection is behind a NAT. When the Oracle Communications Session Border Controller sees that the Via sent-by does not match the source address of the connection, it will automatically reuse the connection to deliver requests to the UA.
- The Contact address of a REGISTER request received on a TCP connection matches the source address and port. This is because the contact address is the ephemeral port the UA used to form the connection to the Oracle Communications Session Border Controller and, therefore, will not be listening on that port for inbound connections.
- The presence of reuse-connections in the options field of the sip-interface will cause the Oracle Communications Session Border Controller to reuse all inbound TCP connections for sending requests to the connected UA.

## SIP TCP Connection Reuse Configuration

This section describes how to enable SIP TCP connection reuse for a session agent. Currently there are two options for the new **reuse-connections** parameter: **none** (which turns the feature off) and **tcp** (which enables the feature for TCP connections). You also set the re-connection interval.

To enable SIP TCP connection reuse for a session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

If you are adding support for this feature to a pre-existing session agent, then you must select (using the ACLI **select** command) the session agent that you want to edit.

4. **reuse-connections**—Enable or disable SIP TCP connection reuse. The default is **none**. This value disables the feature. The valid values are:
  - **tcp** | **none**
5. **tcp-reconn-interval**—Enter the amount of time in seconds before retrying a TCP connection. The default for this parameter is **0**. The valid range is:
  - Minimum—0, 2
  - Maximum—300
6. Save and activate your configuration.

## SIP TCP Keepalive

The Oracle Communications Session Border Controller supports a special TCP keepalive mechanism for SIP. By enabling this feature either for a session agent or for a SIP interface, you allow the Oracle Communications Session Border Controller to use standard keepalive probes to determine whether or not connectivity with a remote peer has been lost.

## SIP Signaling Services

---

This feature adds to the Oracle Communications Session Border Controller's pre-existing TCP keepalive functionality that you can enable in the network parameters configuration. Using existing functionality, you can customize keepalive timing by:

- Specifying the number of unacknowledged packets the Oracle Communications Session Border Controller sends to the remote peer before it terminates the TCP connection.
- Specifying the number of seconds of idle time before TCP keepalive messages are sent to the remote peer.

You can now set three modes for TCP keepalive for session agents and SIP interfaces:

- **none**—(Default) Keepalives are not enabled for use with the session agent/SIP interface; when you select this setting for a session agent, it will use the setting for this feature from the SIP interface.
- **enabled**—Keepalives are enabled for the session agent/SIP interface.
- **disabled**—Keepalives are disabled for the session agent/SIP interface.

Note that the setting for this feature for a session agent takes precedence over that for a SIP interface. In addition, the session agent offers you a way to set the re-connection interval.

### SIP TCP Keepalive Configuration for Session Agents

To enable SIP TCP keepalive for session agents:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET (session-router) # session-agent
ACMEPACKET (session-agent) #
```

If you are adding support for this feature to a pre-existing session agent, then you must select (using the ACLI **select** command) the session agent that you want to edit.

4. **tcp-keepalive**—Enable or disable standard keepalive probes to determine whether or not connectivity with a remote peer is lost. The default value is **none**. The valid values are:

- none | enabled | disabled

```
ACMEPACKET (session-agent) # tcp-keepalive enabled
```

5. Save and activate your configuration.

### SIP TCP Keepalive Configuration for SIP Interfaces

To enable SIP TCP keepalive for SIP interfaces:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-interface) #
```

If you are adding support for this feature to a pre-existing SIP interface, then you must select (using the ACLI **select** command) the SIP interface that you want to edit.

4. **tcp-keepalive**—Enable or disable SIP TCP keepalive. The default value is **none**. The valid values are:

- none | enabled | disabled

```
ACMEPACKET (session-agent) # tcp-keepalive enabled
```

5. Save and activate your configuration.

## SIP Enforcement Profile and Allowed Methods

For this feature, you use a configuration called an enforcement profile that allows you to configure sets of SIP methods that you want applied to: the global SIP configuration, a SIP interface, a realm, or a SIP session agent. The enforcement profile is a named list of allowed methods that you configure and then reference from the configuration where you want those methods applied.

### SIP Enforcement Profile Configuration

To use the enforcement profile, you need configure it with a name and the list of SIP methods you want to designate as allowed. Then you need to configure the global SIP configuration, a SIP interface, a realm, or SIP session agent to use the set.

#### Setting Up and Enforcement Profile

To set up an enforcement profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **enforcement-profile** and press Enter.

```
ACMEPACKET (session-router) # enforcement-profile
ACMEPACKET (enforcement-profile) #
```

4. **name**—Enter the name for the enforcement profile. This parameter has no default, but you must note it so that you can apply this set of allowed SIP headers in: the global SIP configuration, a SIP interface, a realm, or SIP session agent.

```
ACMEPACKET (enforcement-profile) # name EnfProfile1
```

5. **allowed-methods**—Enter a list of SIP methods that you want to allow for this set. The default value is **none**. Valid values are:

- INVITE | REGISTER | PRACK | OPTIONS | INFO | SUBSCRIBE | NOTIFY | REFER | UPDATE | MESSAGE | PUBLISH

To enter multiple methods for the list, type the parameter name followed by a space, then the names of all methods you want to include each separated by a only a comma and in capital letters.

```
ACMEPACKET (enforcement-profile) # allowed-methods INVITE,REGISTER,PRACK
```

6. Save and activate your configuration.

#### Applying an Enforcement Profile

You can apply an enforcement profile to: the global SIP configuration, a SIP interface, a realm, or SIP session agent. This section shows you how to do all four. Remember that if you are adding this functionality to a pre-existing configuration, you need to select the configuration you want to edit.

To apply an enforcement profile to the global SIP configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) #
```

4. **enforcement-profile**—Enter the name of the enforcement profile you want to apply to the global SIP configuration.

5. Save and activate your configuration.

To apply an enforcement profile to a SIP interface:

6. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

7. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

8. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-interface) #
```

9. **enforcement-profile**—Enter the name of the enforcement profile you want to apply to this SIP interface.

10. Save and activate your configuration.

To apply an enforcement profile to a SIP session agent:

11. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

12. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

13. Type **session-agent** and press Enter.

```
ACMEPACKET (session-router) # session-agent
ACMEPACKET (session-agent) #
```

14. **enforcement-profile**—Enter the name of the enforcement profile you want to apply to this session agent.

15. Save and activate your configuration.

To apply an enforcement profile to a realm:

16. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

17. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

18. Type **realm-config** and press Enter.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

19. **enforcement-profile**—Enter the name of the enforcement profile you want to apply to this realm.

20. Save and activate your configuration.

## Local Policy Session Agent Matching for SIP

When you enable the local policy session agent matching option in your global SIP configuration, you change the way local policies match session agents. Normally, the Oracle Communications Session Border Controller looks up and stores matched session agents configured as next hops so it does not need to perform the lookup while processing requests. In this type of matching, the Oracle Communications Session Border Controller does take the realm set in the local policy attributes into consideration. When the Oracle Communications Session Border Controller performs its regular matching method and you have enabled overlapping IP addresses for session agents, the Oracle Communications Session Border Controller might match session agents to different realms than the ones you intended when creating your configuration.

Local policy session agent matching provides a way to match session agents differently, taking realms and nested realms into consideration during the matching process. This difference is key to deployments with multiple peering partners that use the overlapping IP address feature, and have multiple local policies routing to the same IP address in different realms where some target next hops require session constraints but others do not. In the cases where no session constraints are required, session agents are not needed. But session agents still match the local policy, applying their constraints, because they match the next hop IP address.

In addition to modifying this behavior, this feature also affects the use of realms and nested realms. It triggers the use not only of realms, but of all the realms nested however deeply—thereby improving matching efficiency.

You can set the local policy session agent matching option with values that define how the Oracle Communications Session Border Controller performs session agent matching:

- **any**—The Oracle Communications Session Border Controller looks up and stores matched session agents configured as next hops so it does not need to perform the lookup while processing requests, without regard to realms.

This behavior is the default when the SIP configuration does not have the local policy session agent matching option set.

- **realm**—The Oracle Communications Session Border Controller selects session agents in the realm that the local policy attribute indicates; this provides an exact match, rather than not taking the realm into consideration during session agent selection.

For example, the session agent is a match if the session agent **realm-id** and the local policy attribute **realm** parameters are an exact match.

- **sub-realm**—Session agents in the same realm or the same realm lineage—where session agents and realms are related to one another via realm parent-child relationships no matter the depth of realm nesting configured

For example, the session agent is a match if the local policy attribute **realm** is a sub-realm of the realm specified in the session agent **realm-id** parameter.

- **interface**—Session agents in the same realm or same realm lineage via the realm set in the local policy attribute, and whose realm uses the same signaling interface as the realm set in the local policy attribute

For example, the session agent is a match if the session agent **realm-id** is a sub-realm of the local policy attribute **realm**, and both referenced realms use the same SIP signaling interface.

- **network**—Session agents whose realm is in the realm lineage for the same realm set in the local policy attributes, and whose realm is associated with the same network interface as the realm set in the local policy attributes

For example, the session agent is a match if the session agent **realm-id** is a sub-realm of the local policy attribute **realm**, and realm reference by both use the same network interface.

If it cannot find a match, the Oracle Communications Session Border Controller will use the IP address as the next hop. Further, requests matching local policy attributes will not be associated with session agents, and so their constraints will not be applied.

## SIP Signaling Services

---

The Oracle Communications Session Border Controller stores session agent information that it looks up when performing local policing session agent matching. To perform the lookup, it uses the session agent hostname as a key. When the hostname is an FQDN and there is a configured IP address in the **ip-address** parameter, the Oracle Communications Session Border Controller uses the ip-address value as a secondary key. Given this implementation, the following are true when selecting session agents:

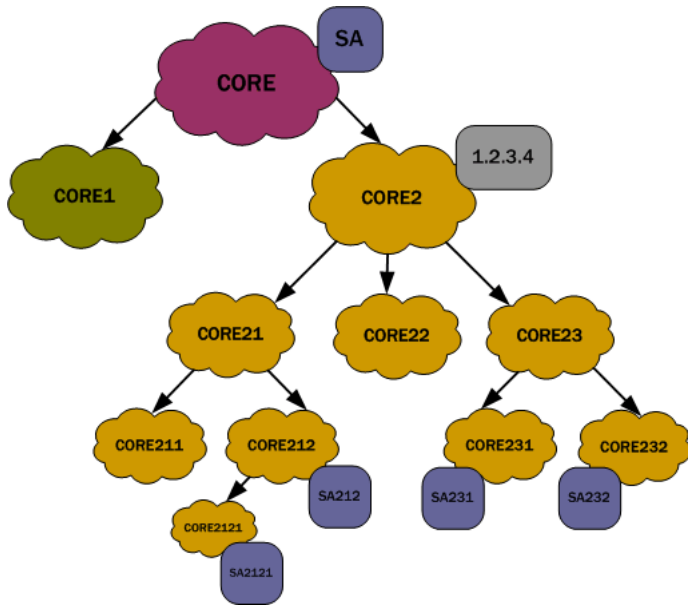
- If multiple session agents share the same IP address, the one with an IP address in the hostname parameter takes precedence.
- If all session agents with the same IP address have an FQDN as their hostname, the one whose name is alphabetically lower will take precedence, where alphabetically lower means earlier in the alphabet (closer to A than to Z).
- For non-global session agents (whose realms are configured but not wildcarded) with an IP address, the Oracle Communications Session Border Controller uses a key that is a combination of the IP address and the realm in the form `<address>:<realm>`.
- For a session agent whose realm has a parent realm, the Oracle Communications Session Border Controller uses a combination of the IP address, realm, and realm-path (or lineage for the realm) in the form `<address>:<realm-path>`. For example, the realm path for a realm `core3` with a parent `core2`, which in turn has a parent `core` would be `core:core2:core3`.

When it looks up a session agent with a realm, the Oracle Communications Session Border Controller first searches for an exact match for the IP address and realm combination. If this fails, it performs a second search if the desired realm has parents or children. The Oracle Communications Session Border Controller locates an entry in its repository of session agent information that is greater than or equal to the IP address with the base realm, which is the ancestor of the desired realm without a parent. Having gathered this set of candidates, the Oracle Communications Session Border Controller narrows down the search for a match by comparing sub-realms and determines there is a match if either:

- The desired realm path is a sub-string of the entry's realm path, or
- The entry's realm path is a substring of the desired realm path (i.e., the desired realm is a sub-realm of the entry's realm)

Then the Oracle Communications Session Border Controller orders the candidates by depth of the entry's realm-path, or number of levels from the base realm relative to the depth of the desired realm. By searching the ordered set until the entry's realm depth equals the desired realm's depth, the Oracle Communications Session Border Controller determines a parent candidate, all subsequent entries being sub-realms of the desired realm. The Oracle Communications Session Border Controller only considers entries at the first level deeper than the desired realm. If at this point there is only one entry, the Oracle Communications Session Border Controller deems it a match. Otherwise, it selects the parent candidate as the matching entry. In the event the search does not yield a matching realm, the Oracle Communications Session Border Controller uses the global session agent for the IP address, if there is one.

The following diagram shows the realm tree, where the clouds are realms and squares are session agents, representing a group of session agents sharing the IP address 1.2.3.4. The Oracle Communications Session Border Controller searches for the session agents lower in the tree along the session agent realm-path and the desired realm.



For the diagram above, the following shows how the hostname would look for this group of session agents.

Key	Session Agent (hostname[realm])
1.2.3.4 (This session agent owns the primary key for the IP address because its hostname is the IP address.)	1.2.3.4[CORE2]
1.2.3.4:CORE (IP+realm key entry)	SA[CORE]
1.2.3.4:CORE (IP+realm key entry)	1.2.3.4[CORE2]
1.2.3.4:CORE212 (IP+realm key entry)	SA212[CORE212]
1.2.3.4:CORE2121 (IP+realm key entry)	SA2121[CORE2121]
1.2.3.4:CORE231 (IP+realm key entry)	SA231[CORE231]
1.2.3.4:CORE232 (IP+realm key entry)	SA232[CORE232]
1.2.3.4:CORE: (IP+realm-path key entry)	SA[CORE]
1.2.3.4:CORE:CORE2: (IP+realm-path key entry)	1.2.3.4[CORE2]
1.2.3.4:CORE2:CORE21:CORE212	SA212[CORE212]

## SIP Signaling Services

Key	Session Agent (hostname[realm])
(IP+realm-path key entry)	
1.2.3.4:CORE2:CORE21:CORE212:CORE2121 (IP+realm-path key entry)	SA2121[CORE2121]
1.2.3.4:CORE2:CORE23:CORE231 (IP+realm-path key entry)	SA231[CORE231]
1.2.3.4:CORE2:CORE23:CORE232 (IP+realm-path key entry)	SA232[CORE232]

For each realm in the table above, the search results for each realm would look like this:

IP Address	Realm	Session Agent (hostname[realm])
1.2.3.4	CORE	SA[CORE]
1.2.3.4	CORE2	1.2.3.4[CORE2]
1.2.3.4	CORE21	SA212[CORE212]
1.2.3.4	CORE211	1.2.3.4[CORE2]
1.2.3.4	CORE212	SA212[CORE212]
1.2.3.4	CORE2121	SA2121[CORE2121]
1.2.3.4	CORE22	1.2.3.4[CORE2]
1.2.3.4	CORE23	1.2.3.4[CORE2]
1.2.3.4	CORE231	SA231[CORE231]
1.2.3.4	CORE232	SA232[CORE232]

## Local Policy Session Agent Matching Configuration

When you enable local policy session agent matching, remember that you can choose from five different ways to use the feature: **all**, **realm**, **sub-realm**, **interface**, and **network**.

This example shows you how to use the **realm** selection.

To enable local policy session agent matching using the realm method:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

4. **options**—Set the options parameter by typing options, a Space, the option name **lp-sa-match=X** (where **X is the local policy session agent matching method you want to use**) with a plus sign in front of it. Then press Enter.



Remember that if you do not specify a method, the system uses the **all** method.

```
ACMEPACKET(sip-config)# options +lp-sa-match=realm
```

If you type **options** and then the option value for either of these entries without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

#### 5. Save and activate your configuration.

- Unordered—Meaning that the endpoint can deliver data within regard for their stream sequence number

You set this preference in the network parameters configuration.

## About Wildcarding

The Oracle Communications Session Border Controller supports wildcarding the event type in the **subscribe-event** configuration. To wildcard the value, you enter an asterisk (\*) for the **event-type** parameter instead of typing in the name of an actual event type.

When you wildcard this value, the Oracle Communications Session Border Controller applies the subscription limitations you set across all event types. Or, if you have entered multiple subscribe-event configurations, the Oracle Communications Session Border Controller applies the wildcard limits across the event types for which you have not set limits.

Consider the following example of a configured enforcement profile with a wildcarded **subscribe-event** configuration:

```
enforcement-profile
  name rulefour
  allowed-methods
  sdp-address-check disabled
  subscribe-event
    event-type *
    max-subscriptions 1
  subscribe-event
    event-type xyz
    max-subscriptions 0
  last-modified-by admin@console
  last-modified-date 2008-11-11 12:49:27
```

In this example, the enforcement profile allows all subscriptions that are event type xyz for a user. But it allows only one maximum for every other subscription event type.

## Monitoring

You can display the number of subscription dialogs per SUBSCRIBE event type using the ACLI **show registration sipd subscriptions-by-user** command. You can display this information per event type, or you can show data for all event types by wildcarding the event type argument.

## Enforcement Profile Configuration with subscribe-event

This section shows you how to configure an enforcement profile with a **subscribe-event** configuration. Remember that you can set up multiple **subscribe-event** configurations to correspond with the event types you want to control. It also shows you how to apply these limitations to a realm.

### Setting Up Subscribe Dialog Limits

Setting up subscribe dialog limits means setting up an enforcement profile. For the sole purpose of setting up the subscription event limits, you only need to configure the name parameters and then as many **subscribe-event** configurations as you require. The enforcement profile has other uses, such as SIP SDP address correlation, so only configure the parameters you need.

To configure subscribe dialog limits:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **enforcement-profile** and press Enter.

```
ACMEPACKET (session-router) # enforcement-profile
ACMEPACKET (enforcement profile) #
```

4. **name**—Enter a name for this enforcement profile. You will use this name later when you apply the enforcement profile to a realm; it is the value you enter into the **enforcement-profile** parameter in the realm configuration.

5. Still in the enforcement profile configuration, type **subscribe-event** and press Enter.

```
ACMEPACKET (enforcement profile) # subscribe-event
ACMEPACKET (subscribe-event) #
```

6. **event-type**—Enter the SIP subscription event type for which you want to set up limits. You can also wildcard this value (meaning that this limit is applied to all event types except the others specifically configured in this enforcement profile). To use the wildcard, enter an asterisk (\*) for the parameter value.

By default, this parameter is blank.



**Note:** The value you enter must be configured as an exact match of the event type expected in the SIP messages (except for the wildcard). Further, the value conforms to the event type BNF specified in RFC 3265.

7. **max-subscriptions**—Enter the maximum number of subscriptions allowed to a user for the SIP subscription event type you entered in the **event-type** parameter. Leaving this parameter set to 0 (default) means that there is no limit. You can set this parameter to a maximum value of 65535.

8. If you are entering multiple **subscribe-event** configurations, then you save them each by using the CLI **done** command and then repeat Steps 6 and 7 to configure a new one. If you do not save each, then you will simply overwrite the first configuration repeatedly.

```
ACMEPACKET (subscribe-event) # done
```

9. When you finish setting up **subscribe-event** configurations and have saved them, exit to return to the enforcement profile configuration.

```
ACMEPACKET (subscribe-event) # exit
```

10. You also need to save the enforcement profile configuration.

```
ACMEPACKET (enforcement profile) # done
```

### Applying an Enforcement Profile to a Realm

For the Oracle Communications Session Border Controller to use the limits you have set up, you need to apply them to a realm.

To apply an enforcement profile to a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter. If you are adding this feature to a pre-existing realm configuration, you will need to select and edit your realm.

```
ACMEPACKET(media-manager) # realm-config
ACMEPACKET(realm-config) #
```

4. **enforcement-profile**—Enter the name of the enforcement profile you want to apply to this realm. This value corresponds to the **name** parameter in the enforcement profile configuration. This parameter has no default value.
5. Save and activate your configuration.

## STUN Server

The Oracle Communications Session Border Controller supports RFC 3489, which defines Simple Traversal User Datagram Protocol (UDP) through Network Address Translators (NATs). Known as STUN, this lightweight protocol that allows applications to:

- Discover the presence and types of both NATs and firewalls between themselves and the public Internet
- Determine the public IP addresses allocated to them by the NAT

SIP endpoints use the STUN protocol to find out the public IP addresses and ports for SIP signaling and RTP media transport. Then they can use the address and port information to create multimedia sessions with other endpoints on the public network.

You can define STUN servers functionality on a per-realm basis, allowing you set up multiple STUN servers.

## About STUN Messaging

STUN messages uses six messages, three of which are used for Binding and three of which are uses for the Shared Secret. While it supports all three Binding messages (request, response, and error), the Oracle Communications Session Border Controller does not support the Shared Secret Request or the message integrity mechanism that relies on the shared secret. When acting as a STUN server, the Oracle Communications Session Border Controller responds to STUN binding requests in accordance with RFC 3489 and the rfc3489bis draft.

STUN messages can contain the following attributes:

Message Type	Attribute Description
MAPPED-ADDRESS	Appears in the Binding Response; contains the source IP address and port from which the Binding Request was sent to the STUN server.
XOR-MAPPED-ADDRESS	Appears in the Binding Response; contains the MAPPED-ADDRESS information encoded in a way the prevents intelligent NAT devices from modifying it as the response goes through the NAT.
SOURCE-ADDRESS	Appears in the Binding Response; contains the IP address and port from which the STUN server sent its response.
CHANGED-ADDRESS	Appears in the Binding Response; contains an alternate STUN server IP address and port, different from the primary STUN server port. The STUN client might use this attribute to perform the NAT tests described in RFC 3489.
CHANGE-REQUEST	Appears in the Binding Request; instructs the STUN server to send its response from a different IP address and/or port. The STUN client might use this attribute to perform the NAT tests described in RFC 3489.
RESPONSE-ADDRESS	Appears in the Binding Request; defines an IP address and port to which the STUN server should send its responses. Appears in the Binding Request;
REFLECTED-FROM	Appears in the Binding Response; reflects the IP address and port from which a Binding Request came. Only included when the Binding Request has used the RESPONSE-ADDRESS attribute.

Message Type	Attribute Description
UNKNOWN-ATTRIBUTES	Appears in the Binding Error; reflects the mandatory attributes in a Binding Request message that the server does not support.
ERROR-CODE	Appears in the Binding Error; indicates an error was detected in the Binding Request, and contains an error code and reason phrase.

To perform NAT discovery, the endpoint (STUN client) sends a Binding Request to the STUN server port (IP address and port) with which it is configured. The STUN server then returns either a;

- Binding Response—Allows the transaction to proceed
- Binding Error—Halts the transaction, and prompts the client to take the action appropriate to the response given in the ERROR-CODE attribute

When the transaction proceeds and the STUN server sends the Binding Response, that response contains the MAPPED-ADDRESS attribute, which contains the IP address and port from which the server received the request. The STUN client then uses the MAPPED-ADDRESS when sending signaling messages.

For example, a SIP endpoint sends Binding Requests from its SIP port to determine the public address it should place in SIP headers, like the Via and Contact, of the SIP requests it sends. When this SIP endpoint prepares to make or answer a call, it sends Binding Requests from its RTP port to find out the public address it should place in SDP included in an INVITE request or response.

### STUN Server Functions on the Oracle Communications Session Border Controller

When the Oracle Communications Session Border Controller receives a STUN message, it first determines its message type. Only STUN Binding Requests are processed, and all other message types are dropped without response.

Then the Oracle Communications Session Border Controller examines the Binding Request's STUN attributes. It returns error responses if it finds any unsupported mandatory attributes. This takes the form of a Binding Error Response, containing the ERROR-CODE attribute with reason 420 (Unknown Attribute) and an UNKNOWN-ATTRIBUTES attribute with a list of the unsupported attributes. If the Oracle Communications Session Border Controller receives a Binding Request with attributes that do not belong in STUN Binding Requests, it returns the Binding Error Response with the ERROR-CODE attribute with reason 400 (Bad Request).

Next the Oracle Communications Session Border Controller determines whether to follow RFC 3489 procedures or rfc3489bis procedures. If the Transaction ID contains the STUN cookie, then the Oracle Communications Session Border Controller follows rfc3489bis procedures; if not, it follows RFC 3489 procedures. Because it defines the procedures for testing the NAT to see what type of NAT it is, RFC 3489 procedures are most complex. Issues with reliability of those results have caused testing procedures and attributes to be deprecated in fc3489bis.

### RFC 3489 Procedures

The Oracle Communications Session Border Controller (the STUN server) constructs the Binding Response and populates it with these attributes:

- MAPPED-ADDRESS and (optionally) XOR-MAPPED-ADDRESS—Containing the source IP address and port from which the server saw the request come
- SOURCE-ADDRESS—Containing the IP address and port from which the server will send the Binding Response
- CHANGED-ADDRESS—Containing the STUN server port that has a different address and different port from the ones on which the server request was received

If the Binding Request contains a RESPONSE-ADDRESS attribute, the server adds the REFLECTED-FROM attribute with the IP address and port from which the server saw the request come. Then the server sends the Binding Response to the IP address and port in the RESPONSE-ADDRESS attribute. If the RESPONSE-ADDRESS attribute's IP address and port are invalid, the server sends a Binding Error Response with an ERROR-CODE attribute reason 400 (Bad Request) to the client.

If the Binding Request contains a CHANGE-REQUEST attribute, the server sends Binding Response from the IP address and port matching the information in the CHANGE-REQUEST. The following variations can occur:

- If the IP address and port flags are set, the server selects the server port with a different IP address and different port.
- If only the IP address flag is set, the server selects the server port with a different IP address but with the same port.
- If only the port flag is set, the server selects the server port with the same IP address but with a different port.

The selected server port appears in the Binding Responses's SOURCE-ADDRESS attribute. When there is no CHANGE-REQUEST attribute, the server uses the server port on which the Binding Request was received.

Finally, the server encodes the outgoing message and sends it to the client at either:

- The destination IP address and port in the REPOSE-ADDRESS attribute, if it was present in the Binding Request.
- The MAPPED-ADDRESS.

### rfc3489bis Procedures

If the Binding Request contains the appropriate cookie in its Transaction ID, the server constructs a Binding Response populated with the XOR-MAPPED-ADDRESS attribute. That attribute will contain the source IP address and port from which the server saw the request come. Then the server encodes and sends the message to the client from the IP address and port on which the request was received. The message is sent to the IP address and port from which the request came.

## Monitoring

- STUN Server Statistics—You can display statistics for the STUN server using the ACLI `show mbcd stun` command when the STUN server has been enabled. However, if the STUN server has not been enabled since the last system reboot, the command does not appear and no statistics will be displayed.
- STUN Protocol Tracing—You can enable STUN protocol tracing two ways: by configuration or on demand.
  - By configuration—The Oracle Communications Session Border Controller's STUN protocol trace file is called `stun.log`, which is classified as a call trace. This means that when the system configuration's call-trace parameter is set to enabled, you will obtain STUN protocol information for the system. As with other call protocol traces, tracing data is controlled by the log-filter in the system configuration.
  - On demand—Using the ACLI `notify mbcd log` or `notify mbcd debug` commands, you enable protocol tracing for STUN. Using `notify mbcd debug` sets the STUN log level to TRACE. You can turn off tracing using the `notify mbcd onlog` or `notify mbcd nodebug` commands. Using `notify mbcd nodebug` returns the STUN log level back to its configured setting.

## STUN Server Configuration

You configured STUN servers on a per-realm basis, one server per realm. To support that various NAT tests it describes, RFC 3489 requires that two different IP addresses and two different UDP port numbers be used for each server. So your STUN server will listen on a total of four STUN server ports. Although newer work does away with this requirement, the Oracle Communications Session Border ControllerC supports it for the purpose of backwards compatibility.

For each realm configuration with an enabled STUN server, untrusted ACL entries will be added to forward all packets received on the four STUN Server Port.

To enable STUN server support for a realm:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type `media-manager` and press Enter.

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```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter. If you are adding this feature to a pre-existing realm configuration, you will need to select and edit your realm.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

4. **stun-enable**—Set this parameter to **enabled** to turn STUN server support for this realm on. This parameter defaults to **disabled**, meaning STUN server support is off.
5. **stun-server-ip**—Enter the IP address for the primary STUN server port. The default for this parameter is 0.0.0.0.
6. **stun-server-port**—Enter the port to use with the **stun-server-ip** for primary STUN server port. The default is 3478.
7. **stun-changed-ip**—Enter the IP address for the CHANGED-ADDRESS attribute in Binding Requests received on the primary STUN server port. This IP address must be different from than the one defined for the **stun-server-ip** parameter. The default for this parameter is 0.0.0.0.
8. **stun-changed-port**—Enter the port combination to define the CHANGED-ADDRESS attribute in Binding Requests received on the primary STUN server port. The default for this parameter is 3479.
9. Save and activate your configuration.

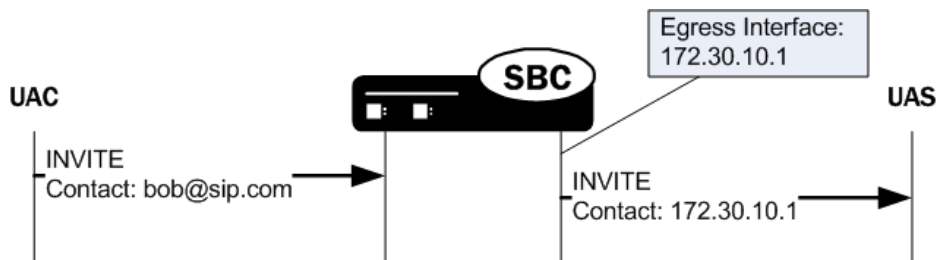
## SIP GRUU

SIP Globally Routable User Agent (UA) URIs (GRUU) are designed to reliably route a SIP message to a specific device or end user. This contrasts with a SIP AoR which can refer to multiple UAs for a single user, thus contributing to routing confusion. The Oracle Communications Session Border Controller can perform different behaviors when it finds SIP GRUUs in Contact headers.

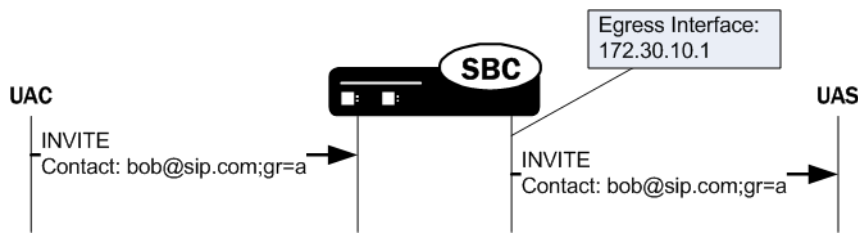
User agents supporting GRUU include a GRUU-identifying parameter in the Contact header of dialog forming and target refresh requests. The Oracle Communications Session Border Controller scans for the GRUU parameter in the Contact header either when the message is received on a realm or interface configured for registered endpoints or when the pass-gruu-contact parameter is enabled. Configure an interface for registered endpoints by enabling nat-traversal or registration caching.

### Contact Header URI Replacement

When no GRUU is encountered in the contact header, and when a SIP message is forwarded to the egress realm, the contact header's URI is replaced with the Oracle Communications Session Border Controller's egress interface. For example:

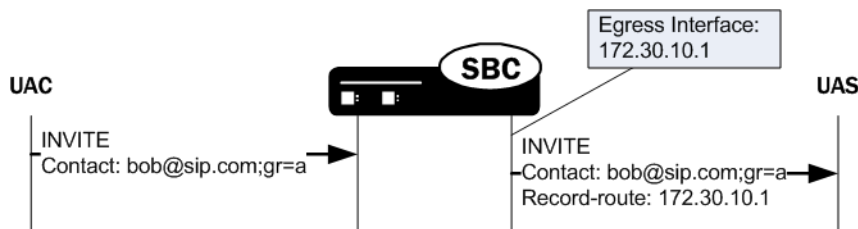


When the Oracle Communications Session Border Controller forwards a request where the original Contact header contains a GRUU, the contact header's URI is forwarded unchanged on the egress side of the call. For example:



## Record-Route Addition

When the request is forwarded to a realm where the endpoint's registrar does not exist, the Oracle Communications Session Border Controller adds a Record-Route header containing the egress SIP interface address. This causes subsequent replies or requests addressed to the GRUU to be routed through the Oracle Communications Session Border Controller first.



When the request is forwarded to the realm where the registrar exists, adding the Record-Route header is unnecessary and does not occur. This is because subsequent requests are directed to the registrar which will ultimately forward them to the Oracle Communications Session Border Controller using the registered Contact in the Request-URI.

## GRUU URI Parameter Name

The Oracle Communications Session Border Controller scans for a gr URI parameter in the contact header to identify it as a GRUU as defined in the ietf draft[2]. The Oracle Communications Session Border Controller can be configured to scan for a gruu URI parameter in the contact header too. This alternate behavior is enabled with the scan-for-ms-gruu option and is used to interact with the Microsoft Office Communications Server unified communications product. When scan-for-ms-gruu is enabled, the Oracle Communications Session Border Controller scans first for the gruu URI parameter. If not found, it then scans for gr URI parameter.

## SIP GRUU Configuration

This section shows you how to configure the GRUU support for non-registered contacts. Enabling GRUU functionality to parse for gr URI parameter rather than the IETF standard gruu parameter is also provided.

To configure SIP GRUU functionality:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

If you are adding this feature to an existing configuration, you need to select the configuration (using the ACLI **select** command) before making your changes.

4. **pass-gruu-contact**—Set this parameter to enabled to parse for gr URI parameter in the contact header in non-registered endpoints' messages and then pass the messages through the system.

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---

5. **options**—Set the options parameter by typing **options**, a Space, the option name **scan-for-ms-gruu**. This option forces the Oracle Communications Session Border Controller to first scan for the gruu URI parameter, then the gr URI parameter.
6. Save and activate your configuration.

## SIP ISUP Features

---

This section describes the Oracle Communications Session Border Controller's features for SIP ISUP.

### SIP Diversion to SIP-ISUP Interworking

For networks in which there are devices that do not support SIP-T or SIP-I (and support native SIP alone), the Oracle Communications Session Border Controller now supports SIP Diversion interworking. This feature enables such devices to function properly in instances that require SIP-T/SIP-I style ISUP IAM message encapsulation in ISUP requests, and to receive any call forwarding information in the IAM according to ISUP standards.

The Oracle Communications Session Border Controller interworks a native SIP INVITE request to SIP-T one by inserting an ISUP IAM body based on the INVITE; this includes redirections information based on the Diversion header. This feature can also perform the reverse translation. That is, it can interwork a SIP INVITE that does have the ISUP IAM body to a non-ISUP INVITE. In this case, the Oracle Communications Session Border Controller generates the necessary Diversion headers based on the IAM's Redirection information.

### SIP-ISUP Configuration

To use this feature, you set up:

- **sip-profile**—Defines the redirection behavior
- **sip-isup-profile**—Defines the ISUP version to use when supporting SIP-T

You can then apply these profiles to realms, session agents, and SIP interfaces where you want this feature enabled.

The **sip-profile** configuration contains information that defines redirection behavior for the configuration where you apply it. You can set the redirection behavior to: **none**, **isup**, or **redirection**. You also uniquely identify the profile so that you can apply it by name in other configurations. This is a multiple-instance configuration, meaning that you can set up as many SIP profiles as needed.

To set up a SIP profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-profile** and press Enter.

```
ACMEPACKET(session-router)# sip-profile
ACMEPACKET(sip-profile)#
```

4. **name**—Enter the name of the SIP profile. You will use this name when you apply this profile to realms, session agents, and SIP interfaces. This parameter is required, and it has no default value.
5. **redirection**—Choose the redirection mode you want to use: **none** (default), **isup**, or **redirection**. The **inherit** value is reserved for future use. Note that when you set this parameter to **isup**, you should configure along with it a SIP ISUP profile; this will avoid any possible incompatibility when support for this feature expands (as expected).
6. Save your work.



## SIP ISUP Profile Configuration

To set up a SIP ISUP profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-isup-profile** and press Enter.

```
ACMEPACKET(session-router)# sip-isup-profile
ACMEPACKET(sip-isup-profile)#
```

4. **name**—Enter the name of the SIP ISUP profile. You will use this name when you apply this profile to realms, session agents, and SIP interfaces. This parameter is required, and it has no default value.
5. **isup-version**—Specify the ISUP version you want used in this profile in order to support SIP-T: **ansi-2000** (default) or **itu-99**.
6. Save your work.

## SIP-ISUP Configuration

When you want to enable this feature for a realm, session agent, or SIP interface, you configure the **sip-profile** and **sip-isup-profile** parameters with the name of the profile you want applied.

The sample here shows this feature being applied to a session agent, but the realm and SIP interface configurations also have the same two parameters you use to set up the feature.

To apply a SIP profile and a SIP ISUP profile to a session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

4. **sip-profile**—Enter the name of the SIP profile, which defines the redirection behavior. This is the value you entered in the **name** parameter of the SIP profile configuration. This parameter has no default value.
5. **sip-isup-profile**—Enter the name of the SIP ISUP profile, which defines the ISUP version to use for SIP Diversion SIP-ISUP interworking. This is the value you entered in the **name** parameter of the SIP ISUP profile configuration. This parameter has no default value.
6. Save your work.

## SIP-ISUP Format Version Interworking

ISUP message can be carried in SIP messages through either a standard body or through a multipart MIME encoded body. While ANSI and ITU are the two major groups, but each contains many specific variants. To facilitate instances where two sides of a call use different versions, the Oracle Communications Session Border Controller supports interworking between the following SIP ISUP formats: ANSI, ITU, ETSI-356 (an ITU variant), and GR-317 (an ANSI variant). To do so, the Oracle Communications Session Border Controller can move, delete, and add parameters to various sections of the message.

### Details

The ISUP message version is determined by one of two things: the content type of the SIP message or the MIME content-type. When the base and version parameters do not match, the Oracle Communications Session Border Controller first uses the base parameter value to determine the format. If there is no base, the Oracle Communications Session Border Controller then checks the version parameter. And if there is neither, the Oracle Communications Session Border Controller uses the **isup-version** configured in the **sip-isup-profile** configuration from the inbound realm, session agent, or SIP interface. Available values for that parameter are **ansi-2000**, **itu-99**, **gr-317**, or **etsi-356**. The Oracle Communications Session Border Controller considers unknown any value for the version that fails to match one of these or is missing.

Messages that contain an unknown ISUP format pass through the Oracle Communications Session Border Controller untouched. If there are operations to be performed on them, however, SIP ISUP HMR will take place. After the body has been converted, the Oracle Communications Session Border Controller updates both the base and version parameters of the content-type.

Custom formats are not supported.

### SIP-ISUP Format Interworking Configuration

This section shows you how to set up a SIP-ISUP format interworking. First, you configure a SIP ISUP profile, and then you apply it to a realm, session agent or SIP interface.

To set up a SIP ISUP profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-isup-profile** and press Enter.

```
ACMEPACKET (session-router) # sip-isup-profile
ACMEPACKET (sip-isup-profile) #
```

4. **name**—Enter the name of the SIP ISUP profile. You will use this name when you apply this profile to realms, session agents, and SIP interfaces. This parameter is required, and it has no default value.
5. **isup-version**—Specify the ISUP version you want to which you want to convert: **ansi-2000**, **itu-99**, **gr-317**, or **etsi-356**.
6. **convert-isup-format**—Set this parameter to **enabled** if you want to perform SIP ISUP format version interworking. The default is **disabled**, meaning that this feature is turned off.
7. Save your work.

### SIP-ISUP Format Interworking Configuration

When you want to enable this feature for a realm, session agent, or SIP interface, you configure the **sip-isup-profile** parameter with the name of the profile you want applied.

The sample here shows this feature being applied to a session agent, but the realm and SIP interface configurations also have the same parameter you use to set up the feature.

To apply a SIP profile and a SIP ISUP profile to a session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

4. **sip-isup-profile**—Enter the name of the SIP ISUP profile, which defines the ISUP version to convert to. This is the value you entered in the **name** parameter of the SIP ISUP profile configuration. This parameter has no default value.
5. Save your work.

## HMR for SIP-ISUP

The Oracle Communications Session Border Controller's HMR functionality can operate on ISDN user party (ISUP) binary bodies. Using the same logic and mechanisms that are applied to SIP header elements, HMR for SIP-ISUP manipulates ISUP parameter fields and ISUP message parts. You can create MIME rules that function in much the same way the SIP header rules do; whereas SIP header rules can change the specific headers of a SIP message, MIME rules can manipulate targeted body parts of a SIP message.

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### Changes and Additions to Equality Operators

The following table defines the additions and changes to HMR equality operators introduced with this feature.

Unlike the Boolean operators the ampersand (&) and the pipe (|), you can use the following equality operators in conjunction with string operators. For example, a header-value with its **comparison-type** set to **boolean**, can have this match-value evaluated: “(\$rule1.\$elem1.\$0 + \$rule1.\$elem2.@1) == \$rule2.\$0”. Equality operators can also be used with Boolean operators, as in this example: (\$rule1.\$0 == \$rule2.\$1) & \$rue3.

Equality operators always evaluate to either true or false.

Equality Operator Symbol	Short Description	Detailed Information
==	String case sensitive quality operator	Performs a character-by-character, case-sensitive string comparison on both the left side and the right side of the operator.
~=	String case insensitive quality operator	Performs a character-by-character, case-insensitive string comparison on both the left side and the right side of the operator.
!=	String case sensitive not equal to equality operator	Performs a character-by-character, case-sensitive string comparison on both the left side and the right side of the operator, returning true if the left side is equal to or less than the right side of the operator.
>=	Greater than or equal to operator	Performs a string-to-integer conversion. If the string-to-integer comparison fails, the value is treated as 0. After the conversion, the operator will compare the two values and return true only if the left side is greater than or equal to the right side of the operator.
<	Less than operator	Performs a string-to-integer conversion. If the string-to-integer conversion fails, the value is treated as 0. After the conversion, the operator will compare the two values and return true only if the left side is less than the right side of the operator.
>	Greater than operator	Performs a string-to-integer conversion. If the string-to-integer conversion fails, the value is treated as 0. After the conversion,

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Equality Operator Symbol	Short Description	Detailed Information
		the operator will compare the two values and return true only if the left side is greater than the right side of the operator.

### Reserved Words

To improve system performance and simplify configuration, the Oracle Communications Session Border Controller now supports pre-defined reserved words for commonly-used URI parameters for HMR.

Reserved words retrieve values directly from the SIP message, without your needing to create rules to store them. Their function is similar to the \$REMOTE\_VIA\_HOST and other already-defined variables. If the header or value does not exist in the SIP message, either an empty string is returned or—for Boolean uses—the value FALSE is returned.

Reserved words apply to these commonly-accessed SIP headers and their prefixes are:

- To—\$TO\_XXX
- From—\$FROM\_XXX
- Contact—\$CONTACT\_XXX
- Request URI—\$RURI\_XXX
- P-Asserted-Identity—\$PAI\_XXX
- P-Preferred-Identity—\$PPI\_XXX
- P-Called-Party-ID—\$PCPID\_XXX

The following table contains the list of supported reserved words and a description of each.

Reserved Word	Description
xxx_USER	The URI name of the header without any user parameters
xxx_PHONE	The URI user of the header as a phone number but without visual separators; may or may not contain a leading plus sign (+)
xxx_HOST	The URI host of the header
xxx_PORT	The URI port of the header; Value set to 5060 even if it is not actually in the message
CALL_ID	Resolves to the Call-ID of the current SIP message; added for convenience, and is a common store rule
TIMESTAMP_UTC	Timestamp is RFC 3339 format: 2009-10-10T22:00:09Z or YYYY-MM-DDTHH:MM:SS:PPPZ. The .PPP refers to partial seconds and is optional; time is based on UTC.

The reserved word CRLF resolves to `\r\n` and is commonly used in MIME manipulation. If you are creating a new body, there might be a need for many CRLFs in the new-value parameter.

All of these operators cause additional overhead to the HMR processing because each operator requires an evaluation of the left and right sides of the expression. To speed up evaluation of new-value expressions, you can now enter escapable characters (`\f`, `\n`, `\r`, `\t`, `\v`) with a backslash (`\`) and the Oracle Communications Session Border Controller will convert them to escaped characters during the compilation of the expression (i.e., CLI configuration time).

### List of Reserved Words

\$ORIGINAL,LOCAL	\$TRUNK_GROUP	\$TARGET_PORT
\$IP,LOCAL_PORT	\$TRUNK_GROUP_CONTEXT	\$MANIP_STRING
\$REMOTE_IP	\$REPLY_IP	\$CRLF


\$REMOTE_PORT	\$REPLY_PORT	\$TO_USER
\$REMOTE_VIA_HOST	\$TARGET_IP	\$TO_PHONE
\$TO_HOST	\$TO_PORT	\$FROM_USER
\$FROM_PHONE	\$FROM_HOST	\$FROM_PORT
\$CONTACT_USER	\$CONTACT_PHONE	\$CONTACT_HOST
\$CONTACT_PORT	\$RURI_USER	\$RURI_PHONE
\$RURI_HOST	\$RURI_PORT	\$PAI_USER
\$PAI_PHONE	\$PAI_HOST	\$PAI_PORT
\$PPI_USER	\$PPI_PHONE	\$PPI_HOST
\$PPI_PORT	\$PCPID_USER	\$PCPID_PHONE
\$PCPID_HOST	\$PCPID_PORT	\$CALL_ID
\$TIMESTAMP_UTC	\$MANIP_PATTERN	\$T_GROUP
\$T_CONTEXT,	\$M_STRING	---

### Changes to Action

In releases prior to S-C6.2.0, the **sip-manip** action is only supported in the header rule (**header-rule**) configuration. This limitation has been deemed unnecessary, and so you can now set the action parameter to **sip-manip** at all levels of HMR configuration, including element rules.

### About MIME Rules

MIME rules (set up in the ACLI **mime-rules** configuration) operate much the same way that SIP header rules do. You can set parameters in the MIME rules that the Oracle Communications Session Border Controller uses to match against specific SIP methods and message types. The system compares the search criteria against the body or body parts using the type of comparison you choose. Offering a variety of selection, you can pick kind of manipulation that suits your needs; the Oracle Communications Session Border Controller then takes action with matching and new values to change the SIP message.

 **Note:** when you use the **delete** action on a multi-part MIME string that reduces a number of bodies down to one, the SIP message remains a multi-part MIME message with only one body part (and thereby avoids the header conflicting with the message itself).

You identify the MIME rule by configuring a content type that refers to the specific body part on which to operate. For example, given a SIP Content-Type header with the value `multipart/mixed;boundary=unique-boundary-1`, you would enter a content-type value of **application/sdp** to manipulate specifically on the SDP portion of the SIP message. The Oracle Communications Session Border Controller knows automatically if it is operating on SIP messages with single or multiple body parts, and the content-type setting applies to both kinds. And when making its comparison, the Oracle Communications Session Border Controller matches the content-type of the body with regard to case (case insensitive), ignoring any header parameters.

Both for making comparisons against the body part and for new/replacement values, the Oracle Communications Session Border Controller treats the match and new values you set for a MIME rule as ASCII strings. Therefore, a mime rule operating on a binary body part will yield an improper conversion of a new value with respect to the binary body part.

Within MIME rules, you configure MIME headers, which operate on the specific headers in the match body part of the SIP message. The Oracle Communications Session Border Controller uses the MIME header name to run a string comparison to match the specific header in the message's body part.

Using these rules, you can also manipulate the preamble—or the SIP message text that follows the headers but precedes the body separator. To do so, enter the keyword **@preamble** for the content type parameter in the MIME

rule. Likewise you can manipulate the epilogue—or the text that follows the last body part after the last separator—using the keyword `@epilogue`.

Note that the ACLI limits character entries to 255 characters before the return character must be entered, but MIME parts can easily exceed this 255-character size. So you might need to enter a value larger than 255 characters. To do so, you start your entry (in the match-value or new-value parameters) with a plus sign (+). The plus sign instructs the system to add the string after it to the pre-existing match or new value. For the new-value parameter, the Oracle Communications Session Border Controller checks the value immediately for validity. Be sure that when you are appending values to a new-value that the entire expression is valid at each point where strings are appended.

### MIME Rules Configuration

This section shows you how to configure MIME rules and MIME headers.

To configure MIME rules:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type `session-router` and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type `sip-manipulation` and press Enter. **If you are adding this feature to an existing configuration, then remember you must select the configuration you want to edit.**

```
ACMEPACKET (session-router) # sip-manipulation
ACMEPACKET (sip-manipulation) #
```

4. **Type `mime-rules` and press Enter.**

```
ACMEPACKET (sip-manipulation) # mime-rules
ACMEPACKET (sip-mime-rules) #
```

5. **name**—Enter a name for this MIME rule. This parameter is required and has no default.
6. **content-type**—Enter the content type for this MIME rule. This value refers to the specific body part in the SIP message body that is to be manipulated. For example, given a SIP Content-Type header with the value `multipart/mixed;boundary=unique-boundary-1`, you would enter a content-type value of `application/sdp` to manipulate specifically on the SDP portion of the SIP message.

To manipulate the SIP preamble or epilogue, enter the keyword `@preamble` or keyword `@epilogue`.

7. **action**—Choose the type of action you want to be performed: `none`, `add`, `delete`, `manipulate`, `store`, `sip-manip`, and `find-replace-all`. These are the same actions you can select when configuring SIP header manipulation. The default is `none`.
8. **comparison-type**—Enter the way that you want body part of the SIP message to be compared. This choice dictates how the Oracle Communications Session Border Controller processes the match rules against the SIP header. The default is `case-sensitive`. The valid values are: `case-sensitive`, `case-insensitive`, `boolean`, `refer-case-sensitive`, `refer-case-insensitive`, and `pattern-rule`.
9. **msg-type**—Enter the SIP message type on which you want the MIME rules to be performed. Valid values are `any`, `request`, and `reply`. The default value is `any`.
10. **methods**—Enter the list of SIP methods to which the MIME rules applies. There is no default for this parameter.
11. **match-value**—Enter the value to match against the body part in the SIP message. This is where you can enter values to match using regular expression values. Your entries can contain Boolean operators.
12. **new-value**—When the action parameter is set to `add` or to `manipulate`, enter the new value that you want to substitute.

To configure MIME headers for performing HMR operations on specific headers in the matched body part of the SIP message:

13. Follows Steps 1 through 4 above.

14. **Type mime-header-rules and press Enter.**

```
ACMEPACKET (sip-mime-rules) # mime-header-rules
ACMEPACKET (sip-mime-header-rules) #
```

15. **name**—Enter a name for this MIME header rule. This parameter is required and has no default.

16. **mime-header**—Enter the value to be used for comparison with the specific header in the body part of the SIP message. There is no default for this parameter.

17. **action**—Choose the type of action you want to be performed: **none**, **add**, **delete**, **manipulate**, **store**, **sip-manip**, and **find-replace-all**. The default is **none**.

18. **comparison-type**—Enter the way that you want the header in the body part of the SIP message to be compared. This choice dictates how the Oracle Communications Session Border Controller processes the match rules against the SIP header. The default is **case-sensitive**. The valid values are: **case-sensitive**, **case-insensitive**, **boolean**, **refer-case-sensitive**, **refer-case-insensitive**, and **pattern-rule**.

19. **match-value**—Enter the value to match against the header in the body part of the SIP message. This is where you can enter values to match using regular expression values. Your entries can contain Boolean operators.

20. **new-value**—When the action parameter is set to **add** or to **manipulate**, enter the new value that you want to substitute.

21. Save your work.

## About MIME ISUP Manipulation

MIME ISUP manipulation supports performing HMR operations on SIP ISUP binary bodies, and is configured in the **mime-isup-rule** configuration. This configuration works the same way that the MIME rule configuration does and contains the same parameters for you to set, but it also includes additional parameters and a sub-configuration targeted specifically for ISUP application.

### Oracle Communications Session Border Controller MIME ISUP Parameters

- **isup-msg-type**—Refers to specific ISUP message types (such as IAM and ACM). The Oracle Communications Session Border Controller uses with the **msg-type** parameter (which identifies the SIP message) in the matching process. You enter values in this parameters as a list of numbers rather than as an enumerated value because of the large number of ISUP message type, and the range is between 0 and 255.
- **isup-spec**—Specifies how the Oracle Communications Session Border Controller is to parse the binary body; valid values are the enumerated type. The values for this parameter are these SIP ISUP formats
  - ANSI-2000—Corresponding to ANSI T1.113-2000
  - ITU-99—Corresponding to ITU Q.763

Because ISUP messages do not identify their format, you must designate which you want to use.
- **isup-parameter-rules** (sub-configuration)—If you are familiar with HMR, then think of this parameter as being similar to the **element-rule** for a SIP header rule. You use it to create, manipulate, and store different parameters in the body of the ISUP message. Two parameters for this rule are unique: **parameter-rule** and **parameter-format**.
  - **parameter-rule**—Using ISUP parameter mapping, this setting identifies which of the ISUP parameters on which your want to perform manipulation. This parameter takes values between 0 and 255, and you must know the correct ISUP mapping value for your entry. The Oracle Communications Session Border Controller calculates the offset and location of this parameter in the body. Note that the value returned from the body does not the type or length, only the parameter value. For example, a **parameter-type** value of 4 acts on the Called Party Number parameter value.

In accordance with the ISUP specifications, only certain message types are allowed to have optional parameters. And if optional parameters are present, an offset field must exist for them; so its value is 0 even if there are no optional parameters in the SIP message. For example, if you define a SIP ISUP rule that applies to

all message types and adds a parameter that is neither fixed nor variable, The system adds it as an optional parameter regardless of whether that message type should not support optional parameters.

If you define an ISUP parameter rule with an **add** action and an empty **new-value**, the Oracle Communications Session Border Controller uses the default for that parameter. If you define an ISUP parameter rule with a **replace** action and no parameters exist, the Oracle Communications Session Border Controller will not perform any action. This behavior is consistent with that of SIP header rules in that a value can only be replaced if it already exists. If there is a value and no new value is set, the Oracle Communications Session Border Controller set it as a zero-length parameter.

- **parameter-format**—This parameter converts the specific parameter to a string representation of that value. Valid values for **parameter-format** are: **number-param**, **hex-ascii** (default), **binary-ascii**, **ascii-string**, and **bcd**.

Both match and new values are encoded and decoded by the designated parameter-format type. In this regard, the **match-value** decodes the parameters and the **new-value** encodes the ASCII string into the respective binary format.

Note if you enter a new-value setting larger than the size of the parameter, the Oracle Communications Session Border Controller will perform no operation and will generate a corresponding error log message.

The following table provides information about the values you can enter:

parameter-format Setting	Description
hex-ascii	Default. Converts the entire binary body. Non-hexadecimal characters fail in matching against the body part if they are in the match-value setting and non-hexadecimal characters places in a new-value setting result in no operation being formed.
binary-ascii	Converts each hexadecimal value to its corresponding string, binary representation. For example, the Oracle Communications Session Border Controller would convert the ISUP parameters with a hexadecimal binary value of 8A to 10001010. Non-binary digit characters fail when matching against the body part if they are contained in the match-value setting and non-binary characters in the new-value setting results in no operation being performed.
ascii-string	Treats the binary parameter as true ASCII in raw format. The Oracle Communications Session Border Controller supports only the printable range of ASCII characters. If a value in the ISUP parameters cannot be decoded to ASCII, the Oracle Communications Session Border Controller returns an empty string. Non-printable or meta characters cannot be entered as new-value settings, so this presents no issues for encoding.
bcd	In ISUP speak, BCD refers to the binary forma of the number used as a half a byte nibble, with the byte's lower nibble containing the first digit and the higher containing the second digit. For example, the number 127 is encoded as the two binary bytes 0x2107 on the wire. Using this mode, the Oracle Communications Session Border Controller treats the binary ISUP content as BCD; it should decode it from 0x2107 to the string 1270, and from a string of 127 it should decode it as 0x2107. Since a byte has two nibbles, a nibble might have to be added. And when the Oracle Communications Session Border Controller performs decoding, it cannot know that a BCD byte represents one or two ASCII digits—so it assumes there are two. The number-param setting decodes the parameter as a common number parameter. The



parameter-format Setting	Description
	<p>Oracle Communications Session Border Controller sees the odd/even bit as in the first bite as telling it how many nibbles to decode correctly, and it will set the odd-even when it decodes.</p> <p>Non-binary digit characters fail to match against the body part if they are contained in the match value, and non-binary characters in the new value results in no operation being performed.</p>
number-param	<p>As the decimal value of the specified number type, treats the parameters as a generic number parameter type. For example, a parameter-type 4 acts on the Called Party Number parameter.</p> <p>When the action type is replace or add, the Oracle Communications Session Border Controller automatically sets the parameter's odd-even bit based on the number being inserted in relation to the new-value setting. If the Numbering Plan Indicator bits are 0b001 (ISDN, E.164), then the Oracle Communications Session Border Controller sets the Nature of Address field to 0b0000100 (international). If this number type is added to a non-existent parameter field, then the Numbering Plan Indicator field is 0b0000011 (national number). If this number type is added to a non-existent parameters field, then the Numbering Plan Indicator field will be set to 0b001 (ISDN,E.164) and the Oracle Communications Session Border Controller will also follow the previous rules.</p> <p>Regardless of the action type you set, the string represented for match-value use for this type will be the numbers of the address fields after the BCD coding. There will be a leading plus sign (+) if the Number Plan is 0b001 and the Nature of Address is 0b0000100 ((international); otherwise, there will not be a plus sign (+).</p> <p>If it cannot convert the data field to a number parameter, the Oracle Communications Session Border Controller will return an empty string. And if the new-value is not in digit form or cannot fit in the specified parameter type field, the Oracle Communications Session Border Controller takes no action.</p>

### Adding an ISUP Body to a SIP Message

Unlike the MIME manipulation you can use by setting the SIP header rules accordingly, you can add MIME parts to SIP messages using the MIME rules configuration.

You can configure a SIP header manipulation to add an ISUP body to a SIP message. and the Oracle Communications Session Border Controller adds them after any SDP parts if they are present. You can add an ISUP body to a SIP message in two ways:

- You can create a **mime-isup-rule** with the **action** type set to **add**, and enter the entire body in string hexadecimal form in the **new-value** parameter.
- You can leave the **new-value** parameter empty at the **mime-isup-rule** level and create an add rule for an **isup-param-rule**.

In this case, the Oracle Communications Session Border Controller creates the corresponding ISUP message based on the **isup-msg-type** value and supply all of the parameters with their default values. Since the **isup-msg-type** takes a list of values as a valid entry, for this case it only uses the first one. However, the Oracle Communications Session Border Controller ignores the **isup-msg-type** value if you set the **new-value** parameter. And the **isup-param-rule**, if configured, overwrite the default value or add a new parameter based on the defined parameter type.

It is also possible that you might supply a **new-value** both at the **mime-isup-rule** level and at the **isup-param-rule** level. If you do, the **new-value** entry from the **mime-isup-rule** is parsed into an ISUP object and the **isup-param-rule** operates on that object.

### MIME ISUP Manipulation Configuration

This section shows you how to configure MIME ISUP manipulation.

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-manipulation** and press Enter. **If you are adding this feature to an existing configuration, then remember you must select the configuration you want to edit.**

```
ACMEPACKET(session-router)# sip-manipulation
ACMEPACKET(sip-manipulation)#
```

4. Type **mime-isup-rules** and press Enter.

```
ACMEPACKET(sip-manipulation)# mime-isup-rules
ACMEPACKET(sip-mime-isup-rules)#
```

5. **name**—Enter a name for this MIME ISUP rule. This parameter is required and has no default.
6. **content-type**—Enter the content type for this MIME rule. This value refers to the specific body part in the SIP message body that is to be manipulated. For example, given a SIP Content-Type header with the value `multipart/mixed;boundary=unique-boundary-1`, you would enter a content-type value of **application/sdp** to manipulate specifically on the SDP portion of the SIP message.

To manipulate the SIP preamble or epilogue, enter the keyword **@preamble** or keyword **@epilogue**.

7. **action**—Choose the type of action you want to be performed: **none**, **add**, **delete**, **manipulate**, **store**, **sip-manip**, and **find-replace-all**. These are the same actions you can select when configuring SIP header manipulation. The default is **none**.
8. **comparison-type**—Enter the way that you want body part of the SIP message to be compared. This choice dictates how the Oracle Communications Session Border Controller processes the match rules against the SIP header. the default is **case-sensitive**. The valid values are: **case-sensitive**, **case-insensitive**, **boolean**, **refer-case-sensitive**, **refer-case-insensitive**, and **pattern-rule**.
9. **msg-type**—Enter the SIP message type on which you want the MIME rules to be performed. Valid values are **any**, **request**, and **reply**. The default value is **any**.
10. **methods**—Enter the list of SIP methods to which the MIME rules applies. There is no default for this parameter.
11. **match-value**—Enter the value to match against the body part in the SIP message. This is where you can enter values to match using regular expression values. Your entries can contain Boolean operators.
12. **new-value**—When the action parameter is set to **add** or to **manipulate**, enter the new value that you want to substitute.
13. **isup-spec**—Specify how the Oracle Communications Session Border Controller is to parse the binary body; valid values are the enumerated type. The values for this parameter are these SIP ISUP formats:
  - **ANSI-2000** (default)—Corresponding to ANSI T1.113-2000
  - **ITU-99**—Corresponding to ITU Q.763
14. **isup-msg-type**—Identify the specific ISUP message types (such as IAM and ACM) on which to operate. The Oracle Communications Session Border Controller uses with the **msg-type** parameter (which identifies the SIP message) in the matching process. You enter values in this parameters as a list of numbers rather than as an

enumerated value because of the large number of ISUP message type, and the range is between 0 and 255. There is no default for this parameter.

15. **mime-header**—Enter the value to be used for comparison with the specific header in the body part of the SIP message. There is no default for this parameter.

To configure ISUP parameters rules:

16. Follows Steps 1 through 4 above.

17. **Type isup-parameter-rules and press Enter.**

```
ACMEPACKET (sip-mime-isup-rules) # isup-param-rules
ACMEPACKET (sip-isup-param-rules) #
```

18. **name**—Enter a name for this ISUP parameter rule. This parameter is required and has no default.
19. **mime-header**—Enter the value to be used for comparison with the specific header in the body part of the SIP message. There is no default for this parameter.
20. **action**—Choose the type of action you want to be performed: **none**, **add**, **delete**, **manipulate**, **store**, **sip-manip**, and **find-replace-all**. The default is **none**.
21. **comparison-type**—Enter the way that you want the header in the body part of the SIP message to be compared. This choice dictates how the Oracle Communications Session Border Controller processes the match rules against the SIP header. the default is **case-sensitive**. The valid values are: **case-sensitive**, **case-insensitive**, **boolean**, **refer-case-sensitive**, **refer-case-insensitive**, and **pattern-rule**.
22. **match-value**—Enter the value to match against the header in the body part of the SIP message. This is where you can enter values to match using regular expression values. Your entries can contain Boolean operators.
23. **new-value**—When the action parameter is set to **add** or to **manipulate**, enter the new value that you want to substitute.
24. **parameter-type**—Using ISUP parameter mapping, enter which of the ISUP parameters on which your want to perform manipulation. This parameter takes values between 0 and 255, and you must know the correct ISUP mapping value for your entry. The Oracle Communications Session Border Controller calculates the offset and location of this parameter in the body. Note that the value returned from the body does not the type or length, only the parameter value. For example, a parameter-type value of 4 acts on the Called Party Number parameter value.
25. **parameter-format**—Enter how you want to convert specific parameter to a string representation of that value. Valid values for **parameter-format** are: **number-param**, **hex-ascii** (default), **binary-ascii**, **ascii-string**, and **bcd**. Both match and new values are encoded and decoded by the designated parameter-format type. In this regard, the **match-value** decodes the parameters and the **new-value** encodes the ASCII string into the respective binary format.
26. Save your work.

## Configuration Example

This section provides an example of a SIP manipulation configuration that shows MIME rules and MIME ISUP rules.

```

sip-manipulation
  name                               manip
  description
  header-rule
    name                             headerRule1
    header-name                       Date
    action                            add
    comparison-type                   case-sensitive
    msg-type                           reply
    methods
    match-value
    new-value
    element-rule
      name                             elemRule1
      parameter-name

```

```

                                type                header-value
                                action              add
                                match-val-type      any
                                comparison-type     case-sensitive
                                match-value
                                new-value          "August 19, 1967"
mime-rule
    name                          mimeType1
    Content-Type                   application/SDP
    action                          manipulate
    comparison-type                case-sensitive
    msg-type                       request
    methods
    match-value
    new-value
    mime-header
        name                        mimeType1
        mime-header-name           Content-Disposition
        action                      add
comparison-type
    comparison-type                case-sensitive
    match-value
    new-value                       "signal;"
handling=required"
    mime-isup-rule
        name                        mimeType1
        content-type                application/ISUP
        action                      manipulate
        comparison-type            case-sensitive
        msg-type                   request
        methods                    INVITE
        match-value
        new-value
        isup-spec                  {ansi00, itu-92}
        isup-msg-type              0 (0-256 IAM, ACM, etc.)
        mime-header
            name                    mimeType1
            mime-header-name        Content-Disposition
            action                  add
            comparison-type        case-sensitive
            match-value
            new-value               "signal;"
handling=optional"
    isup-param-rule
        name                        isupRule1
        parameter-type              # {0-256 specific type}
        parameter-format            {number-parameter,
hex, binary, ascii, bcd}
        action                      add
        comparison-type            case-sensitive
        match-value
        new-value                   "signal;"
handling=optional"

```

## SIP Session Timer Feature

SIP does not have a keepalive mechanism for established sessions and it does not have the capability of determining whether a session is still active. User agents (UAs) may be able to determine whether a session has timed out by using session specific mechanisms, but proxies cannot always determine when sessions are still active.

The Oracle Communications Session Border Controller provides a SIP session timer feature that, when enabled, forwards the re-INVITE or UPDATE requests from a User Agent Client (UAC) to a User Agent Server (UAS) in order to determine whether or not a session is still active. This refresh feature works for both UAs and proxies. The

following paragraphs provide additional information about the session timers on the Oracle Communications Session Border Controller.

## How the Session Timer Feature Works

During an active SIP call session, when a UA fails to send a BYE message at the end of the session, or when the BYE message gets lost due to network problems, the proxy cannot determine when the session has ended. Therefore, the proxy may hold onto resources associated with the call session for indefinite periods of time causing the session to never time out.

The SIP session timer feature adds the capability to periodically refresh SIP sessions by sending repeated INVITE (re-INVITE) or UPDATE Session Refresh Requests. These requests are sent during active call legs to allow UAs or proxies to determine the status of a SIP session. The Session Refresh Requests along with the session timers determine if the active sessions stay active and completed sessions are terminated.

When you enable the session timer feature on the Oracle Communications Session Border Controller, it periodically sends out a Session Refresh Request (re-INVITE or UPDATE). The Response that is returned to the Oracle Communications Session Border Controller contains a success status code (2xx) that contains a session timer interval. The Oracle Communications Session Border Controller then refreshes the session timer each time it receives the 2xx Response containing that session timer interval.

The initial INVITE message sent from the UAC to the UAS contains two fields that make up the session timer interval in the SIP Session Header:

- Session-Expires (SE) - Specifies the maximum amount of time, in seconds, that can occur between session refresh requests in a dialog before the session is considered timed out.
- Minimum-SE (Min-SE) - Specifies the minimum allowed value, in seconds, for the session expiration.

The following displays the session timer interval values inserted in the SIP session INVITE message per RFC 4028:

```
INVITE sip:9109621001@192.168.200.99 SIP/2.0
Via: SIP/2.0/UDP 192.168.200.49:5060;branch=z9hG4bK0g6t23200gd0res41580.1
Max-Forwards: 69
From: <sip:rick@192.168.1.48>;tag=SDr08od01-188c3fbc-b01a-4d68-b741-09e5dc98a064
To: sip:149@192.168.1.49
Contact: <sip:rick@192.168.200.49:5060;transport=udp>
Call-ID: SDr08od01-9c12b48e3b0f7fad39ff3a2e0ced5ed3-v3000i1
CSeq: 3941 INVITE
Allow: INVITE, ACK, BYE, CANCEL, UPDATE, PRACK
Supported: timer
Session-Expires: 1800
Min-SE: 90
Content-Type: application/sdp
Content-Length: 236

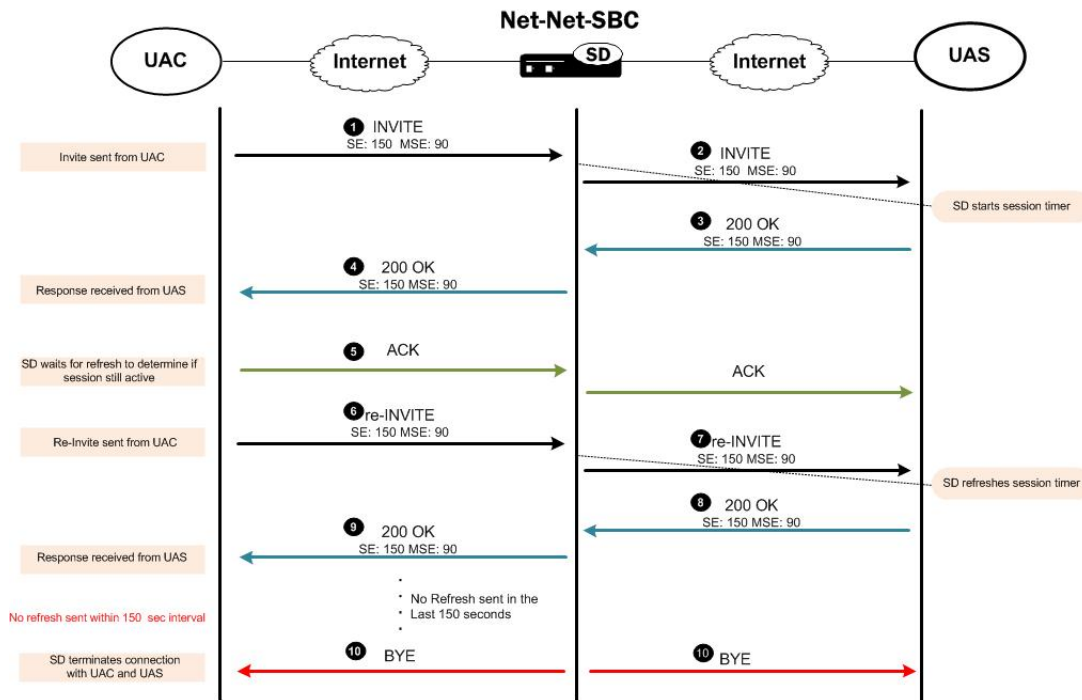
v=0
o=- 3462189550 3462189550 IN IP4 192.168.200.49
s=pjmedia
c=IN IP4 192.168.200.49
t=0 0
m=audio 20000 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

If the Oracle Communications Session Border Controller receives an INVITE from the UAC with a Session-Expires header, it starts a new session timer, or refreshes an existing session timer and then forwards the INVITE to the UAS. The subsequent 2xx Responses and re-INVITES also include the session timer intervals. If the Oracle Communications Session Border Controller does not receive a session refresh within the time specified in the session

timer interval, the session timer expires, and the Oracle Communications Session Border Controller terminates the session between the UAC and the UAS.

The following occurs when you enable the session timer feature on the Oracle Communications Session Border Controller:

1. The UAC sends an INVITE to the Oracle Communications Session Border Controller with the SE and min-SE values (session timer interval). The Oracle Communications Session Border Controller starts a session timer.
2. The Oracle Communications Session Border Controller forwards the INVITE to the User Agent Server (UAS) with the same values.
3. The UAS sends the 200 OK Response to the Oracle Communications Session Border Controller with the session interval values.
4. The Oracle Communications Session Border Controller forwards the Response to the UAC.
5. The UAC sends an ACK (Acknowledge) to the Oracle Communications Session Border Controller, and the Oracle Communications Session Border Controller forwards the ACK to the UAS.
6. The UAC sends out a re-INVITE (Session Refresh Request) to the Oracle Communications Session Border Controller with the session interval values. The Oracle Communications Session Border Controller refreshes the session timer.
7. The Oracle Communications Session Border Controller forwards the re-INVITE to the UAS.
8. The UAS sends the 200 OK response to the Oracle Communications Session Border Controller with the session interval values.
9. The Oracle Communications Session Border Controller sends the 200 OK response to the UAC.
10. If the Oracle Communications Session Border Controller does not receive a Response within the session interval (150 seconds in the following illustration), the timer expires, and the Oracle Communications Session Border Controller terminates the session between the UAC and the UAS. The following illustration shows an example of a dialog between the UAC, the Oracle Communications Session Border Controller, and the UAS during an active session.



When the Oracle Communications Session Border Controller terminates a session it sends a BYE to both the ingress and egress call legs. If accounting is configured, the Oracle Communications Session Border Controller also sends a RADIUS stop record with Acct-Terminate-Cause = Session-Timeout. You can enable or disable the use of the session timers using the ACLI interface at `session-router > sip-conf<sup>g</sup> > options`.

## SIP Session Timer Configuration

You can configure the session timer feature on the Oracle Communications Session Border Controller to periodically refresh SIP sessions and determine whether or not a session is still active. If the Oracle Communications Session Border Controller determines that a session is no longer active, it terminates the session based on the session timer interval settings.

To configure the session timer feature on the Oracle Communications Session Border Controller:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-config** and press Enter to access the SIP config-related configurations. The system prompt changes to let you know that you can begin configuring individual parameters for this object.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

4. Enter options followed by the following value:

- +session-timer-support

```
ACMEPACKET(sip-config)# options +session-timer-support
```

This value enables the system to start the session timer for session refreshes coming from the UAC. The system determines whether or not a session is active based on session refreshes or responses. It terminates the session when no session refreshes occur within the session timer interval. To disable the session timer feature, enter options followed by **-session-timer-support**.

```
Oracle Communications Session Border Controller(sip-config)# options -
session-timer-support
```

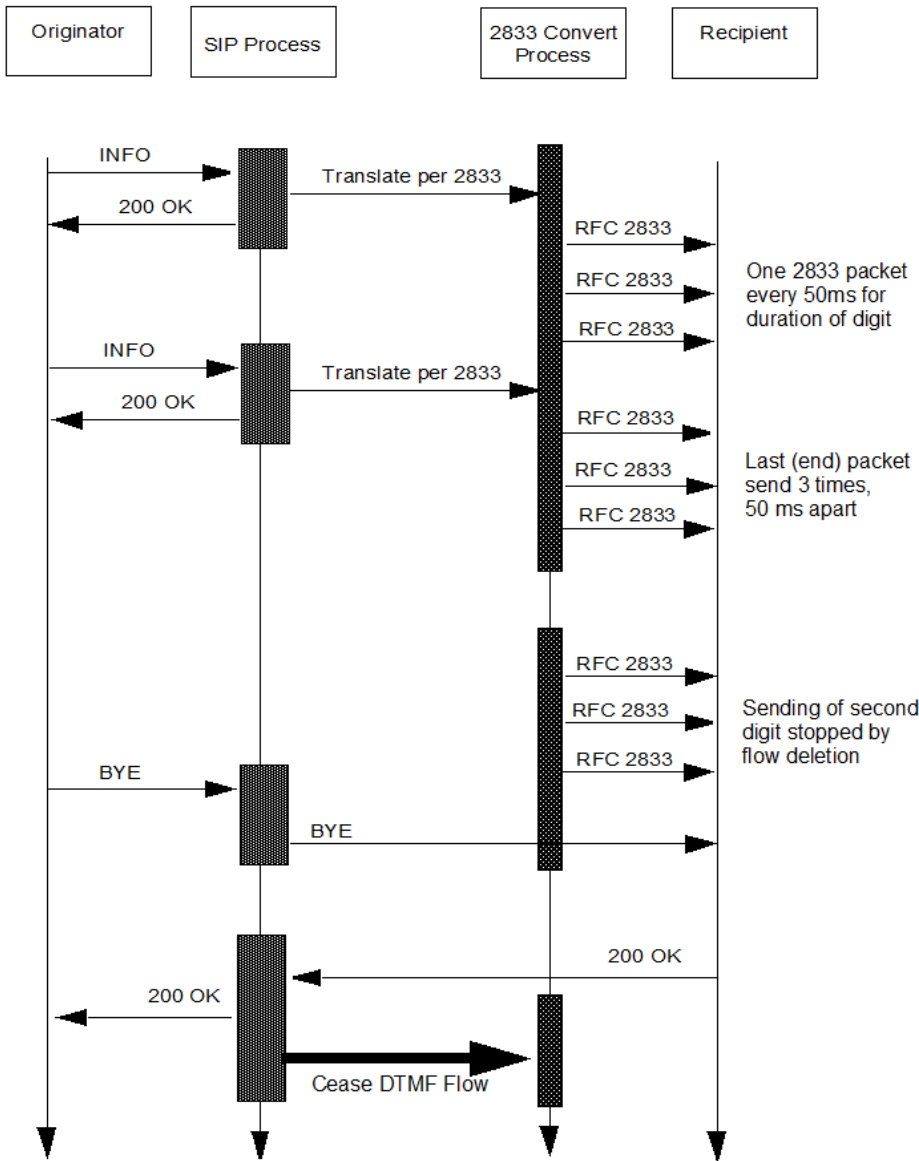


**Note:** To disable session timers, you must add a minus sign (-) before the session-timers-support value.

## DTMF Conversion Processing

Release S-CX6.3F1 provides a configurable SIP option to implement DTMF-to-RFC2833 tone translation. The current, default implementation, which performs well in most network topologies is shown below.

## SIP Signaling Services



When the Oracle Communications Session Border Controller receives an INFO DTMF request, the SIP process determines whether or not it needs to perform DTMF-to-RFC2833 translation. If translation is required, the process forwards the DTMF to the 2833 convert process for translation and transmission to the recipient. Immediately after off-loading the DTMF, the SIP process sends a 200 OK response for the INFO. As shown in the figure, the 2833 convert process generates a number of RFC2833 packets to represent received DTMF digits.

Specifically, the 2833 convert process generates one RFC 2833 packet every 50 milliseconds for the duration of the DTMF digit, whose length is specified in the INFO request, and two retransmits of the last packet (known as the end packet) 50 milliseconds apart.

Consequently, the time interval between the 200 OK and the actual transmission of the RFC 2833 translation is the sum of the DTMF duration and 100 ms.

**Note:** This time interval can be shortened to 100 ms by enabling the `rfc2833-end-pkts-only-for-non-sig` parameter in `media-manger` which results in SD only generating the last packet and its two retransmits.

The early 200 OK allows the endpoint to send the next DTMF digit before the SD has sent all the RFC2833 packets, resulting in the next digit being queued internally by the 2833 convert process before being sent.



A problem arises if the SIP process receives a BYE request from the DTMF originator while queued digits are awaiting translation and transmission. In such an event, the SIP process immediately forwards the BYE request to the recipient, ending the session with DTMF digits awaiting translation and transmission.

An alternative DTMF conversion model provides for a feedback mechanism from the 2833 convert process to the SIP process. With this model enabled, the SIP process buffers a received BYE until it obtains confirmation that all queued DTMF digits have been translated and transmitted. Only after obtaining confirmation, does it forward the BYE to terminate the session.

This processing model is enable by a SIP option, **sync-bye-and-2833**, and requires that **rfc2833-mode** parameter on the egress interfaces is NOT set to dual , any value other than dual , is supported.

1. From superuser mode, use the following command sequence to access sip-config configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

2. Use the SIP option **sync-bye-and-2833** to delay BYE processing until DTMF-to-RFC2833 translation has been completed.

```
ACMEPACKET(sip-config)# options +sync-bye-and-2833="enabled"
ACMEPACKET(sip-config)#
```

3. Use the **done** and **exit** commands to complete configuration.

```
ACMEPACKET(sip-config)# done
ACMEPACKET(sip-config)# exit
ACMEPACKET(session-router)#
```

## SIP-KPML to RFC 2833 Conversion for DTMF Events

The Oracle Communications Session Border Controller supports SIP KPML to RFC 2833 conversion for DTMF events.

### SIP KPML to RFC 2833 Negotiation

The Oracle Communications Session Border Controller removes the Allow-Event header if it contains the value KPML.

A SUBSCRIBE request is sent for each KPML event upon a successful invite.

After the SUBSCRIBE even is successful, the Oracle Communications Session Border Controller will process all NOTIFY requests and respond with 200 OK messages.

The SDP-Insertion feature will be used if the Access-side does not offer a SDP in the invite and the core side does not accept an offer-less invite.

### RFC 2833 to SIP KPML Negotiation

The Oracle Communications Session Border Controller adds an Allow-Event header with the value KPML to the INVITE message.

If a SUBSCRIBE message is received from the Access side, the Oracle Communications Session Border Controller will respond with a 200 OK message.

A KPML NOTIFY request is sent for each 2833 DTMF entered, until the subscription is terminated.

### KPML-2833 Interworking Configuration

To configure KPML - 2833 interworking:

1. In Superuser mode, type **configure terminal** and press Enter.

## SIP Signaling Services

---

```
ACMEPACKET# configure terminal
```

2. Type and **session-router** press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-interface  
ACMEPACKET (sip-interface) #
```

4. **kpml-interworking**—Set this parameter to enabled to use KPML-2833 interworking.
5. Save and activate your configuration.

## LMSD SIP Call Progress Tone Interworking

---

The Oracle Communications Session Border Controller supports Legacy Mobile Station Domain interworking that allows SIP interworking with User Agents that support the 3GPP2 LMSD. The LMSD provides support for existing Mobile Stations in a network that supports IP bearers.

LMSD uses Alert-Info headers in a 180 Ringing response to an INVITE to indicate to a User Agent specific tones to generate locally by the UAC. Most User Agents that do not support LMSD will ignore the SDP in the Alert-Info and will not play ringback locally. The `lmsd-interworking` option allows for the system to suppress SDP in 180 Ringing, 486 Busy Here, and 503 Service Unavailable responses, so that the UAC plays local ringback.

There are no additional license requirements.

### LMSD Interworking Configuration

You can apply `lmsd-interworking` to the sip-interface facing the LMSD User Agents. For Example, if the endpoints supporting LMSD are located in the core realm, then the `lmsd-interworking` option would be added to the core `sip-interface`.

To enable the LMSD Interworking option:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal  
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **sip-interface** and press Enter. If you are adding this feature to a pre-existing configuration, you will need to select and edit it.

```
ACMEPACKET (session-router) # sip-interface  
ACMEPACKET (sip-interface) #
```

4. **options**—Set the options parameter by typing options, a Space, the option name `lmsd-interworking` with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (sip-interface) # options +lmsd-interworking
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the realm configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save your work.

## SIP re-INVITE Suppression

Some of the Interactive Voice Response (IVR) systems that support SIP frequently change the media transport address (IP address and/or port number) when switching between voice menus and/or prompts by sending a re-INVITE.

Often, no other parameters or properties of the session are changed in these re-INVITEs. The frequent re-INVITEs can create performance and capacity problems in other systems along the path of the IVR system and the caller's User Agent.

You can configure the **suppress-reinvite** option on your Oracle Communications Session Border Controller, allowing it to store the previous INVITE and its 200-OK response. Having this information allows the system to reply locally when a re-INVITE that changes only the media transport addresses is received.

No license requirements to enable this feature.

### SIP re-INVITE Suppression Configuration

To enable SIP re-INVITE Suppression:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-interface) #
```

4. **options**—Set the options parameter by typing **options**, a Space, the option-name **suppress-reinvite** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (sip-interface) # options +suppress-reinvite
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the SIP interface configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

## RFC 4028 Session Timers

The Oracle Communications Session Border Controller supports RFC 4028 Session Timers. In this role, it acts as a B2BUA between two endpoints and then enforces the timer values on each call leg independently. The RFC 4028 abstract states:

This document defines an extension to the Session Initiation Protocol (SIP). This extension allows for a periodic refresh of SIP sessions through a re-INVITE or UPDATE request. The refresh allows both user agents and proxies to determine whether the SIP session is still active. The extension defines two new header fields:

- **Session-Expires**—which conveys the lifetime of the session
- **Min-SE**—which conveys the minimum allowed value for the session timer

The following parameters in the session-timer-profile configuration element are used for this feature:

**session-expires**—The value of the session expires header in seconds

**min-se**—The value of the Min-SE header in seconds (this is a minimum session expires value)

**force-reinvite**—Sets if the Oracle Communications Session Border Controller will send a reINVITE to refresh the session timer when applicable.

**request-refresher**—Set on the outbound side of a call what the Oracle Communications Session Border Controller sets the refresher parameter to. Valid values are uac, uas, or none.

**response-refresher**—Set on the inbound side the value of the refresher parameter in the 200OK message. Valid values are uac or uas.

In this section, the notion that a UAC or UAS supports Session Timers is indicated by the presence of the Supported: timer header and option tag.

## Ingress Call Leg

### Setting 200 OK's Session-Expire value

The session timer is based on the negotiation between each side's session expires value. The final value on the ingress leg is returned by the Oracle Communications Session Border Controller to the UAC, unless there is an error.

The Oracle Communications Session Border Controller can always reduce the session expires value it returns to the UAC. It checks that the Session-Expires: header is larger than the SIP Interface's min-se value. The Oracle Communications Session Border Controller then compares the received Session-Expires: header to the configured session-expires configuration element and uses the lower value for the 200 OK's Session-Expires: header. If this outbound Session-Expires: value is lower than the received Min-SE: header, it will be bumped up to the Min-SE: header's value.

If the Oracle Communications Session Border Controller's Min-SE value is larger than the Session-Expires: header, a 422 (Session Interval Too Small) message is returned to the UAC containing the Oracle Communications Session Border Controller's configured Min-SE value.

When the UAC supports (but does not require) Session Timers and the Oracle Communications Session Border Controller does not support session timers, a 200 OK is returned to the UAC with no indication of session timer support.

### Refresher


The initial UAC, the side that sends the INVITE, can set itself to be the refresher (uac) or the Oracle Communications Session Border Controller as the refresher (uas). Whoever is the refresher is indicated in the 200 OK. If the UAC does not specify any refresher, the Oracle Communications Session Border Controller uses its response-refresher value in the 200 OK. If that value is set to uas, the Oracle Communications Session Border Controller creates and sends a re-INVITE toward the UAC with previously negotiated session expiration values.

Once the Oracle Communications Session Border Controller becomes the refresher, it does not relinquish that role. Then, when the Oracle Communications Session Border Controller sends refresh requests, it does not change any parameters (refresher role & timers) from the initial request negotiation.

### UAC does not Support Session Timers

If the UAC's initial request does not include a Session-Expires: header, then the 200 OK will include the **session-timer-profile > session-expires** value on the ingress leg in the Session-Expires: header.

The Oracle Communications Session Border Controller also inserts the refresher parameter as configured. The orientation of UAC/UAS on the Oracle Communications Session Border Controller's view of a call leg can change if later in the call flow the endpoint designates the Oracle Communications Session Border Controller as the refresher.

 **Note:** When the request doesn't support Session Timers, the Oracle Communications Session Border Controller's reply adds session timer support according to configuration.

If the Oracle Communications Session Border Controller receives a message with a Require: timer header, and the inbound SIP interface or the final UAS do not support session timers, a 420 (Bed Extension) is returned to the UAC.

## Egress Call Leg

### Outbound INVITE Message


When the Oracle Communications Session Border Controller's outbound interface is configured with session timers, it forwards an INVITE to the UAS with the following headers:

Session-Expires— Oracle Communications Session Border Controller inserts the outbound SIP interface's session-timer parameter

Session-Expires refresher parameter— Oracle Communications Session Border Controller inserts the request-refresh parameter

Min-SE— Oracle Communications Session Border Controller inserts the outbound SIP interface's session-timer parameter

Supported—Supported header has the timer option tag

 **Note:** Require/Proxy-Require—If the timer parameter is present in the Require or Proxy-Require: header field in the request received from the UAC, it will be removed.

### No Session Timer Configuration

If the ingress SIP interface supports session timers, and the original INVITE from the UAC included session timer support, the INVITE request sent to the UAS will have no session timer support. However, the Supported: timer header will be created. This ensures that the Oracle Communications Session Border Controller does not get a 421 response for 'timer' from the UAS.

If the ingress SIP interface supports session timers, and the UAC's initial INVITE did not include session timer support, then the INVITE sent to the UAS will have no session timer support (headers) as well.

If the ingress SIP interface does not support session timers, the INVITE is forwarded with no Session Timer alteration.

### UAS Initial Response

Upon receiving a 200 OK from the UAS, if the response specifies uac as the refresher, the 200 OK includes a Session-Expires header and specifies uac as the refresher, the Oracle Communications Session Border Controller will assume the refresher role. If the 200 OK does not include a Session-Expires header, and the egress interface supports session timers, then the Oracle Communications Session Border Controller assumes the refresher role.

### UAS Returns Errors

422 Session Interval Too Small—The Oracle Communications Session Border Controller in response sends the request again with new values in the 'Session-Expires' header field based on the 'Min-SE' value present in the 422 response.

421 Extension Required for 'timer'—This response can only happen if none of the other three entities (UAC, ingress SIP interface and egress SIP interface) support session timers. The 421 is forwarded through the system to the original UAC.

420 Bad Extension for 'timer'—This response should never happen because the Oracle Communications Session Border Controller will never send Require: timer header. But the event this error is received, it will be forwarded to the original UAC.

## Session Refreshes

On either side of the call, the Oracle Communications Session Border Controller can be responsible for initiating the session refreshes or responding to the session refreshes.

### Oracle Communications Session Border Controller as Refresher

The Oracle Communications Session Border Controller sends the refresh request when half the session expiration has elapsed. The Oracle Communications Session Border Controller always wants to remain the refresher and maintain the initially agreed upon session expiration timers.

### Creating the Refresh Message

The refresh message takes the form of a re-INVITE when the force-reinvite parameter in the session timer profile is enabled. If this parameter is disabled and the remote end supports UPDATE requests, an UPDATE message will be sent.

UPDATE messages contain no SDP information.

Re-INVITE messages contain the SDP that is the same as what was sent before.

The refresh request's Session-Expires: header value is set to the existing value for the session. The refresher parameter is set to uac since the Oracle Communications Session Border Controller acts like a UAC for this refresh transaction. The Min-SE header is also included.

### Processing the Refresh Response

The session expires value in the 2xx response is accepted and the timer restarts.

If the remote end does not include any session expiration parameters, the Oracle Communications Session Border Controller continues to support session timers, and assumes that the refresh interval is the same as before.

Any response that is 422 Session Interval Too Small is handled as expected. The Oracle Communications Session Border Controller resends the refresh request again with new values based on the 422 response. Any other response to the refresh request that is not a dialog/usage destroying response is treated like a 200 OK response.

Subsequent refresh requests are created and sent after half the previous refresh interval. If non-2xx, dialog / usage destroying responses are received, the Oracle Communications Session Border Controller reduces the following refresh intervals by half, as long as the final interval is not less than 32 seconds. The Oracle Communications Session Border Controller then uses this period for sending refresh requests until it successfully receives a 2xx response.

### Oracle Communications Session Border Controller as Refresh Responder Processing the Refresh

The refresh request is processed similarly to the initial request regarding the session timer parameters. The session timer for this call leg is restarted when the Oracle Communications Session Border Controller when it sends the 200 OK response for the refresh request.

### Forwarding the Refresh

When the Oracle Communications Session Border Controller receives an UPDATE request, it is forwarded to the other end since the Oracle Communications Session Border Controller cannot determine whether this request is only for session refreshing, or for other purposes as well.

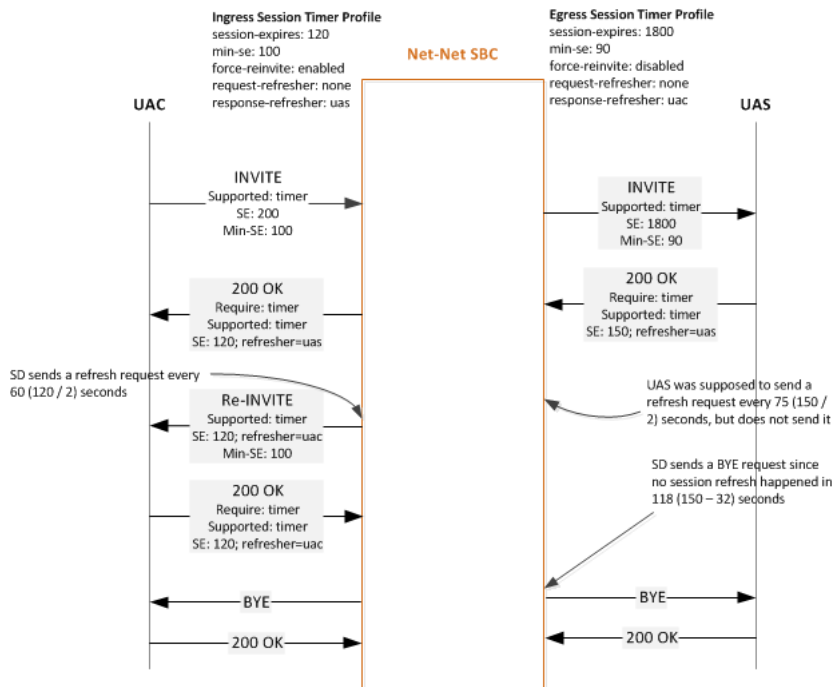
When the Oracle Communications Session Border Controller receives a re-INVITE request, it will determine whether this request needs to be suppressed, or should it be forwarded to the other end.

### Timer Expiration

If the Oracle Communications Session Border Controller fails to receive a session refresh request before the session expiration, the session will be terminated before the full session time. This is computed according to:

$$\text{time} = \text{period} - \min\left(\frac{\text{period}}{3}, 32\right)$$

After the real expiration time elapses, the Oracle Communications Session Border Controller sends a BYE request in both directions to terminate the session.



## Interaction with SIP Features

Consider the following sections that have interactions with RFC 4028 Support.

### sip-config option session-timer-support

A configured session-timers-profile on a SIP interface overrides the session-timer-support option in the SIP config. The Oracle Communications Session Border Controller can still act in proxy mode for some calls and B2BUA for other calls considering which SIP interfaces session timer profiles are not configured for.

### sip-feature Support

When the UAC sends a `Require: timer` header in the initial request and the Oracle Communications Session Border Controller does not support session timers, and no sip-feature configuration element is configured for 'timer' for that realm, the Oracle Communications Session Border Controller replies with a 420 (Bad Extension) response for 'timer'.


When the UAC sends a `Require: timer` header in the initial request and the Oracle Communications Session Border Controller does support session timers, the timer tag is removed from the `Require:` header even if a sip-feature configuration element is configured for 'timer'. This also applies for the `Proxy-Require:` header.

Oracle recommends you do not configure a sip feature configuration element while using the Session Timers feature.

### sip-interface option suppress-reinvite

SIP re-INVITE suppression is automatically enabled for a SIP interface when session timers are enabled. This behavior prevents re-INVITES whose purpose is only for session refreshes from being forwarded to the other call leg. The first re-INVITE received from a UAS on the terminating call leg will be passed to the UAC on the originating call leg since the Oracle Communications Session Border Controller has no prior INVITE request coming from the UAS to match against.

Re-INVITES are suppressed only when the Oracle Communications Session Border Controller receives the same INVITE request back-to-back without any intervening re-INVITE request in the opposite direction, or an UPDATE or PRACK request in either direction.

 **Note:** The SIP reINVITE Suppression parameters are not replicated on the standby system in an HA environment. The first re-INVITE after a switchover will be forwarded to the far end.

Examples

Ex	Messages on Originating Call Leg	Ingress SIP Interface Config	Ingress SIP Interface Config	Messages on Terminating Call Leg
1	INVITE → Supported: timer SE: 200 <hr/> ← 200 OK Require: timer SE: 200; refresher=uas	session-expires: 500 min-se: 200 request-refresher: none response-refresher: uas this element becomes refresher	session-expires: 500 min-se: 400 request-refresher: none response-refresher: uas	INVITE → Supported: timer SE: 500 Min-se: 400 <hr/> ← 200 OK Require: timer SE: 400; refresher=uas
2	INVITE → Supported: timer SE: 1200; refresher=uas <hr/> ← 200 OK Require: timer SE: 500; refresher=uas	session-expires: 500 min-se: 200 request-refresher: none response-refresher: uac this element becomes refresher	session-expires: 500 min-se: 400 request-refresher: uas response-refresher: uas this element becomes refresher	INVITE → Supported: timer SE: 500; refresher=uas Min-se: 400 <hr/> ← 200 OK
3	INVITE → Supported: timer SE: 1200; refresher=uac Min-se: 800 <hr/> ← 200 OK Require: timer SE: 800; refresher=uac	session-expires: 500 min-se: 200 request-refresher: none response-refresher: uas	No session timer configuration this element becomes refresher	INVITE → Supported: timer <hr/> ← 200 OK Require: timer SE: 400; refresher=uac
4	INVITE → Supported: timer SE: 200; refresher=uac <hr/> ← 200 OK Require: timer SE: 200; refresher=uac	session-expires: 500 min-se: 200 request-refresher: none response-refresher: uas	No session timer configuration	INVITE → Supported: timer <hr/> ← 200 OK



Ex	Messages on Originating Call Leg	Ingress SIP Interface Config	Ingress SIP Interface Config	Messages on Terminating Call Leg
5	INVITE → Supported: timer SE: 200 <hr/> ← 200 OK	No session timer configuration	session-expires: 500 min-se: 400 request-refresher: uas response-refresher: uas this element becomes refresher	INVITE → Supported: timer SE: 500; refresher=uas Min-se: 400 <hr/> ← 200 OK
6	INVITE → Supported: timer SE: 200 <hr/> ← 200 OK Require: timer SE: 400; refresher=uas	No session timer configuration SBC behavior stays same as current behavior	No session timer configuration	INVITE → Supported: timer SE: 200 <hr/> ← 200 OK Require: timer SE: 400; refresher=uas
7	INVITE → Require: timer SE: 200 <hr/> ← 420 Unsupported: timer	No SIP feature for timer No session timer configuration SD behavior stays same as current behavior	No session timer configuration	
8	INVITE → Require: timer SE: 200 <hr/> ← 420 Unsupported: timer	SIP feature configured for timer No session timer configuration SD behavior stays same as current behavior	No session timer configuration	INVITE → Required: timer SE: 200 <hr/> ← 420 Unsupported: timer
9	INVITE → Require: timer SE: 200 <hr/> ← 200 OK	SIP feature configured for timer No session timer configuration	session-expires: 500 min-se: 400 request-refresher: none response-refresher: uas this element becomes refresher	INVITE → Supported: timer SE: 500 Min-se: 400 <hr/> ← 200 OK

## SIP Signaling Services

Ex	Messages on Originating Call Leg	Ingress SIP Interface Config	Ingress SIP Interface Config	Messages on Terminating Call Leg
				Require: timer SE: 500; refresher=uac
10	INVITE → Require: timer SE: 200 _____ ← 200 OK Require: timer SE: 200; refresher=uac	No SIP feature for timer session-expires: 500 min-se: 200 request-refresher: none response-refresher: uac	session-expires: 500 min-se: 400 request-refresher: uas response-refresher: uas	INVITE → Supported: timer SE: 500; refresher=uas Min-se: 400 _____ ← 200 OK Require: timer SE: 500; refresher=uas
11	INVITE → Require: timer SE: 200 _____ ← 420 Unsupported: timer	No SIP feature for timer No session timer configuration	session-expires: 500 min-se: 400 request-refresher: none response-refresher: uas	
12	INVITE → SE: 200 _____ ← 200 OK SE: 500; refresher=uas	session-expires: 500 min-se: 500 request-refresher: none response-refresher: uas this element becomes refresher	No session timer configuration	INVITE → _____ ← 200 OK
13	INVITE → _____ ← 200 OK	No session timer configuration	session-expires: 500 min-se: 400 request-refresher: none response-refresher: uas	INVITE → Supported: timer SE: 500 Min-se: 400 _____ ← 200 OK Require: timer SE: 400; refresher=uas
14	INVITE → _____ _____	No session timer configuration	No session timer configuration	

Ex	Messages on Originating Call Leg	Ingress SIP Interface Config	Ingress SIP Interface Config	Messages on Terminating Call Leg
	← 421 Require: timer	SD behavior stays same as current behavior		
15	INVITE → Supported: timer SE: 200 _____ _____ ← 422 Min-se: 400	session-expires: 500 min-se: 400		
16	INVITE → Supported: timer SE: 200 _____ _____ ← 200 OK Require: timer SE: 200; refresher=uac	session-expires: 800 min-se: 90 request-refresher: none response-refresher: uac	session-expires: 800 min-se: 90 request-refresher: none response-refresher: uac	INVITE → Supported: timer SE: 800 Min-se: 90 _____ _____ ← 422 Min-se: 900 _____ _____ INVITE → Supported: timer SE: 900 Min-se: 900 _____ _____ ← 200 OK Require: timer SE: 900; refresher=uas

### RADIUS Interim record Generation

When refresh requests (UPDATE or Re-INVITE) are sent by the Oracle Communications Session Border Controller , no RADIUS Interim records are generated because session parameters do not change when these requests are sent.

When UPDATE requests are received by the Oracle Communications Session Border Controller , no RADIUS Interim records are generated.

When Re-INVITE requests are received by the Oracle Communications Session Border Controller , RADIUS Interim records are generated if the generate-interim parameter is enabled.

### ACLI Configuration

To configure a session timer profile object:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-timer-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# session-timer-profile
ACMEPACKET(session-timer-profile)#
```

4. **name**—Enter a name for this session timer profile.
5. **session-expires**—Enter the session timer value in seconds you wish this object to use natively.
6. **min-se**—Enter the minimum session timer value in seconds for this object.
7. **force-reinvite**—Leave the default of enabled for the Oracle Communications Session Border Controller to always use reINVITEs for session refreshes. Set this parameter to disabled for the Oracle Communications Session Border Controller to try using UPDATEs for session refreshes.
8. **request-refresher**—Set this to the value to insert in the refresher parameter in the Session-Expires: header on the originating call leg that the Oracle Communications Session Border Controller includes in the 200 OK response message. Valid values are **uac** and **uas**.
9. **response-refresher**—Set this to the value to insert in the refresher parameter in the Session-Expires: header on the terminating call leg that the Oracle Communications Session Border Controller includes in the INVITE message. Valid values are **uac**, **uas**, **none**.
10. Type **done** to save your work and continue.

To apply a session timer profile to a SIP interface:

11. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

12. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

13. Type **sip-interface** and press Enter.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

If you are adding this feature to an existing configuration, then you will need to select the configuration you want to edit.

14. **session-timer-profile**—Enter the name of a session timer profile object you have configured and want applied to this SIP interface.
15. Type **done** to save your work and continue.

### Verify Config Validation

The Oracle Communications Session Border Controller's verify-config function checks that the value configured in all sip-interfaces' session-timer-profiles correspond to a configured session-timer-profile name. The following is generated when this check fails:

```
ERROR: sip-interface [id] has reference to session-timer-profile [xyz] which
does not exist
```

**show sipd status**

The show sipd status command now contains new statistic, called Refreshes Sent which reflects the number of refresh requests that the Oracle Communications Session Border Controller has sent. For example:

```
ACMEPACKET#show sipd status
SIP Status          -- Period -- ----- Lifetime -----
Active             High    Total    Total  PerMax   High
Sessions           0       1       0       2       1       1
Subscriptions      0       0       0       0       0       0
Dialogs            0       2       0       4       2       2
CallID Map         0       2       0       4       2       2
Rejections        -       -       0       0       0
ReINVITES         -       -       0       2       1
ReINV Suppress    -       -       0       1       1
Media Sessions    0       1       0       2       1       1
Media Pending     0       0       0       0       0       0
Client Trans      0       3       2      10       3       3
Server Trans      0       0       0       6       3       3
Resp Contexts    0       0       0       6       3       3
Saved Contexts   0       0       0       0       0       0
Sockets          2       2       0       2       2       2
Req Dropped      -       -       0       0       0
Refreshes Sent   0       0       0       0       0       0
DNS Trans        0       0       0       0       0       0
DNS Sockets      0       0       0       0       0       0
DNS Results      0       0       0       0       0       0
Rejected Msgs    0       0       0       0       0       0
```

**305 Response to Registrations on Secondary Interfaces**

Certain devices are provisioned with point of contact (POC) lists for registration. In the context of geographic redundancy, if a UE is unable to register with its primary POC, it proceeds to its secondary POC. It is desirable to designate certain Oracle Communications Session Border Controller as secondary in order to redirect the UE to re-register with the primary SBC.

This feature is designed for Oracle Communications Session Border Controllers sitting behind Oracle Communications Subscriber-Aware Load Balancers (SLBs). This feature allows users to designate SBCs sitting behind SLBs as secondary. When registered endpoints in a realm attempt to register with the primary POC, and registration fails, the UEs proceed to their secondary point of contact (POC). Once connection to the primary POC has been restored, users can execute a command to offload the UE registrations in a given realm with a 305 response to the UEs in order for the UEs to re-register with the next POC. When the scenario consists of two POCs, the endpoint re-registers with the primary POC.

You enable this feature with an option through the SIP interface of the secondary SBC. The **notify sipd offload-users <realm>** command marks endpoints associated with a given realm to receive a 305 message. Once this command is executed, any subsequent requests from UEs in this designated realm receive a 305 message. Once a 305 message is issued, the registration cache is cleared to allow the UE request to be forwarded to the primary SBC. If the UE re-registers again with the secondary SBC, it will not receive a second 305 message. The **notify sipd offload-users <realm>** command must be executed again in order to repeat the process.

The **show registration** command is expanded to include a counter for the number of registrations received on secondary interfaces, as well as the number of endpoints marked to receive a 305.

SNMP support for this feature tracks the total count of registrations on secondary interfaces. The **apSipSecInterfaceRegThresholdExceededTrap** is generated when the total number of registrations on secondary interfaces exceeds the configured threshold, and **apSipSecInterfaceRegThresholdClearTrap** is generated when the total count falls below the configured threshold.

### ACLI Instructions and Examples

To enable this feature, you must select an option for the SIP-interface of the secondary Oracle Communications Session Border Controller:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. **options**—Type **secondary** to designate this system as the secondary point of contact.

```
ACMEPACKET(session-router)# options secondary
```

5. Save and activate your configuration.

### notify sipd offload-users

The **notify sipd offload-users <realm>** command marks the endpoints associated with the given <realm> to receive a 305 message. Once this command is executed, any re-registrations received from an endpoint in that realm receives a 305 message.

A confirmation message is sent if the command succeeds:


```
Realm <realm> is marked to offload users at HH:MM:SS
```

If the given <realm> is not a secondary realm, the following message is displayed:

```
Realm <realm> is not secondary
```


If the given <realm> does not exist, the following message is displayed:

```
Realm <realm> is not found
```

 **Note:** This command is only valid for secondary systems. If the secondary option is not enabled for the Oracle Communications Session Border Controller, this command produces an error message.

### show registration

The **show registration** command displays the total number of endpoints associated with secondary interfaces, and the total number of endpoints marked to receive a 305 response.

 **Note:** The total number of SIP calls to receive a 305 response are included in the Registrations marked for 305 counter.

The total number of SIP calls associated with secondary realms are included in this count.

```
AcmePacket# show registration
11:02:13-102
SIP Registrations          -- Period --  ----- Lifetime -----
Active   High   Total   Total  PerMax   High
User Entries           1       1       1       1       1       1
...
Secondary Interface Registrations=NNN
Registrations marked for 305=MMM
access1: N1
access2: N2
```

## show registration sipd

The **show registration sipd** command includes the following parameters:

**show registration sipd sec-by-ip**—displays the registrations sent to secondary interfaces by IP address.

```
Registration Cache                               MON FEB 06 2012 16:33:49
                                                Num
  IP Address      User                          Contacts Registered at
-----
. . . . .
```

**show registration sipd sec-by-user**—displays the registrations sent to secondary interfaces by user name.

```
Registration Cache                               MON FEB 06 2012 16:33:49
                                                Num
  Phone Number    User                          Contacts Registered at
-----
. . . . .
```

## show sipd endpoint-ip

If the registered endpoint is associated with a secondary SIP interface, the **show sipd endpoint-ip** command displays SIP Interface: secondary.

```
# show sipd endpoint-ip 1
User <sip:12341@172.16.101.67>
  Contact exp=290
    UA-Contact: <sip:12341@172.16.26.1:5060> UDP keep-acl
                realm=net172 local=172.16.101.67:5060 UA=172.16.26.1:5060
    SD-Contact: <sip:12341-2k86jbp9bj925@192.168.101.67:5060> realm=net192
    Call-ID: 1-839@172.16.26.1'
    SA=192.168.26.2
    Service-Route='<sip:h19216806003.dg.com;lr>'
SIP Interface: secondary
```

## SNMP Configuration

To configure the threshold for the number of registrations on secondary SIP interfaces in order to generate a trap:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

4. **options**—Type **sec-reg-threshold =X**, where X equals the number of registrations received on all secondary interfaces. Once that number has been reached, the apSipSecInterfaceRegThresholdExceededTrap is generated.

```
ACMEPACKET(session-router)# options secondary
```

5. **options**—Type **sec-reg-threshold-clear =Y**. When the number of registrations received on all secondary interfaces drop below Y, the apSipSecInterfaceRegThresholdExceededClearTrap is generated.



**Note:** The value of **sec-reg-threshold-clear =Y** must be less than **sec-reg-threshold =**. The values cannot be equal.

```
ACMEPACKET(session-router)# options secondary
```

## SIP Signaling Services

6. **options**—Type **sec-reg-threshold-alarm-severity** = [minor | major | critical] to set the alarm severity for APP\_ALARM\_SEC\_INTF\_REG\_EXCEED.
7. Save and activate your configuration.

## SNMP

The apSipSecInterfaceRegThresholdExceededTrap trap is generated when the total count of all registrations on the secondary interfaces cross **sec-reg-threshold-exceed** is exceeded.

Trap Name	Object Identifier Name: Number	Description
Object Identifier Name: apSipInterfaceRegNotificationsGroup		
apSipSecInterfaceRegThresholdExceededTrap	.1.3.6.1.4.1.9148.3.15.2.1.2.0.1	The trap will be generated if the total number of registrations on all secondary SIP interfaces exceed threshold.
apSipSecInterfaceRegThresholdClearTrap	.1.3.6.1.4.1.9148.3.15.2.1.2.0.2	The trap will be generated if the total number of registrations on all secondary SIP interfaces go below clear threshold.

An alarm is also generated for this event. The severity of the alarm is configured in the **sec-reg-threshold-alarm-severity** option.

Alarm Name	Alarm ID	Alarm Severity	Cause(s)	Example Log Message	Trap Generated (Trap Reference)
Hardware Alarms					
APP_ALARM_SEC_INTF_REG_EXCEED	0X50018	Minor by default. The severity is configurable.	The total number of registrations on all secondary SIP interfaces exceeds the configured threshold.	Number of Secondary Interface registrations <total> has exceeded the secondary interface registration threshold of <max>.	apSyslogMessageGenerated (ap-slog.mib) apEnvMonStatusChangeNotification (ap-env-monitor.mib) apSysMgmtFanTrap (ap-smgmt.mib)

The following objects monitor the number of registrations on secondary interfaces:

SNMP GET Query Name	Object Identifier Name: Number	Description
Object Identifier Name: apSipInterfaceRegObjectsGroup		
apSipSecInterfaceTotalRegistrations	.1.3.6.1.4.1.9148.3.15.1.1.1.1.0	The total number of registrations on all secondary SIP interfaces.
apSipSecInterfaceRegThreshold	.1.3.6.1.4.1.9148.3.15.1.1.1.2.0	The max threshold for registrations on all secondary SIP interfaces beyond which trap and alarm will be raised



<b>SNMP GET Query Name</b>	<b>Object Identifier Name: Number</b>	<b>Description</b>
apSipSecInterfaceClearThreshold	.1.3.6.1.4.1.9148.3.15.1.1.1.3.0	The threshold for registrations on all secondary SIP interfaces below which if alarm was raised before, it will be cleared.



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## H.323 Signaling Services

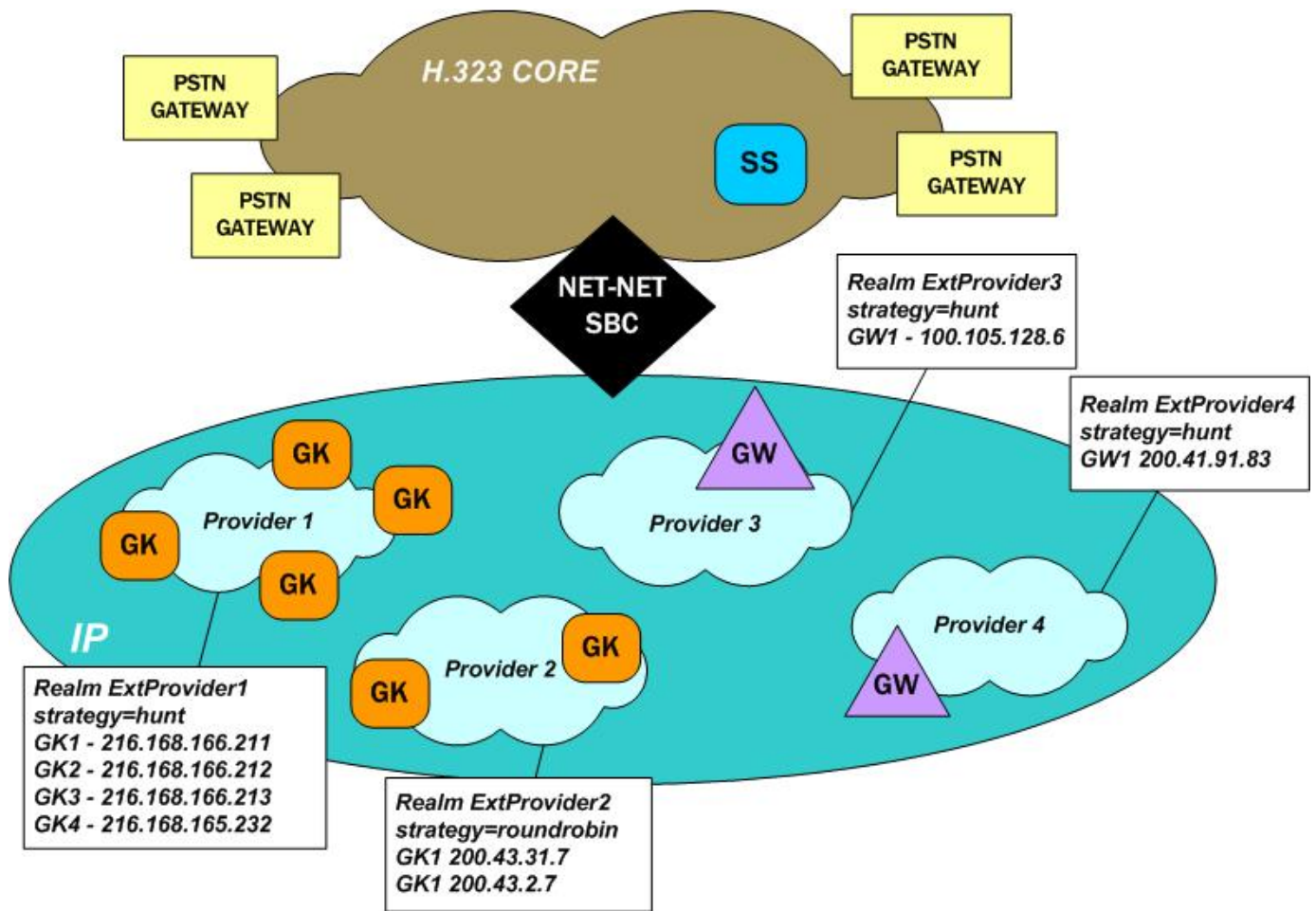
The Oracle Communications Session Border Controller supports H.323 signaling in a way that permits interworking between different H.323 configurations from different providers and carriers. H.323 signaling capabilities on the Oracle Communications Session Border Controller include:

- H.323 V4—Improves on previous versions of the protocol in functionality, scalability, and reliability
- H.225 call signaling with RAS—Establishes connections between H.323 endpoints so real-time data can be exchanged
- H.245—Establishes the type of media flow and manages that flow after it has started
- H.245 tunneling—Encapsulates H.245 messages within H.225/Q.931 messages; when enabled and used with a firewall, one less TCP port is needed for incoming connections
- Fast Start (and Fast Start with parallel H.245)
- H.323 Annex E support for UDP signaling—Provides for multiplexed call signaling over UDP to increase potential call volume and enhance performance

### Peering Environment for H.323

---

The following diagram shows a peering environment for H.323, with the Oracle Communications Session Border Controller positioned between the H.323 core and external providers.



The configuration information shown in the diagram can help you to understand how some basic Oracle Communications Session Border Controller concepts work. The providers in this depiction are configured as realms, and the strategies you see are for session agent group. What you do not see in this diagram is the fact that the Oracle Communications Session Border Controller is configured with sets of H.323 interfaces within it. These interfaces are internal (for an internal provider) and external (for the external providers you see).

## Overview

Using H.323 on your Oracle Communications Session Border Controller, you can implement different signaling modes and use features to enhance H.323 capabilities. In the information that follows, you will find detailed explanations of the H.323 signaling mode and of the features available. This chapter gives operational details and later outlines the steps you need to take when features require configuration.

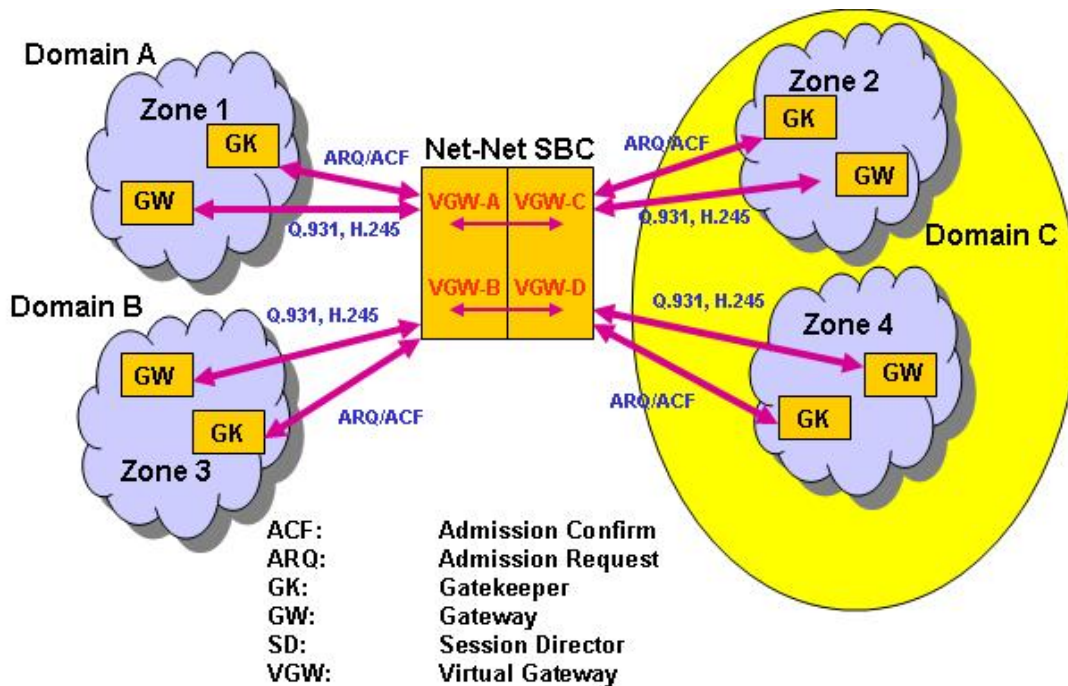
## Signaling Modes of Operation

Your Oracle Communications Session Border Controller can operate in different H.323 signaling modes:

- Back-to-back gateway signaling
- Back-to-back gatekeeper proxy and gateway
- Interworking gatekeeper/gateway

### Back-to-Back Gateway Signaling

This section explains how signaling takes place when the Oracle Communications Session Border Controller functions as a B2BGW for H.323. The following diagram illustrates the Oracle Communications Session Border Controller acting as a B2BGW.



When configured as a B2BGW, the Oracle Communications Session Border Controller appears as multiple H.323 gateways to multiple networks. You can think of the Oracle Communications Session Border Controller as having virtual gateways, that discovers and registers with a gatekeeper in its respective domain. In this configuration, you need to set the service mode (`isgateway`) parameter for the H.323 interface to enabled for two H.323 interfaces. These interfaces are related either through their outgoing interface (`assoc-stack`) parameters or through routing policies.

If you configure your Oracle Communications Session Border Controller to operate in this mode, it does not issue or respond to LRQs by either confirming them or rejecting them.

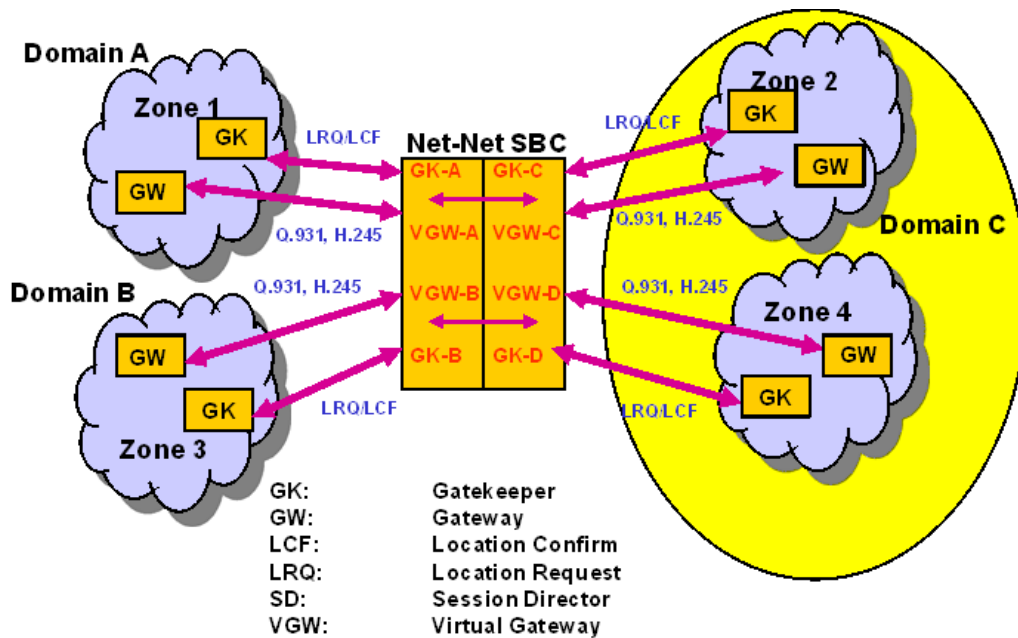
In the diagram above, the Oracle Communications Session Border Controller sends ARQs to the corresponding gatekeeper in its zone when a call is received on the associated interface. In this behavior, the Oracle Communications Session Border Controller acts as a gateway, complying with the H.323 standard, and registers with the configured gatekeeper in its assigned zone. You set all parameters related to the gateway registrations, such as gateway prefix numbers, in the H.323 interface configuration.

In this mode, you can also configure the Oracle Communications Session Border Controller to run like a gateway without a gatekeeper by turning off automatic discovery (`auto-gk-discovery`) for the remote gatekeeper. When the Oracle Communications Session Border Controller receives a Setup message, it does not send an ARQ and there is no registration for admission requests. Without automatic gateway discovery, the Oracle Communications Session Border Controller uses the local policy to find the appropriate destination for the call. This destination is normally the IPv4 address of the endpoint or gateway, using the well-known port 1720.

If you enable this capability, then the Oracle Communications Session Border Controller finds a gatekeeper.

### Back-to-Back Gatekeeper Proxy and Gateway

This section explains how signaling takes place when the Oracle Communications Session Border Controller functions as a back-to-back gatekeeper proxy and gateway for H.323. The following diagram illustrates the Oracle Communications Session Border Controller acting as a B2B gatekeeper proxy and gateway.



In this application, with the service mode (`isgateway`) parameter set to disabled, the Oracle Communications Session Border Controller responds to LRQs and issues LCFs and LRJs. It sends LRQs and LCFs/LRJs to the local IPv4 address for the H.323 interface. The Oracle Communications Session Border Controller responds to the LRQs by providing a signaling address that performs gateway functions.

When you use it as a back-to-back gatekeeper proxy and gateway, the Oracle Communications Session Border Controller does not issue ARQs. In addition, all parameters related to registration, such as gateway prefix numbers, are ignored.

When you do not configure a gatekeeper, the Oracle Communications Session Border Controller uses the local policy to find the appropriate destination for the call. If there is a matching local policy, the Oracle Communications Session Border Controller returns an LCF to the originating gateway. If no local policy matches, the Oracle Communications Session Border Controller rejects the call by sending an LRJ.

### Interworking Gatekeeper-Gateway

This section explains how signaling takes place when the Oracle Communications Session Border Controller functions as an interworking gatekeeper-gateway for H.323. The following diagram shows the Oracle Communications Session Border Controller acting as an interworking gatekeeper-gateway.



## H.323 Signaling Services

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You should also decide how you want to set up realms and routing (including the use of session agents and session agent groups) to support H.323 operations.

## Global H.323 Settings

---

When you configure H.323 signaling for your Oracle Communications Session Border Controller, you set global and per-interface parameters. The global parameters govern how the Oracle Communications Session Border Controller carries out general H.323 operations, and these settings are applied to all interfaces you configure for H.323 use. For example, you can turn H.323 support on and off for the entire Oracle Communications Session Border Controller using these settings.

## Global H.323 Settings Configuration

For the CLI, global H.323 parameters are:

```
state          State of the H.323 protocol
log-level      Log level for H.323 stacks
response-tmo   maximum waiting time in sec for response to a SETUP message
connect-tmo    maximum waiting time in sec for establishment of a call
options        optional features/parameters
```

### Accessing Global H.323 Parameters

To access the global H.323 configuration parameters in the CLI:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **h323** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # h323
```

From this point, you can configure global H.323 parameters. To view all H.323 configuration parameters, enter a **?** at the system prompt. Access to the H.323 interface (**h323-stack**) configuration also appears.

### Global H.323 Settings

To configure global H.323 parameters:

1. **state**—Enable or disable the state of H.323 signaling. The default value is **enabled**. Valid values are:
  - enabled | disabled
2. **response-tmo**—Enter the amount of time in seconds that the Oracle Communications Session Border Controller waits between sending a Setup message and tearing it down if there is no response. The default value is **4** and we recommend you leave this parameter set to this value. The valid range is:
  - Minimum—0
  - Maximum—999999999

A response might be any of the following messages: Call Proceeding, Connect, or Alerting.

3. **connect-tmo**—Enter the amount of time in seconds that the Oracle Communications Session Border Controller waits between sending a Setup message and tearing it down if it does not specifically receive a Connect message from the endpoint. The default is **32** and we recommend that you leave this parameter set to this value. The valid range is:
  - Minimum—0
  - Maximum—999999999




Receiving a Proceeding or Alert message from the endpoint does not keep this timer from expiring.

4. **options**—Set any options for H.323 features that you want to use. This parameter has a global impact on H.323 behavior, rather than being applied on a per-interface basis.

If you do not configure options for global H.323 behavior, none appears in the configuration display.

5. **log-level**—Set the process log level for monitoring all H.323 activity on the Oracle Communications Session Border Controller. The default is **INFO** and leaving this parameter set to this value provides an intermediate amount of detail in the logs. Other valid values are:

 **Note:** Any log level you set here overrides the log level you set in the system configuration’s process log level parameter.

Numerical Code	Acme Packet Log Enumeration	Description
1	EMERGENCY	Logs conditions of the utmost severity that require immediate attention.
2	CRITICAL	Logs events of serious condition that require attention as soon as possible.
3	MAJOR	Logs conditions indicating that functionality is seriously compromised.
4	MINOR	Logs conditions indicating that functionality has been impaired in a minor way.
5	WARNING	Logs conditions indicating irregularities in performance.
6	NOTICE	For Acme Packet customer support.
7	INFO	
8	TRACE	
9	DEBUG	

## H.323 Interfaces

You need to configure H.323 interfaces for inbound and outbound traffic. When you configure H.323 interfaces, you can set:

- Identity and state
- Realm and H.323 interface associations
- H.323 interface settings for the interface’s IPv4 address, RAS and Q.931 ports, maximum number of Q.931 ports to allow, and any Annex E support you need
- H.323 system resource allocation

### H.323 Interfaces Configuration

These are the CLI parameters that you set:

name	Name of the stack
state	State of the stack
isgateway	Enable the stack to run as a gateway
terminal-alias	List of aliases for terminal
ras-port	Listening port for RAS request
gk-identifier	Gatekeeper's identifier
q931-port	Q.931 call signalling port
alternate-transport	Alternate transport addresses/ports

## H.323 Signaling Services

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q931-max-calls	Maximum number of Q.931 calls
max-calls	Stack's maximum number of calls
max-channels	Maximum number of channels per channel

To access the H.323 interface (h323-stack) and service mode parameters:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET (session-router) # h323
```

4. Type **h323-stacks** and press Enter.

```
ACMEPACKET (h323) # h323-stacks  
ACMEPACKET (h323-stacks) #
```

From this point, you can configure H.323 interface and service mode parameters. To view all H.323 interface parameters, enter a **?** at the system prompt. The display also includes H.323 service mode parameters.

### Identity and State

To set the identity and state of the H.323 interface:

1. **name**—Enter a name for the H.323 interface using any combination of characters entered without spaces. For example: InternalGK1.
2. **state**—Enter the state of this H.323 interface. The default value is **enabled**. Valid values are:
  - enabled | disabled

### Realm and Interface Associations

To link this H.323 interface to a realm and to an outgoing H.323 interface:

1. **realm-id**—Enter the identifier for the realm served by this H.323 interface. This parameter must be configured with a valid identifier value from a realm configuration.
2. **assoc-stack**—Enter the name of the outgoing H.323 interface that you want to associate with the H.323 interface you are configuring. To use realm bridging with static routing, you need to set the outgoing H.323 interface. If you do not enter a name, the Oracle Communications Session Border Controller uses dynamic, policy-based selection using the local policy.

### H.323 Signaling Interface Settings

You can set the following parameters to define basic settings for your H.323 interface. This is where you set the IPv4 address for opening sockets, the RAS and Q.931 ports, and the maximum number of Q.931 calls that you want to allow.

This is also where you establish Annex E alternate transport. Annex E supports multiplexed call signaling over UDP so that call volume and performance are potentially enhanced. If you do not configure Annex E support, then this H.323 interface does not listen for Annex E requests.

To configure H.323 interface settings:

1. **local-ip**—Enter the IPv4 address that the H.323 interface uses when opening sockets; this is the default H.323 interface IPv4 address. You must use a valid IPv4 address. For example: 192.168.2.5. The default value is **0.0.0.0**.
2. **ras-port**—Enter the number of the port on the local IPv4 address (**local-ip**) on which the Oracle Communications Session Border Controller listens for RAS requests. We recommend that you set this parameter to its default, the well-known port **1719**. The valid range is:
  - Minimum—0

- Maximum—65535

If you set this parameter to **0**, the Oracle Communications Session Border Controller uses a port assigned by the operating system.

3. **q931-port**—Enter the number for the port on the local IP address for the Q.931 call signaling port. We recommend that you leave this parameter set to its default, **1720**. The valid range is:

- Minimum—0
- Maximum—65535

4. **q931-max-calls**—Enter the maximum number of concurrent Q.931 calls you want to allow. The default value is **200**; however, this value should be less than the maximum number of calls you set when configuring H.323 features. The valid range is:

- Minimum—0
- Maximum—65535

If the number of received Q.931 calls exceeds this number, the H.323 interface returns a busy state.

5. **alternate-transport**—Enter a list of one or more Annex E IPv4 address and port combinations for alternate transport. If you do not configure this list, then the Oracle Communications Session Border Controller does not listen for incoming Annex E requests. You must enter the IPv4 address and port combination in the following format, where the two are separated by a colon: **IPv4Address : Port**.

## H. 323 System Resource Allocation

You can set the following parameters to determine how many concurrent calls and concurrent channels you want to allow for each H.323 interface.

To allocate H.323 system resources:

1. **max-calls**—Enter the maximum number of concurrent calls allowed on this H.323 interface. The default value is **200**. The valid range is:

- Minimum—0
- Maximum—4294967295

2. **max-channels**—Enter the maximum number of concurrent channels allowed for each call associated with this H.323 interface. The default value is **6**. The valid range is:

- Minimum—0
- Maximum—4294967295

The Oracle Communications Session Border Controller checks this parameter on initialization to reserve the appropriate network resources.

## H.323 Service Modes

When you set the H.323 service mode, you configure parameters that define what type of service an H.323 interface provides. These parameters govern how the interface functions when you want it to behave as a gatekeeper or as a gateway.

This is also where you set options that support particular H.323 features for a specific interface. These options are different from the ones you set in the global H.323 configuration because they apply only to the interface where you specify them.

## H.232 Service Modes Configuration

These are the CLI parameters that you set:

<code>isgateway</code>	Enable the stack to run as a gateway
<code>registration-ttl</code>	Number of seconds before the registration becomes
<code>invalid</code>	

## H.323 Signaling Services

<code>terminal-alias</code>	List of aliases for terminal
<code>auto-gk-discovery</code>	Enable automatic gatekeeper discovery
<code>multicast</code>	RAS multicast address
<code>gatekeeper</code>	Gatekeeper's address and port
<code>gk-identifier</code>	Gatekeeper's identifier
<code>h245-tunneling</code>	Enable H.245 Tunneling support
<code>prefixes</code>	List of supported prefixes
<code>process-registration</code>	Enable Registration Request processing
<code>allow-anonymous</code>	allowed requests from H.323 realm

To configure the service mode for the H.323 interface:

1. **allow-anonymous**—Enter the admission control of anonymous connections from an H.323 realm accepted and processed by this H.323 stack. The default value is **all**. The valid values are:
  - **all**—Allow all anonymous connections
  - **agents-only**—Allow requests from session agents only
  - **realm-prefix**—Allow session agents and addresses matching the realm prefix
2. **is-gateway**—To use this interface as an H.323 gateway, leave this parameter set to **enabled**, its default value. If you want to use this interface as an H.323 gatekeeper, set this parameter to **disabled**. Valid values are:
  - enabled | disabled
3. **terminal-alias**—Enter a list of one or more aliases that identify the H.323 interface. This value is either the gateway alias or the gatekeeper identifier, depending on the mode you configure for the interface. The aliases are set in the sourceInfo information element of outgoing ARQs.

### Configuring Gateway Only Settings

If you are using the H.323 interface as a gateway, you might want to set registration time-out and address prefix parameters.

To configure gateway only settings:

1. **registration-ttl**—Enter the number of seconds before a registration becomes invalid. This value is used during the initial registration process. However, when a registration is confirmed, the time-to-live (TTL) value set by the gatekeeper in the Registration Confirm (RCF) message overrides this value. The default value is **120**. The valid range is:
  - Minimum—0
  - Maximum—4294967295
2. **prefixes**—Enter a list of prefixes for this H.323 interface. Possible prefix types include:
  - H.323 ID | E.164 | URL | IPv4 address

These prefixes are sent from a gateway interface to a gatekeeper and indicate valid prefixes accepted by that interface for incoming calls. They are used if the interface is configured as a gateway (the **is-gateway** parameter is set to enabled).

Your entries for this parameter must appear as they do in the following example:

```
e164=17817566800 url=http://www.companyname.com
h323-ID=xyz email=user@companyname.com
ipAddress=63.67.143.4:2000
```

### Gatekeeper Proxy Settings

If you are using the H.323 stack as a gatekeeper proxy, you might want to set:

- Whether registration processing is enabled or disabled
- Whether or not this H.323 interface is signaling-only
- At what H.225 call stage the H.245 procedures should be initiated

To configure gatekeeper proxy settings:

1. **process-registration**—To have the Oracle Communications Session Border Controller drop all RRQs, meaning that it does not acknowledge any requests, leave this parameter set to **disabled**, its default. To have the Oracle Communications Session Border Controller process any RRQs that arrive on this H.323 interface, set this parameter to **enabled**. Valid values are:
  - enabled | disabled

When registration processing is enabled and the Oracle Communications Session Border Controller receives an RRQ on this H.323 interface, it will route the request to the appropriate gatekeeper. After the gatekeeper confirms that registration with an RCF, the Oracle Communications Session Border Controller also confirms it with the endpoint that sent the RRQ. Then the registration becomes part of the Oracle Communications Session Border Controller's registration cache. If this endpoint does not confirm the registration, then the Oracle Communications Session Border Controller will reject the registration with an RRJ and will not cache it.
2. **proxy-mode**—Set this field to the proxy mode that you want to use for the signaling only operation mode. Valid values are:
  - H.225 | H.245

You can leave this field blank (default) if you are not using a proxy mode.
3. **h245-stage**—Set this field to the stage at which the Oracle Communications Session Border Controller transfers the H.245 address to the remote side of the call, or acts on the H.245 address sent by the remote side. The default value is **connect**. Valid values are:
  - Setup | Alerting | Connect | Proceeding | Early | Facility | noh245 | Dynamic

## H.323 Features

This section provides general descriptions of the H.323 features available on the Oracle Communications Session Border Controller and instructs you in how to configure them. Not all of the features described in that chapter require configuration.

### Fast Start Slow Start Translations

The Oracle Communications Session Border Controller can translate between Fast Start H.323 endpoints and Slow Start H.323 endpoints. Using this feature, you can reduce delay in establishing media, improve performance, and reduce network congestion caused by a high number of messages being exchanged. Fast Start and Slow Start calls handle information about media for a session in different ways. In a Fast Start call, information about the media is contained in the Setup message. In a Slow Start call, that information is exchanged between endpoints after the session has been established.

When you Fast Start/Slow Start translation, the Oracle Communications Session Border Controller can take a Slow Start call from an H.323 endpoint that does not support Fast Start and re-initiate that call as Fast Start. It also allows an H.323 endpoint that does not support Fast Start to receive a Slow Start call from a Fast Start source because the Oracle Communications Session Border Controller performs all necessary translations.

For the ACLI, the following parameters apply:

<code>fs-in-first-msg</code>	Fast Start must be sent in 1st response to Setup message
<code>call-start-fast</code>	Enable outgoing Fast Start call
<code>call-start-slow</code>	Enable outgoing Slow Start call
<code>media-profiles</code>	list of default media profiles used for outgoing call

### Fast Start to Slow Start Translation

The Oracle Communications Session Border Controller supports translations from H.323 Fast Start to Slow Start. Using this feature, an H.323 endpoint that only supports Slow Start can call from a Fast Start source when that call goes through the Oracle Communications Session Border Controller.

## H.323 Signaling Services

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In a Fast Start call, the originating H.323 endpoint sends a fastStart element in its Setup message. This element contains H.245 OLC messages that allow Fast Start endpoints to establish a media stream when the call is connected. As a result fewer messages are exchanged between the H.323 endpoints than there would be for a Slow Start call (where the fastStart element does not appear). Because media information is sent in the Setup request for the session, there is no need to use the media profiles when converting a Fast Start call to Slow Start.

When you enable the slow start option in the H.323 stack configuration, the Oracle Communications Session Border Controller performs Fast Start to Slow Start conversion. During the translation, the Oracle Communications Session Border Controller retains the media information included in the incoming Fast Start call as it negotiates a connection with the Slow Start endpoint. After a connection with the Slow Start endpoint has been established, the Oracle Communications Session Border Controller negotiates the media capabilities.

### Slow Start to Fast Start Translation

When you configure your Oracle Communications Session Border Controller to support H.323 Slow Start to Fast Start translations, you enable an H.323 endpoint that only supports Slow Start to initiate and sustain communication with an H.323 Fast Start endpoint. The Oracle Communications Session Border Controller resolves the Slow Start limitation of exchanging information about media (OLC messages) after the call is connected. The OLC message opens a logical channel, or a unidirectional or bi-directional path used to transmit media packets. Using the Oracle Communications Session Border Controller, you can negotiate the construction of media flows differently, which is described in this section.

When you enable the Fast Start option for calls in the H.323 stack configuration, the Oracle Communications Session Border Controller performs the translation of a Slow Start call into Fast Start. When it receives a Slow Start call, the Oracle Communications Session Border Controller determines its destination and the H.323 stack it uses for the outgoing call.

It is a requirement of this kind of translation that you configure and use media profiles. Since a Slow Start call does not negotiate media until after the call is connected, there needs to be an assumption made about the media to set up a Slow Start to Fast Start call. Media profiles fill this role, and they are assumed to be part of a correct configuration.

The following describes possible scenarios for Slow Start to Fast Start translations.

- When a Slow Start call arrives at the Oracle Communications Session Border Controller and matches one of the session agents that has a media profiles list configured, the outgoing call is set up as a Fast Start call. The session agent's media profiles are used for the logical channels. You must configure the media profiles to reference a codec the endpoint accepts.

If there are no media profiles configured for the session agent, then the Oracle Communications Session Border Controller uses the media profiles list in the H.323 stack configuration to open the logical channels.

- If a Slow Start call arrives at the Oracle Communications Session Border Controller and its destination does not match one of the session agents, the Oracle Communications Session Border Controller uses the media profiles list in the H.323 stack configuration for the outgoing call. If there is a list of media profiles, the outgoing call is set up as a Fast Start call with the media profiles list used to open the logical channels.

If there is no list of media profiles for the outgoing H.323 interface, the Oracle Communications Session Border Controller does not perform Slow Start to Fast Start translation. The Slow Start call exits the Oracle Communications Session Border Controller as it arrived—as a Slow Start call.

- If the egress H.323 interface has the Fast Start option disabled, then the outgoing call uses the Slow Start mode, and the Oracle Communications Session Border Controller does not perform Slow Start to Fast Start translation. In this case, the Slow Start call also exits the Oracle Communications Session Border Controller as it arrived—as a Slow Start call.

### Slow Start Fast Start Prerequisites

To perform Fast Start/Slow Start translations, you need to have a standard two-interface configuration already in place.

If you are using the Slow Start to Fast Start translations, you must configure appropriate entries in the media profiles list which is part of the translation parameters. The Fast Start/Slow Start Translations section of the Oracle

Communications Session Border Controller Feature chapter describes how the media profiles are used. The list contains the names of media profiles that you configure in the media profile configuration.

Some media profiles are configured by default. If the information you have configured for a media profile collides with the defaults, then your configured ones are loaded. If there are no collisions, then the Oracle Communications Session Border Controller loads the configured and default profiles. The default media profiles are:

Type	Payload	Encoding	Bandwidth
audio	0	PCMU	0
audio	2	G726-32	0
audio	4	G723	0
audio	8	PCMA	0
audio	9	G722	0
audio	15	G728	0
audio	18	G729	0
audio	101	telephone-events	0

Ensure that you use the name of a configured media profile when you enter values in the media profiles list.

### Media Profile Configuration

In the CLI, you can set media profiles that are required for translating H.323 Slow Start to Fast Start. In the CLI, you set the following:

```

name          encoding name used in sdp rtpmap attribute
media-type    media type used in sdp m lines
payload-type  rtp payload type used in sdp m lines
transport     transport protocol used in sdp rtpmap attribute
req-bandwidth amount of bandwidth in kilobits required
frames-per-packet maximum number of frames per packet
parameters    list of <name=value> pairs separated by space
average-rate-limit average rate limit of rtp flow

```

To configure a media profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **media-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# media-profile
```

From this point, you can configure media profiles parameters. To view all media profiles configuration parameters, enter a **?** at the system prompt.

4. **name**—Enter the encoding name used in the SDP rtpmap attribute. You must enter a name to uniquely identify the media profile, and you will use this value to make lists of media profiles in H.323 interface configurations.
5. **media-type**—Leave this parameter set to its default, **audio**. Valid values are:
  - audio | video | application | data | image | text
6. **payload-type**—Enter the payload type number that corresponds to the encoding name you entered in Step 4. This value identifies the format in the SDP m lines. There is no default value for this parameter. The About Payload Types section contains a table of standard audio and visual encodings.

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**Note:** When you use the RTP/AVP transport method, this value must be numeric.

7. **transport**—Enter the type of transport protocol used in the SDP rtpmap attribute. The default is **RTP/AVP**. Valid values are:
  - RTP/AVP | UDP
8. **req-bandwidth**—Enter the total bandwidth in kilobits that the media requires. The default value is **0**. The valid range is:
  - Minimum—0
  - Maximum—4294967295
9. **frames-per-packet**—Enter the maximum number of frames to use per RTP packet. Leaving this parameters set to **0**, its default value means that it is not being used. The valid range is:
  - Minimum—0
  - Maximum—256

The interpretation of this value varies with codec type and with specific codec.

  - For frame-based codecs, the frame size is specific to each. For example, a G.729 frame contains ten milliseconds of audio, while a G.723.1 codec frame contains thirty milliseconds.
  - For sample-based codecs such as G.711, each frame contains one millisecond of audio.
10. **parameters**—Enter additional codec information. For example, the G.723.1 codec can have an additional `silenceSuppression` parameter.
11. **average-rate-limit**—Enter the maximum speed in bytes per second for the flow that this media profile applies to. The default value is **0**. The valid range is:
  - Minimum—0
  - Maximum—125000000
12. **peak-rate-limit**—Enter the peak rate for RTP flows in bytes per seconds. The default is **0**. The valid range is:
  - Minimum—0
  - Maximum—125000000
13. **max-burst-size**—Enter the maximum data size at peak rate in bytes. The default is **0**. The valid range is:
  - Minimum—0
  - Maximum—125000000
14. **sdp-bandwidth**—Enable this parameter to use the AS bandwidth modifier in the SDP in the conditions for the application specific bandwidth modifier. The default is **disabled**. Valid values are:
  - enabled | disabled
15. **sdp-rate-limit-headroom**—Specify the percentage of headroom to be added while using the AS bandwidth parameter while calculating the **average-rate-limit** (rate limit for the RTP flow). The default is **0**. The valid range is:
  - Minimum—0
  - Maximum—100

### Fast Start/Slow Start Configurations

When you configure an H.323 interface, you configure it for either Fast Start to Slow Start translation or for Slow Start to Fast Start translation. You cannot configure one H.323 interface for both translation modes.

In the ACLI, you will set the following:

```
fs-in-first-msg      Fast Start must be sent in 1st response to Setup message
call-start-fast      Enable outgoing Fast Start call
call-start-slow      Enable outgoing Slow Start call
media-profiles       list of default media profiles used for outgoing call
```



To configure H.323 interfaces for Fast Start/Slow Start translations:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
```

4. Type **h323-stacks** and press Enter.

```
ACMEPACKET(h323)# h323-stacks
ACMEPACKET(h323-stacks)#
```

From this point, you can configure H.323 interface and service mode parameters. To view all H.323 interface parameters, enter a **?** at the system prompt. The display also includes H.323 service mode parameters.

5. **fs-in-first-msg**—Enable this parameter if you want to include Fast Start fields in the first message that the Oracle Communications Session Border Controller uses to respond to a Setup message. Usually, the first message sent is a Proceeding message. If you do not want Fast Start fields included, leave this parameter set to its default value **disabled**. Valid values are:

- enabled | disabled

6. **call-start-fast**—Enable this parameter if you want Slow Start calls to be translated to Fast Start when this H.323 interface is chosen as the outgoing interface. If this parameter is **enabled**, **call-start-slow** has to remain disabled. The default value is **enabled**. Valid values are:

- enabled | disabled

If you set this parameter set to disabled (default), the outgoing call will be set up in the same mode as the incoming call.

7. **call-start-slow**—Enable this parameter if you want Fast Start calls to be translated to Slow Start when this H.323 interface is chosen as the outgoing interface. If this parameter is **enabled**, **call-start-fast** has to remain disabled. The default value is **disabled**. Valid values are:

- enabled | disabled

If you leave this parameter set to **disabled**, the outgoing call will be set up in the same mode as the incoming call.

8. **media-profiles**—Enter the list of media profiles that you want to use when translating Slow Start calls to Fast Start. This information is used to open logical channels for the outgoing call.

If you enter the name of a media profile that does not exist, the Oracle Communications Session Border Controller will not perform translation. If you leave this parameter empty, the Oracle Communications Session Border Controller will not perform translation.

## H.235 Encryption

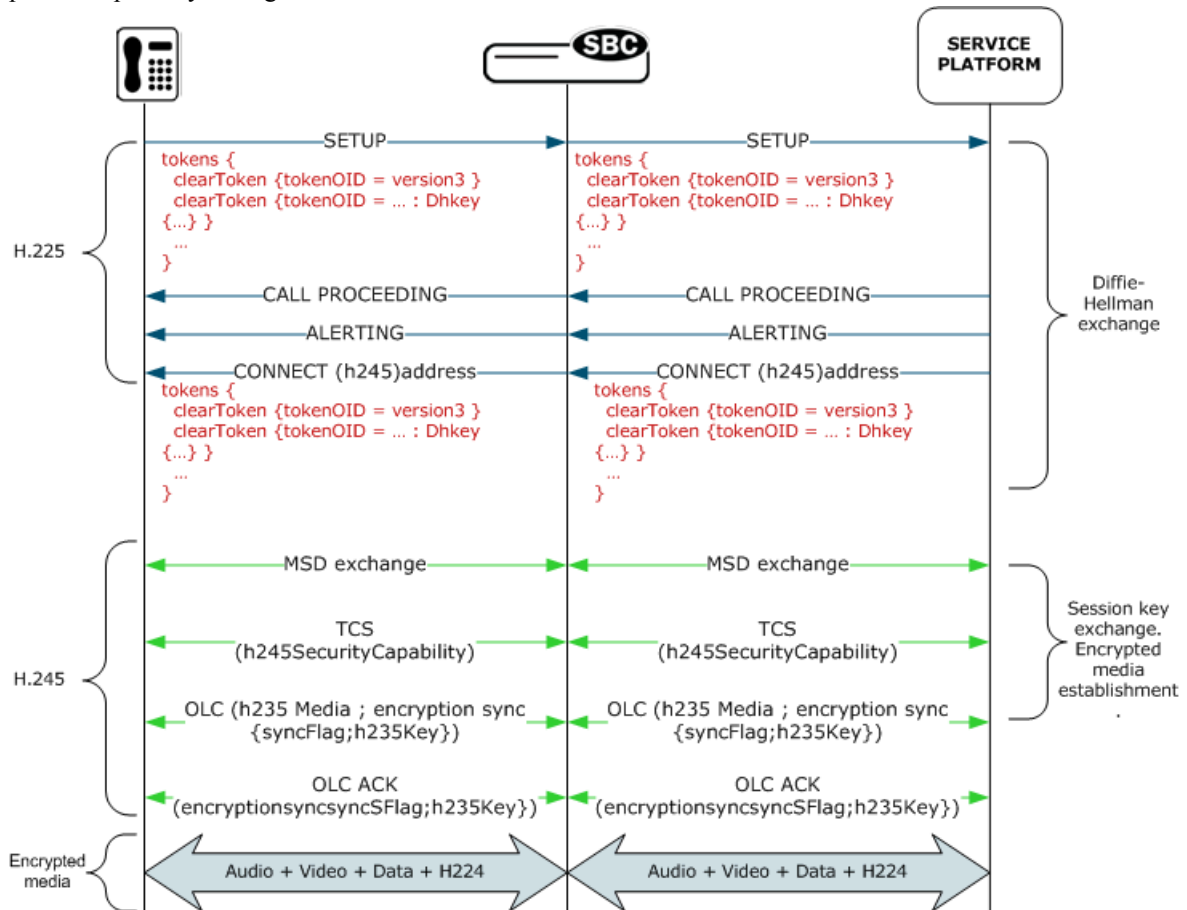
Following the ITU-T H.235 encryption standard, the Oracle Communications Session Border Controller allows media (audio, video, and data) media that has already been encrypted by endpoints to pass through it, thereby supporting videoconferencing applications where media confidentiality is key. The ITU-T standard provides a profile with key management using Diffie-Hellman keys and the specification of an encryption algorithm.

Specifically, the Oracle Communications Session Border Controller permits the following:

- H.225 Setup and connect—The tokens parameter and its subfields in H.225 Setup and Connect message to pass transparently through the Oracle Communications Session Border Controller
- H.245 Terminal CapabilitySet—The H.245 TerminalCapabilitySet messages to pass transparently through the Oracle Communications Session Border Controller, including:
  - Audio, video, and data capabilities

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- The h235SecurityCapability capability
- H.245 OpenLogicalChannel and OpenLogicalChannelAck—OLC messages with dataType h235Media to pass transparently through the Oracle Communications Session Border Controller; to accomplish this, the Oracle Communications Session Border Controller uses the mediaType subfield instead of the dataType field when the dataType is h235Media. The encryptionSync parameter and its subfields found in OLC and OLCAck messages to pass transparently through the Oracle Communications Session Border Controller.



You do not need to follow special configuration steps to enable this functionality; it works automatically.

## RFC 2833 DTMF Interworking

This section explains the Oracle Communications Session Border Controller's support of transporting Dual Tone Multi-Frequency (DTMF) in Real-Time Transport Protocol (RTP) packets (as described in RFC 2833) to H.245 User Input Indication (UII).

Multimedia devices and applications must exchange user-input DTMF information end-to-end over IP networks. The Oracle Communications Session Border Controller provides the interworking capabilities required to interconnect networks that use different signaling protocols. Also, the Oracle Communications Session Border Controller provides DTMF translation to communicate DTMF across network boundaries.

The Oracle Communications Session Border Controller supports RFC 2833 to H.245 UII translation for H.323-to-H.323 calls, when one side is a version 4 H.323 device requiring RFC-2833 DTMF event packets, and the other side is a pre-version 4 H.323 device that only uses H.245 UII.

### About RFC 2833

RFC 2833 specifies a way of encoding DTMF signaling in RTP streams. It does not encode the audio of the tone itself, instead a signal indicates the tone is being sent. RFC 2833 defines how to carry DTMF events in RTP packets. It defines a payload format for carrying DTMF digits used when a gateway detects DTMF on the incoming messages and sends the RTP payload instead of regular audio packets.

### **About H.245 UII**

H.245 provides a capability exchange functionality to allow the negotiation of capabilities and to identify a set of features common to both endpoints. The media and data flows are organized in logical channels. H.245 provides logical channel signaling to allow logical channel open/close and parameter exchange operations. The H.245 signaling protocol is reliable, which ensures that the DTMF tones will be delivered.

H.245 User Input Indication (UII) plays a key role in all the services that require user interaction. For video messaging, typical uses of UII include selection of user preferences, message recording and retrieval, and typical mailbox management functions. H.245 UII provides two levels of UII, alphanumeric and signal.

### **About 2833 to H.245 UII Interworking**

The Oracle Communications Session Border Controller provides 2833 to H.245-UII interworking by checking 2833-enabled RTP streams for packets matching the payload type number for 2833. It then sends the captured packet to the host for processing and translation to H.245 UII messages. A H.245 UII message received by the Oracle Communications Session Border Controller is translated to 2833 packets and inserted into the appropriate RTP stream.

### **About DTMF Transfer**

DTMF transfer is the communication of DTMF across network boundaries. It is widely used in applications such as interactive voice response (IVR) and calling card applications.

The multiple ways to convey DTMF information for packet-based communications include:

- In-band audio: DTMF digit waveforms are encoded the same as voice packets. This method is unreliable for compressed codecs such as G.729 and G.723
- Out-of-band signaling events:

H.245 defines out-of-band signaling events (UII) for transmitting DTMF information. The H.245 signal or H.245 alphanumeric methods separate DTMF digits from the voice stream and send them through the H.245 signaling channel instead of through the RTP channel. The tones are transported in H.245 UII messages.

All H.323 version 2 compliant systems are required to support the H.245 alphanumeric method, while support of the H.245 signal method is optional.

- RTP named telephony events (NTE): uses NTE to relay DTMF tones, which provides a standardized means of transporting DTMF tones in RTP packets according to section 3 of RFC 2833.

Of the three RTP payload formats available, the Oracle Communications Session Border Controller supports RTP NTE.

RFC 2833 defines the format of NTE RTP packets used to transport DTMF digits, hookflash, and other telephony events between two peer endpoints. With the NTE method, the endpoints perform per-call negotiation of the DTMF transfer method. They also negotiate to determine the payload type value for the NTE RTP packets.

The NTE payload takes the place of codec data in a standard RTP packet. The payload type number field of the RTP packet header identifies the contents as 2833 NTE. The payload type number is negotiated per call. The local device sends the payload type number to use for 2833 telephone event packets using a SDP or H.245 Terminal Capability Set (TCS), which tells the other side what payload type number to use when sending the named event packets to the local device. Most devices use payload type number 101 for 2833 packets, although no default is specified in the standard.

The 2833 packet's RTP header also makes use of the timestamp field. Because events often last longer than the 2833 packets sending interval, the timestamp of the first 2833 packet an event represents the beginning reference time for subsequent 2833 packets for that same event. For events that span multiple RTP packets, the RTP timestamp identifies the beginning of the event. As a result, several RTP packets might carry the same timestamp.

See RFC 2833 and draft-ietf-avt-rfc2833bis-07.txt for more information.

### **Preferred and Transparent 2833**

To support preferred (signaled) 2833 and transparent 2833, the Oracle Communications Session Border Controller provides 2833 detection and generation (if necessary) when the endpoint signals support for 2833.

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- Preferred: the Oracle Communications Session Border Controller only generates and detects 2833 for endpoints if they negotiate support for 2833 through signaling
- Transparent: the Oracle Communications Session Border Controller behaves as it has prior to this release, offering and answering based on end-to-end signaling and transparently relaying 2833

### Preferred 2833 Support

If one side of the call, or a session agent, is configured for preferred 2833, the Oracle Communications Session Border Controller only generates and detects 2833 for endpoints if they signal support for 2833. The Oracle Communications Session Border Controller will offer 2833 in the TCS SDP, even if the originating caller did not.

- When the Oracle Communications Session Border Controller manages calls originating from a preferred source going to a preferred target, it:

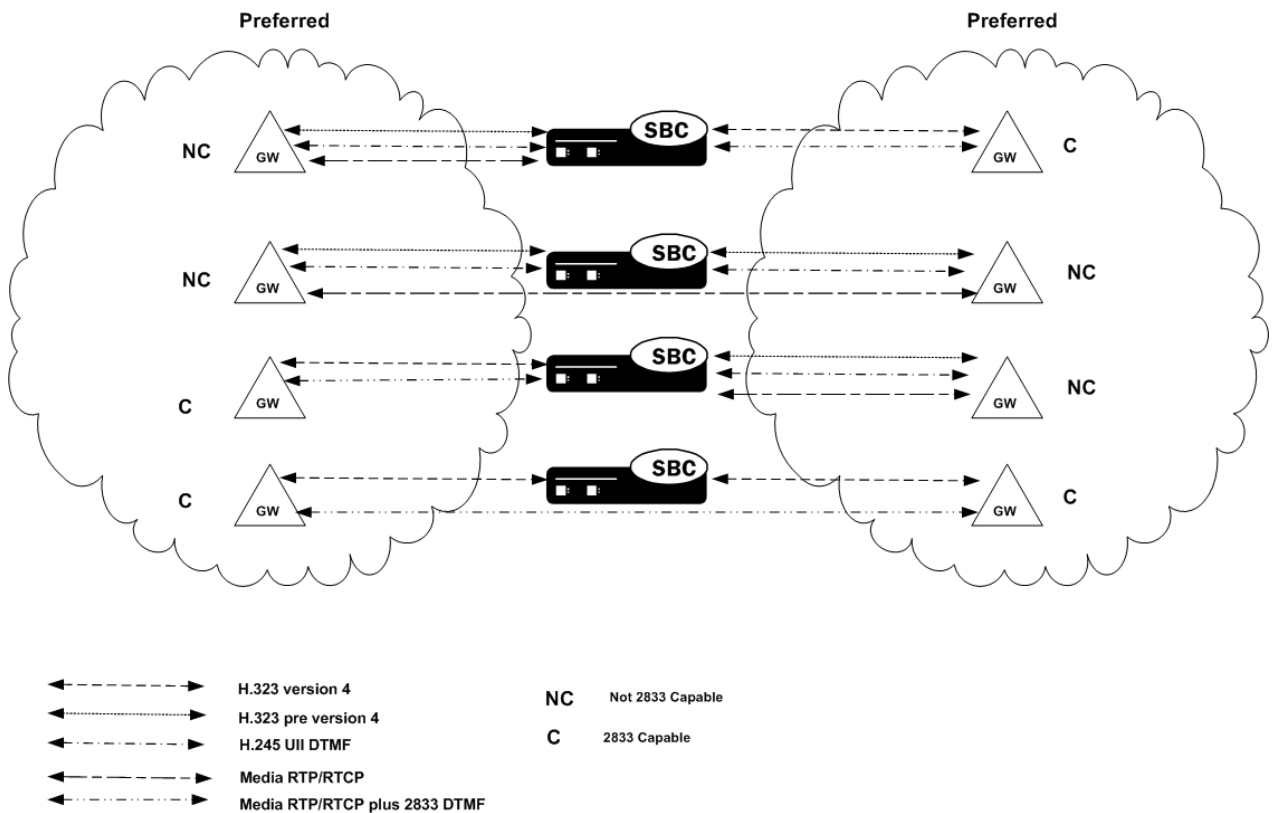
Performs 2833 translation for an endpoint when the originating side requests 2833 but the target does not negotiate 2833

Allows 2833 to pass through if the originating side and target of the call are configured as preferred and negotiate 2833

- When the Oracle Communications Session Border Controller manages calls originating from a preferred source going to a transparent target, it:

Performs 2833 translation when the originating side requests 2833 but the target is configured as transparent and does not negotiate 2833.

Allows 2833 to pass through if the originating side and the target of the call are configured as transparent and negotiate 2833. The Oracle Communications Session Border Controller does not perform active translation because both ends support 2833.



### Transparent 2833 Support

The default configuration of the Oracle Communications Session Border Controller for H.323 is transparent 2833. The Oracle Communications Session Border Controller passes on the offered capabilities to the next-hop signaling element. If the next-hop endpoint is for a transparent 2833 target, typical capability negotiation determines the DTMF



## H.323 Signaling Services

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If the Oracle Communications Session Border Controller is the answering device, configured for preferred support, and the calling device sends 2833, the Oracle Communications Session Border Controller accepts the 2833 regardless of the next-hop's DTMF capabilities. The received dynamic RTP payload type is used for detecting 2833 packets, while the response dynamic payload type is used for generating 2833 packets.

The Oracle Communications Session Border Controller supports:

- RFC-2833 audio telephony events in the version 4 H.323/H.245 TCS as the H.323 calling gateway, when the Oracle Communications Session Border Controller calls an H.323 endpoint configured for preferred RFC 2833 support. The Oracle Communications Session Border Controller sends 2833 to the called party regardless of whether the calling party sends it.
- H.245 UII and RFC-2833 packets sent at the same time, to the same endpoint, even if only half of the call is being provided 2833 support by the Oracle Communications Session Border Controller.

If one half of the call supports H.245 UII, and the other half is being provided 2833 translation by the system, the Oracle Communications Session Border Controller can also forward the H.245 UII it receives to the 2833 endpoint. For example, when the signaling goes through a gatekeeper or third party call control, sending the H.245 UII in the signaling path allows those devices to learn the DTMF digits pressed.

### Signal and Alpha Type Support

The Oracle Communications Session Border Controller supports:

- H.245 signal and alpha type UII in the H.323/H.245 TCS as the H.323 calling gateway when the:  
Oracle Communications Session Border Controller calls an H.323 endpoint configured for transparent 2833 support

calling endpoint's target is configured as preferred

If the originating preferred side also sends 2833, the Oracle Communications Session Border Controller forwards it to the transparent side. The Oracle Communications Session Border Controller sends signal and alpha UII support to the called party regardless of whether the calling party sends it, if the call originates from a preferred side to a transparent side.

- H.245 alphanumeric UII for DTMF for H.323 endpoints that do not signal 2833 or contain explicit H.245 UII capability, for stacks configured for transparent 2833 support.

When the other half of the call is an H.323 endpoint of a stack configured for preferred 2833, the Oracle Communications Session Border Controller translates incoming H.245 UII on the transparent side, to 2833 packets on the preferred side, and vice versa. If the other half of the call is an H.323 endpoint of a transparent stack, the Oracle Communications Session Border Controller relays the H.245 UII messages.

- H.245 signal type UII for DTMF for H.323 endpoints that do not signal 2833, but do signal explicit H.245 UII capability, for stacks configured for transparent 2833 support.

When the other half of the call is an H.323 endpoint of a stack configured for preferred 2833, the Oracle Communications Session Border Controller translates incoming H.245 signaled UII on the transparent side, to 2833 packets on the preferred side, and vice versa. If the other half of the call is an H.323 endpoint of a transparent stack, the Oracle Communications Session Border Controller relays the H.245 UII messages if both sides support it.

### H.323 Endpoints

Because there are different H.323 endpoints based on different versions of H.323, the DTMF can be either be transferred out-of-band as UII or in-band using RFC 2833. Most H.323 endpoints:

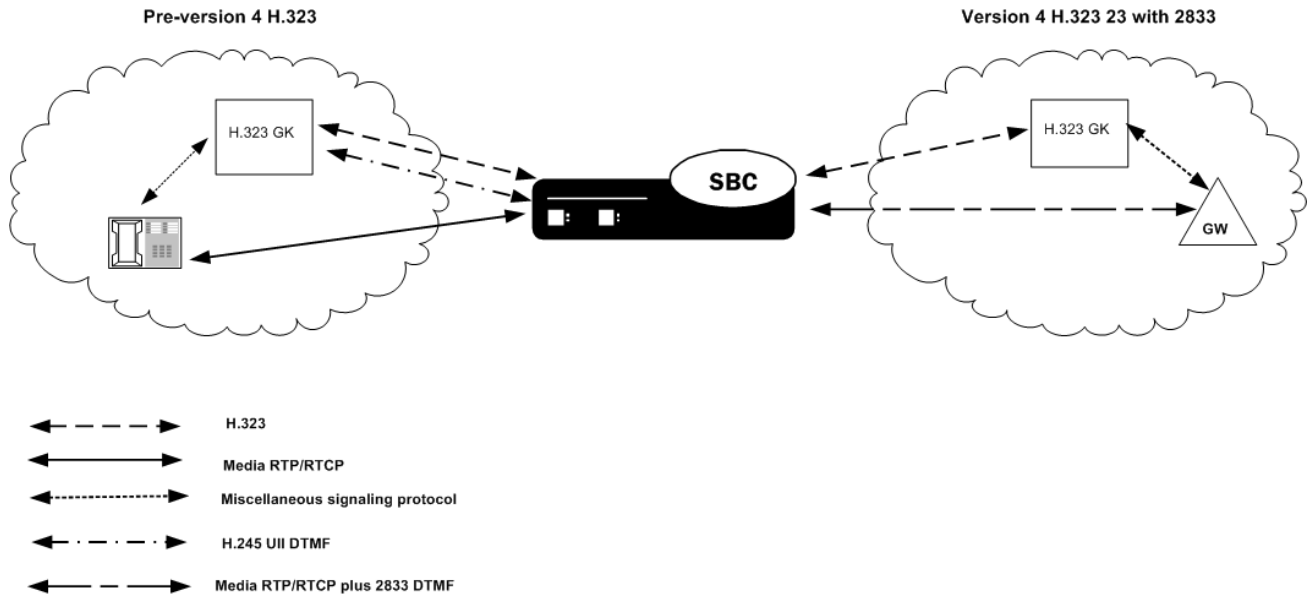
- version 4 and above support RFC 2833
- version 2 and pre-version 4 support UII-Signal
- version 1 and pre-version 2 support UII-Alphanumeric

## Translating H.245 UII to 2833 for H.323 Calls

A majority of H.323 endpoints are not version 4 H.323 compliant and do not support RFC 2833 for DTMF transfer. However, some networks include version 4 H.323 devices that require the DTMF events to be signaled in 2833 packets. Network-based version 4 H.323 gateways use RFC 2833 instead of H.245 UII. (Version 4 H.323 devices should support H.245 UII.)

The Oracle Communications Session Border Controller translates 2833 to H.245 UII for H.323-to-H.323 calls when one side is a version 4 H.323 device requiring RFC-2833 DTMF event packets, and the other side is a pre-version 4 H.323 device which only uses H.245 UII.

The Oracle Communications Session Border Controller can translate H.245 UII to RFC2833 and back, based on the admin configuration and H.245 TCS exchanges. This translation enables DTMF to work end-to-end.



## RFC 2833 Mode Configuration

To configure RFC 2833 mode:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the media-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
```

4. Type **h323-stacks** and press Enter.

```
ACMEPACKET(h323)# h323-stacks
ACMEPACKET(h323-stack)#
```

From this point, you can configure H.323 stack parameters. To view all H.323 stack parameters, enter a **?** at the system prompt.

5. **rfc2833-mode**—Set the RFC2833 mode. The default value is **transparent**. The valid values are:

- **transparent**:—The Oracle Communications Session Border Controller and H.323 stack behave exactly the same way as before and the 2833 or UII negotiation is transparent to the Oracle Communications Session Border Controller.

## H.323 Signaling Services

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- **preferred**—The H323 stack uses 2833 for DTMF transfer, which it signals in its TCS. However, the remote H323 endpoint makes the decision. If the endpoint supports 2833, 2833 is used. If not, the H.323 stack reverts back to using UII. You configure the payload format by configuring the h323-config element.

### RFC 2833 Payload Configuration

To configure the RFC 2833 payload in preferred mode:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **h323** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # h323
```

From this point, you can configure global H.323 parameters. To view all H.323 configuration parameters, enter a **?** at the system prompt.

4. **rfc2833-payload**—Enter a number that indicates the payload type the Oracle Communications Session Border Controller will use for RFC 2833 packets while interworking 2833 and UII. The default value is **101**. The valid range is:
  - Minimum—96
  - Maximum—127

You configure session agents with:

- payload type the Oracle Communications Session Border Controller wants to use for RFC 2833 packets while interworking 2833 and UII.

The default value for this attribute is **0**. When this value is zero, the global rfc2833-payload configured in the h323-configuration element will be used instead. For SIP session agents, the payload defined in the SIP interface is used, if the SIP interface is configured with the preferred RFC 2833 mode.

- 2833 mode

A value of transparent or preferred for the session agent's 2833 mode will override any configuration in the h323-stack configuration element.

### RFC 2833 SA Configuration

To configure session agents:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # session-agent
```

```
ACMEPACKET (session-agent) #
```

4. **rfc2833-mode**—Set the RFC 2833 mode you want the session agent to use. The default is **none**. The valid values are:
  - **none**—2833 to UII interworking is based on the H.323 stack configuration.
  - **transparent**:—The 2833 or UII negotiation is transparent to the Oracle Communications Session Border Controller. This overrides the H.323 stack configuration, even if the stack is configured for preferred mode.



- **preferred:**—2833 for DTMF transfer is preferred, which is signaled in the TCS. If the endpoint supports 2833, 2833 is used. If not, the H.323 stack configured as preferred will revert back to using UII. This overrides any configuration in the h323-stack even if the stack is configured for transparent mode.
5. **rfc2833-payload**—Enter a number that indicates the payload type the session agent will use for RFC 2833 packets while interworking 2833 and UII. The default value is 0. The valid range is:
- Minimum—0, 96
  - Maximum—127

## H.323 Registration Proxy

The Oracle Communications Session Border Controller provides a registration proxy feature that allows a gatekeeper to authenticate a registration before accepting it. This feature is key when two factors are present: authentication is required, and an RRQ from an endpoint includes a token and/or cryptographic token. If authentication for that endpoint is to work, the Oracle Communications Session Border Controller must forward the registration requests received from the endpoint to the gatekeeper separately. When you do not use the H.323 registration proxy, the Oracle Communications Session Border Controller combines all registrations received from H.323 endpoints into a single RRQ and sends it to the gatekeeper. Using the H.323 registration proxy, you can configure the Oracle Communications Session Border Controller to use separate forwarding.

When registration requests are forwarded separately, each RRQ must have a unique CSA. This means that the Oracle Communications Session Border Controller must perform a one-to-one translation of the CSA in the incoming RRQ to a distinct transport address. The translated address replaces the endpoint's CSA in the outgoing RRQ. Then the Oracle Communications Session Border Controller must listen for incoming calls that arrive at this translated transport address for the registered endpoint.

## H.235 Authentication Transparency

When operating in this mode, H.235 authentication tokens (cryptotokens) in RAS messages proxied through the Oracle Communications Session Border Controller are passed through transparently.

For applications where Oracle Communications Session Border Controller is between H.323 gateways and a network hosted gatekeeper, the H.235 cryptotokens are passed through unmodified in RAS messages: RRQs, ARQs, and DRQs. This feature allows for secure gateway authentication.

## Unique CSA Per Registered Gateway

When operating in this mode, each CSA is mapped to a registered gateway for call routing. The core gatekeeper does not support additive registrations, so a different CSA must be used for each unique registration that goes to the gatekeeper. The gatekeeper does not overwrite previously registered aliases. Also, since the gatekeeper initiates calls to an endpoint on the CSA specified in the RRQ, the Oracle Communications Session Border Controller must listen on the assigned address for incoming calls to that client as long as the client is registered.

## Virtual Call Signaling Address

You can configure the Oracle Communications Session Border Controller with:

- A TCP port range for Q.931—Q.931 ports that are frontend ports handled by a real backend socket, and are therefore virtual
- ATCP port range for separate H.245 TCP connections—Actual sockets that the Oracle Communications Session Border Controller handles separately

Virtual call signaling address is an H.323 call signaling address that is registered with a gatekeeper, but does not have a corresponding listening socket in the Oracle Communications Session Border Controller. Using the virtual call signaling address means that numerous network transport addresses do not need to be allocated.

Virtual call signaling addresses work by attaching a range of TCP server ports to a single listening TCP socket. After a connection is accepted, the accepting socket created by the server socket operated normally, as though it were created by the server socket that listens on the same transport address as the destination of the arriving packet.

## H.323 Signaling Services

To use virtual call signaling addresses, you specify a Q.931 port range from which the Oracle Communications Session Border Controller can allocate ports. This port range is associated with the virtual call signal IPv4 address you specify. To bind dynamic TCP connections to a port within a port range, you configure a dynamic H.245 port range. The dynamic H.245 port range refers to the separate TCP connection for H.245 that takes place when tunneling is not being used. This enables the Oracle Communications Session Border Controller to select the port to which the TCP socket is bound. These two port ranges cannot overlap.

When a new RRQ has to be forwarded to the gatekeeper, the Oracle Communications Session Border Controller caches the registration and then forwards a modified copy of the RRQ. The Oracle Communications Session Border Controller allocates a virtual call signal address on the gateway stack and uses it to replace the CSA of the registering endpoint in the forwarded RRQ.

### Virtual RAS Address

The Oracle Communications Session Border Controller also allocates a virtual RAS address for each endpoint registration. Before forwarding an RRQ from an endpoint, the Oracle Communications Session Border Controller replaces the RAS address of the registering endpoint with the virtual RAS address on the gateway interface.

### RAS Message Proxy

When the Oracle Communications Session Border Controller's registration proxy feature is configured, RAS messages to and from endpoints are forwarded, except for the following: GRQ, GCF, GRJ, IRQ, IRR, IACK, and INACK. If the system receives a valid GRQ on the RAS port of the gatekeeper stack that supports H.323 registration, it responds with a GCF message. Otherwise, it sends a GRJ message.

If the gateway interface receives IRR or IRQ messages, the Oracle Communications Session Border Controller attempts to respond based on the information about the call, and does not forward the messages.


Other RAS messages are forwarded after some modifications:

- Translating the transport address
- Deleting fields that the Oracle Communications Session Border Controller does not support

### About Setting Port Ranges

When you configure the H.323 registration proxy feature, you set the Q.931 port range and the dynamic H.245 port range for H.245 connections. If you configure a Q.931 port range, you must also configure a dynamic H.245 port range.

These port ranges cannot overlap because of TCP ports must be unique. The dynamic H.245 port range is used to allocate a real TCP socket, but the Q.931 port range allocates a virtual call signaling address that does not have an associated listening TCP socket.

 **Note:** You should choose these sockets with future Oracle Communications Session Border Controller features about security in mind because future development will support performing admission control based on these port ranges. You will be able to set up filtering rules to allow only inbound packets to configured port ranges.

The following table shows how the Q.931 and dynamic H.245 port ranges work. If you set the start port of 1024 and the number of ports to 1024, you will have configured a port range that starts at 1024 and ends at 2047. So the final port in the range is the start port number added to the number of ports, minus 1. Remember that you cannot overlap the Q.931 and dynamic H.245 port ranges. Notice that the higher the number of the start ports, the fewer ranges of ports you have remaining from which to choose.

Number of Ports	Start Port	n
1024	1024 * n	1-63
2048	2048 * n	1-31
4096	4096 * n	1-15
8192	8192 * n	1-7

Number of Ports	Start Port	n
16384	16384 * n	1-3
32768	32768 * n	1

### H.323 Registration Proxy Configuration

In the CLI, the parameters that apply to this feature are:

q931-start-port	Starting port number for port range used for Q.931 call signalling
q931-number-ports	Number of ports in port range used for Q.931 call signalling
dynamic-start-port	Starting port number for port range used for dynamic TCP connections
dynamic-number-ports	Number of ports in port range used for dynamic TCP connections

To configure the H.323 registration proxy:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
```

4. Type **h323-stack** and press Enter.

```
ACMEPACKET(h323)# h323-stacks
ACMEPACKET(h323-stack)#
```

5. **q931-start-port**—Enter the number where you want the Q.931 port range to start. The default value is **0**. Valid values are:

- 0 | 1024 | 2048 | 4096 | 8192 | 16384 | 32768

6. **q931-number-ports**—Enter the number of ports to be included in the Q.931 port range to use for the call signalling address forwarded in the RRQ. The default value is **0**. Valid values are:

- 0 | 1024 | 2048 | 4096 | 8192 | 16384 | 32768



**Note:** If you have enabled process registration for this H.323 interface, this value must be set to zero because the interface is a gatekeeper that does not support the virtual call signaling address feature.

7. **dynamic-start-port**—Enter the number where you want the dynamic H.245 port range to start. The default value is **0**. Valid values are:

- 0 | 1024 | 2048 | 4096 | 8192 | 16384 | 32768

8. **dynamic-number-ports**—Enter the number of ports to be included in the Q.931 port range to use for the call signalling address forwarded in the RRQ. The default value is **0**. Valid values are:

- 0 | 1024 | 2048 | 4096 | 8192 | 16384 | 32768

### H.323 Registration Caching

The Oracle Communications Session Border Controller can cache and proxy an H.225 RRQ between an H.323 endpoint and a gatekeeper. Registration caching has two benefits:

- It allows the aggregation of RRQs sent to a gatekeeper stack and proxies those requests through the gateway stack. If the external gatekeeper associated with the outbound (gateway) interface does not support additive registration, then the Oracle Communications Session Border Controller consolidates the requests by placing them all in the same packet. Otherwise, additive registration is used on the outbound (gateway) interface.

## H.323 Signaling Services

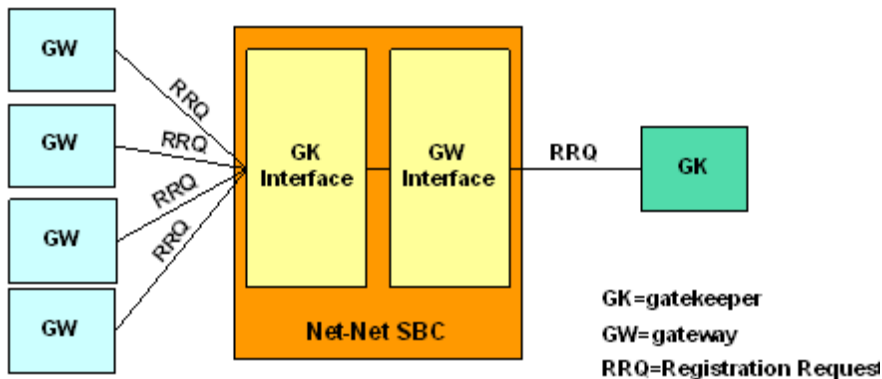
- It allows the gatekeeper stack to use the registration information to route calls from other realms to the endpoints in its realm.

For registration caching, you need to configure at least two H.323 interfaces:

- One gatekeeper interface to receive registrations
- One gateway interface to proxy registrations

The Oracle Communications Session Border Controller caches all successful registrations, using the cache to route calls back to the associated endpoint.

The following diagram shows how RRQs flow during registration caching.



### Caveats for Registration Caching

This feature has the following caveats:

- If a gateway stack receives a URQ message from the gatekeeper, it confirms the request with an UCF message. It flushes all registration caching for that stack. However, the Oracle Communications Session Border Controller does not send URQs to the registered endpoints.
- The Oracle Communications Session Border Controller must be rebooted so that the gateway interface can rediscover the gatekeeper under the following circumstances:

Automatic gateway discovery is turned on for the gateway interface by setting the automatic gateway discovery parameter to enabled.

### Configuration Requirements

For the Oracle Communications Session Border Controller to determine where to route an RRQ, either the associated stack parameter or the gatekeeper identifier field is used.

First, the Oracle Communications Session Border Controller uses the associated interface (assoc-stack) of the gatekeeper interface to find the interface for the outgoing RRQ. If you do not configure an associated interface and the incoming RRQ has a gatekeeperIdentifier field, the Oracle Communications Session Border Controller finds a configured gateway interface with a matching gk-identifier field and use it as the outgoing interface. If the incoming RRQ does not have a gatekeeperIdentifier field and the gatekeeper interface has a configured gatekeeper identifier, the Oracle Communications Session Border Controller finds a gateway interface with a gatekeeper identifier that matches the one set for the gatekeeper interface and then use it as the outgoing interface. If an outgoing interface cannot be determined, the Oracle Communications Session Border Controller rejects the RRQ with the reason discoveryRequired.

A configured H.323 interface can be the gateway interface for more than one gatekeeper interface. If a call is received on the gateway interface, the registration cache will be queried to find a registration matching the call's destination. If a registration is found, the interface on which the registration was received will be used as the outgoing interface for the call.

Subsequent ARQ or URQ messages coming from a registered endpoint will be proxied to the gatekeeper using the outgoing gateway interface established during registration. If a registration is not found, an ARJ or a URJ will be sent to the endpoint originating the ARQ or URQ.

A gatekeeper interface can respond to a GRQ if the GRQ is received on its RAS interface. The Oracle Communications Session Border Controller supports GRQ on a multicast address.

### H.323 Registration Caching Configuration

In the CLI, the parameters that apply to this feature are:

<code>isgateway</code>	Enable the stack to run as a gateway
<code>registration-ttl</code>	Number of seconds before the registration becomes invalid
<code>terminal-alias</code>	List of aliases for terminal
<code>gatekeeper</code>	Gatekeeper's address and port
<code>gk-identifier</code>	Gatekeeper's identifier

To configure the gateway interface parameters for registration caching:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
```

4. Type **h323-stack** and press Enter.

```
ACMEPACKET(h323)# h323-stacks
ACMEPACKET(h323-stack)#
```

5. **isgateway**—Enable H.323 stack functionality as a Gateway. Leave this parameter set to its default, **enabled**, so the H.323 stack runs as a Gateway. When this field is set to **disabled**, the H.323 stack runs as a Gatekeeper proxy. Leave this parameter for the service mode set to its default, **enabled**. Valid values are:

- enabled | disabled

Enabling this parameter ensures that registration with the gatekeeper upon startup. It also ensures that all calls will be preceded by an ARQ to the gatekeeper for admission control.

6. **registration-ttl**—Set the registration expiration parameter to the value of the `timeToLive` field in the RRQ sent to the gatekeeper. The default is **120**. The valid range is:

- Minimum—0
- maximum—4294967295

When the Oracle Communications Session Border Controller receives an RCF from the gatekeeper, it extracts the `timeToLive` field and uses that value as the time interval for keeping the registration of the gateway interface alive. The Oracle Communications Session Border Controller sends a keep-alive RRQ about ten seconds before the registration expires.

The registration expiration you set value should not be too low because some gatekeepers simply accept the `timeToLive` in the RRQ, resulting in a potentially high volume of RRQs.

7. **terminal-alias**—Set this parameter if the gatekeeper requires at least one terminal alias in an RRQ. On startup, the gateway interface registers with the gatekeeper using this terminal alias.

When the Oracle Communications Session Border Controller forwards an RRQ from an endpoint and if the gatekeeper does not support additive registration, the RRQ has the interface's terminal alias, the aliases of the registering endpoint, and other aliases of all registered endpoints. Otherwise, the RRQ only contains the aliases of the registering endpoint.

8. **gatekeeper** and **gk-identifier**—Configure these parameters if you do not want the Oracle Communications Session Border Controller to perform automatic gatekeeper discovery. If the gatekeeper identifier is empty, then the Oracle Communications Session Border Controller learns the gatekeeper identifier from the `gatekeeperIdentifier` field in the GCF.

### Configuring the Gatekeeper Interface for Registration Caching

In the CLI, the parameters that apply to this feature are:

<code>isgateway</code>	Enable the stack to run as a gateway
<code>gatekeeper</code>	Gatekeeper's address and port
<code>gk-identifier</code>	Gatekeeper's identifier
<code>registration-ttl</code>	Number of seconds before the registration becomes invalid

To configure the gatekeeper interface parameters for registration caching:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET (session-router) # h323
```

4. Type **h323-stack** and press Enter.

```
ACMEPACKET (h323) # h323-stacks  
ACMEPACKET (h323-stack) #
```

5. **isgateway**—Set this parameter to **disabled** to run the H.323 stack as a Gatekeeper proxy.
6. **gatekeeper**—Leave this parameter empty.
7. **auto-discovery**—Disable the Automatic Gatekeeper discovery feature upon start-up. Set this parameter to **disabled**.
8. **gk-identifier**—Set this parameter to the identification of the gatekeeper to which RRQs received on this interface must be proxied.
9. **registration-ttl**—Enter the number of seconds to set the timeToLive field in the RCF destined for an endpoint. If you do not configure another value, this timer is set to **120** seconds (default).

This value should not be set too high or too low:

- Setting a value that is too high causes the registration to be alive too long. If an endpoint reboots during this interval and re-registers with the same terminal aliases (but changes its call signaling address), the registration will be rejected with the reason `duplicateAlias`.
- Setting a value that is too low puts an unnecessary load on the Oracle Communications Session Border Controller because it has to handle keep-alive registrations from the endpoint constantly, especially when there are many registered endpoints. If an endpoint does not set the `timeToLive` field in its RRQ, the registration of that endpoint will not expire.

If an endpoint registers again without first unregistering itself (e.g., when it crashes and reboots), the Oracle Communications Session Border Controller rejects the registration using the reason `duplicateAlias`. The Oracle Communications Session Border Controller uses this reason when the endpoint's call signaling address (IP address and port) is changed but its terminal aliases remain the same.

### CLI Registration Caching Configuration Example

In the following example, the H.323 gatekeeper interface (`h323-stack`) is private and the gateway interface (`h323-stack`) is public.

```
h323-config  
state                enabled  
log-level            DEBUG  
response-tmo         4  
connect-tmo          32  
h323-stack  
name                 private  
state                disabled
```

```

realm-id                private
assoc-stack            public
local-ip               192.168.200.99
max-calls              200
max-channels           4
registration-ttl       120
terminal-alias
prefixes
ras-port               1719
auto-gk-discovery     disabled
multicast              0.0.0.0:0
gatekeeper             0.0.0.0:0
gk-identifier
q931-port             1720
alternate-transport
q931-max-calls        200
h245-tunneling        disabled
fs-in-first-msg       disabled
call-start-fast       disabled
call-start-slow       disabled
media-profiles
process-registration   enabled
anonymous-connection  disabled
proxy-mode
filename
h323-stack
name                  public
state                 enabled
isgateway             enabled
realm-id              public
assoc-stack           private
local-ip              192.168.1.99
max-calls              200
max-channels           2
registration-ttl      120
terminal-alias
prefixes
ras-port               1719
auto-gk-discovery     disabled
multicast              0.0.0.0:0
gatekeeper             192.168.1.50:1719
gk-identifier         gk-public.acme.com
q931-port             1720
alternate-transport
q931-max-calls        200
h245-tunneling        disabled
fs-in-first-msg       disabled
call-start-fast       disabled
call-start-slow       disabled
media-profiles
process-registration   disabled
anonymous-connection  disabled
proxy-mode
filename

```

## H.245 Stage

The Oracle Communications Session Border Controller allows you to set the earliest stage in an H.323 call when the Oracle Communications Session Border Controller initiates the procedure to establish an H.245 channel for the call. If you have enabled H.245 tunneling by setting the h245-tunneling parameter to enabled, then you do not need to configure your system for this feature.

The Oracle Communications Session Border Controller initiates the H.245 procedure by either:

## H.323 Signaling Services

- Sending its H.245 address, or
- Creating a TCP connection to an H.245 address that it has received

You can set this parameter to any of the following stages of an H.323 call: setup, proceeding, alerting, connect, early, facility, noh245, and dynamic. With the exception of early, noh245, and dynamic, these values correspond to types of H.225/Q.931 messages. The dynamic value is described in detail in the next section.

When you configure the early value, your Oracle Communications Session Border Controller begins the H.245 procedure at the time the Setup message is sent or received, or when the Connect message is received.

While these values allows for some flexibility about when the H.245 process is started, they are inherently static. All calls in the H.323 stack configuration use the same value, and it cannot be changed from call to call on that stack.

### Dynamic H.245 Stage Support

You can configure your Oracle Communications Session Border Controller for dynamic H.245 support, meaning that the point at which the H.245 process begins can be determined dynamically. To support dynamic H.245, the Oracle Communications Session Border Controller sends its H.245 address in the incoming call when it receives an H.245 address in the outgoing call.

#### Dynamic H.245 Stage for Incoming Calls

When a call comes in on an H.323 interface that you have configured for dynamic H.245 stage support.

The Oracle Communications Session Border Controller includes its H.245 address in the h245Address field of the first H.225/Q.931 message. The Oracle Communications Session Border Controller does this after it receives the first H.225/Q.931 message with an H.245 address in the outgoing call. Based on the first H.225/Q.931 message received by the Oracle Communications Session Border Controller that has an H.245 address, the Oracle Communications Session Border Controller selects the message in which to include the H.245 address as outlined in the table below.

Message Received with H.245 Address	Message Sent with H.245 Address
Call Proceeding	Call Proceeding, Progress, Alerting, Connect or Facility.  The H.245 address is sent in the Call Proceeding message if the system has not sent a Call Proceeding message in the incoming call. This is true only when you enable the Fast Start in first message parameter for the incoming stack; this parameter establishes whether or not Fast Start information must be sent in the first response to a Setup message.  Otherwise, the message in which the H.245 address is sent depends on what message is received after the Call Proceeding message. This is because the Oracle Communications Session Border Controller sends its Call Proceeding message directly after receiving the Setup message.
Progress	Progress
Alerting	Alerting
Connect	Connect
Facility	Facility

When it receives the first H.225/Q.931 message with an H.245 address in the outgoing call, the Oracle Communications Session Border Controller creates a listening socket on the incoming interface. It also includes the socket address and port in the H.245 address of the next H.225/Q.931 message that it sends. If there is no pending H.225/Q.931 message for the Oracle Communications Session Border Controller to send, it instead sends a Facility message with the reason startH245. Then the H.245 channel is established when a TCP connection is made to the listening socket.



For the outgoing leg of a call that came in on the H.323 stack configured for H.245 dynamic stage support, the Oracle Communications Session Border Controller starts establishing the H.245 channel when it receives the first H.225/Q.931 message with H.245 address information. It also starts to establish a TCP connection to the address and port specified in the H.245 address information. The H.245 channel for the outgoing call is established while the H.245 address (h245Address) is sent in the incoming call as described above.

### Dynamic H.245 Stage for Outgoing Calls

This section describes what happens when a message exits the Oracle Communications Session Border Controller on an H.323 stack that you have configured for dynamic H.245 stage support.

When the Oracle Communications Session Border Controller receives the first H.225/Q.931 message that has H.245 address information, it establishes an H.245 channel. The Oracle Communications Session Border Controller initiates a TCP connection to the address and port specified in the H.245 address information.

If the incoming call for the session is also on an H.323 stack with dynamic H.245 configured, the Oracle Communications Session Border Controller starts the H.245 procedure in the incoming call. Otherwise, the system sends its H.245 address in the incoming call based on the H.245 stage support that you have configured.

The process is different when the Oracle Communications Session Border Controller receives a TCS message on the outgoing call before the incoming call reaches its H.245 stage. In this instance, the Oracle Communications Session Border Controller sends a Facility message with the reason startH245 with its H.245 address in order to start the H.245 procedure. The reason is needed in order for the Oracle Communications Session Border Controller to exchange TCS messages with the incoming side of the call.

### H.245 Stage Configuration

To configure H.245 stage support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the media-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
```

4. Type **h323-stacks** and press Enter.

```
ACMEPACKET(h323)# h323-stacks
ACMEPACKET(h323-stacks)#
```

5. **h245-stage**—Set this field to the stage at which the Oracle Communications Session Border Controller transfers the H.245 address to the remote side of the call, or acts on the H.245 address sent by the remote side. The default value is **Connect**. Valid values are:

- Setup | Alerting | Connect | Proceeding | Early | Facility | noh245 | Dynamic

## H.323 HNT

This section explains how H.323 hosted NAT traversal (HNT) works and how to enable this capability on your Oracle Communications Session Border Controller.

The feature enables endpoints behind NATs to originate and terminate calls by resolving the address differences between the NAT and the actual endpoint.

H.323 communication through a NAT becomes an issue when engaging in RAS messaging. While the H.323 standard specifies specific information elements in the RAS messages that indicate the address to which the replies should be sent, these addresses will be behind the NAT and therefore unroutable. The Oracle Communications Session Border Controller solves this problem by sending RAS replies to the layer 3 address from which the associated RAS request was received.

## H.323 Signaling Services

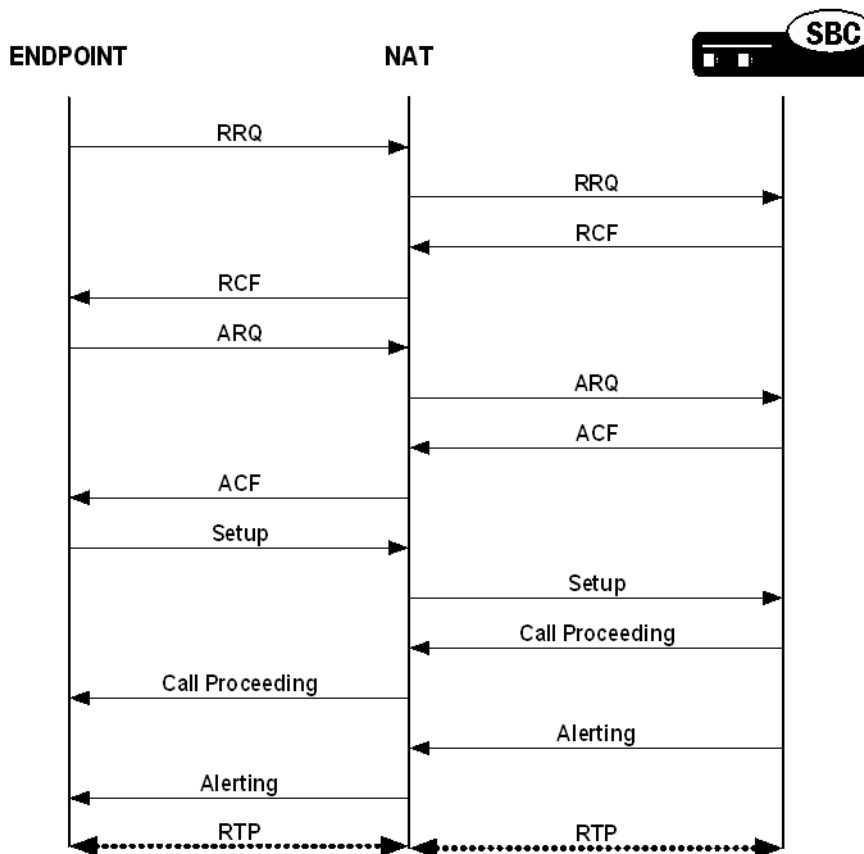
A second issue exists for media channels as the address specified in the H.323 OLC message will be behind the NAT and likewise unroutable. This is resolved by relying on the fact that the forward and reverse channels will utilize the same address and port on the endpoint. By sending media packets to the same address from which the packet are received, media and flow through the NAT.

If you do not use H.323 HNT, the following behavior will occur:

- When an H.323 endpoint is behind a NAT and it registers with a gatekeeper through the Oracle Communications Session Border Controller, the Oracle Communications Session Border Controller tries to send a response back to the endpoint's RAS address rather than to the NAT from which the request was received.
- The same is true for LRQ and IRQ messages because responses without H.323 HNT for outbound sessions, responses were being sent back to the replyAddress or the rasAddress.
- In addition, the Oracle Communications Session Border Controller always induces one-way media because it tries to send the RTP to the media IP address and port it receives in the OLC messages rather than the ephemeral port on the intermediary NAT.

With this ability enabled, however, the Oracle Communications Session Border Controller sends RAS responses back to the address from which the request was received (the NAT). It does not send responses to the endpoint's rasAddress or replyAddress mentioned in the signaling message. The same is true for RTP. With H.323 HNT for outbound sessions enabled, the Oracle Communications Session Border Controller sends RTP to the IP address and port from which it receives the RTP packets (the NAT).

The call flow below illustrates how this feature works:



### Caveats

Keep in mind the following caveats when you are enabling H.323 HNT for outbound sessions on your Oracle Communications Session Border Controller:

- This capability does not apply to calls that require IWF translation between SIP and H.323.

## H.323 HNT Configuration

You can enable this capability for specific H.323 interfaces.

To enable H.323 HNT:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
```

4. Type **h323-stack** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters for the H.323 interface.

```
ACMEPACKET(h323)# h323-stack
```

5. If you are adding this service to a new H.323 interface that you are creating, type **options hnt** (to enable H.323 HNT), and then press Enter.

```
ACMEPACKET(h323-stack)# options hnt
```

6. If you are adding this service to an H.323 interface that already exists, type **select** to select the interface to which you want to add the service. Then use the options command and prepend the option with a plus (+) sign.
  - If you know the name of the interface, you can type the name of the interface at the name: prompt and press Enter.
  - If you do not know the name of the interface, press Enter at the name: prompt. A list of interfaces will appear. Type the number corresponding to the interface you want to modify, and press Enter.
  - **If are adding service to an existing interface and you type options hnt without a plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.**

## H.323 Party Number-E.164 Support

Some H.323 gateways cannot handle partyNumber alias addresses in H.225 messages. The system lets you convert this address type to dialedDigits (E.164). This conversion applies to sourceAddress, destinationAddress, and destExtraCallInfo aliases in Setup messages.

To enable this feature, use the convertPNTtoE164 value in the **options** field of the H.323 stack configuration.

## Signaling Only Operation

When you set the Oracle Communications Session Border Controller to operate in signaling-only mode, it acts like a signaling server. It proxies the call signaling messages between two endpoints. Note, however, that the NOracle Communications Session Border Controller does not function as a RAS proxy; it does not proxy RAS messages.

You have two options for the proxy mode:

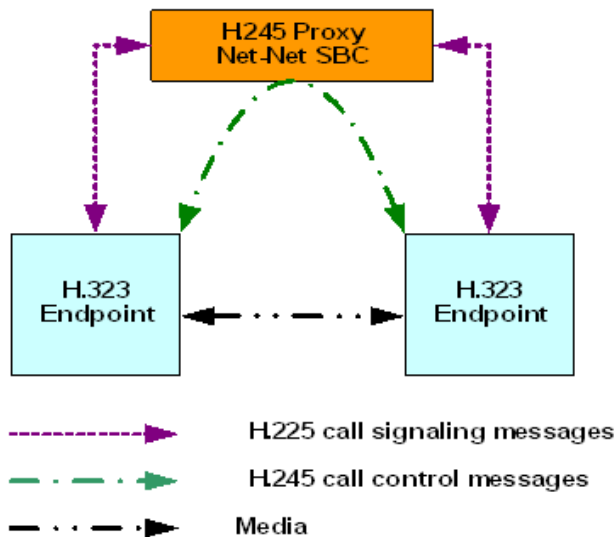
- H.245 proxy mode—The Oracle Communications Session Border Controller handles call signaling (H.225) and call control (H.245) messages.
- H.225 proxy mode—The Oracle Communications Session Border Controller handles call signaling

To use this feature, you need to set the proxy mode parameter in the H.323 interface configuration to H.225 or H.245.

### H.245

When in H.245 proxy mode, the Oracle Communications Session Border Controller proxies or passes through the call signaling (H.225) messages and the call control (H.245) messages. It allows media to flow between the two H.323 endpoints, as shown in the following diagram.

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In some deployments, the media might be treated by a NAT device. When the Oracle Communications Session Border Controller is in H.245 proxy mode, any tunneled H.245 message on the ingress side is tunneled in the egress side. However, if the tunneling is refused on the egress side, a separate H.245 session is established.

H.245 proxy mode support is defined in the following table.

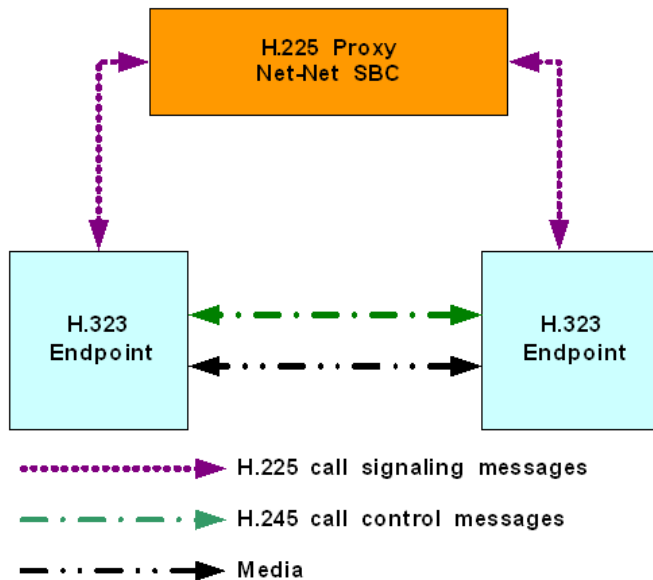
Ingress	Egress
Tunneled	Tunneled
Tunneled	Separate H.245 session
Separate H.245 session	Tunneled
Separate H.245 session	Separate H.245 session

### H.225

When in H.225 proxy mode, the Oracle Communications Session Border Controller only proxies call signaling (H.225 messages). The call control (H.245 messages) and the media associated with the session do not go through the Oracle Communications Session Border Controller. Instead, they flow directly between the two H.323 endpoints.

 **Note:** H.225 proxy mode is only used in specific applications and should not be enabled without consultation from your Acme Packet Systems Engineer.

The following diagram shows the flow.



In certain deployments, the call control message and media are exchanged between the two H.323 endpoints themselves. When the Oracle Communications Session Border Controller is in H.225 proxy mode, any tunneled H.245 message on the ingress side is tunneled in the egress side; this is irrespective of the value configured in the value you set for the `h.245-tunneling` parameter in the H.323 stack configuration.

## Maintenance Proxy Function

The Oracle Communications Session Border Controller supports a maintenance proxy function for H.323 and enhances the way the Oracle Communications Session Border Controller creates unique RAS ports. You can register endpoints through the Oracle Communications Session Border Controller with unique RAS port. You can also set the H.323 interface on the enterprise side to represent enterprise-side endpoints and thereby register on the carrier side.

The maintenance proxy creates a many-to-one association between the enterprise and the carrier side. Interfaces on the enterprise side can be associated with the carrier side interface, which also must be configured to for the maintenance proxy feature.

You configure the maintenance proxy feature by simply setting an option in the H.323 interface configuration.

## Maintenance Proxy Configuration

To configure the maintenance proxy function, you need to set two values in the options parameters for the H.323 interface (`h323-stack`):

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter to access the media-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type `h323` and press Enter.

```
ACMEPACKET(session-router)# h323
```

4. Type `h323-stacks` and press Enter.

```
ACMEPACKET(h323)# h323-stacks
ACMEPACKET(h323-stacks)#
```

5. `options`—Set the options parameter to `maintenanceProxy`.

## Applying TCP Keepalive to the H.323 Interface

To apply these settings individually per H.323 interface:

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---

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET (session-router) # h323
```

4. Type **h323-stack** and press Enter.

```
ACMEPACKET (h323) # h323-stacks  
ACMEPACKET (h323-stack) # tcp-keepalive
```

5. **tcp-keepalive**—Disable this parameter if you do not want the TCP keepalive network parameters to be applied. The default value is **disabled**. Valid values are:

- enabled | disabled

6. Click OK at the bottom of the window to complete configuring TCP keepalives and the maintenance proxy.

## Automatic Gatekeeper Discovery

Available only when the H.323 interface is functioning as a gateway, this feature allows for automatic gatekeeper discovery on start-up.

This feature is based on the Oracle Communications Session Border Controller sending a GRQ to the multicast address of the RAS Multicast Group, which is the device group listening on this address. If you do not configure a multicast address, Oracle Communications Session Border Controller uses the well-known address and port 224.0.1.41:1718 in the address-port combination making up this parameter.

Multicast only functions when the Oracle Communications Session Border Controller is discovering an external gatekeeper. The Oracle Communications Session Border Controller does not respond to multicast gatekeeper queries.

When it receives a GCF message from a gatekeeper, the Oracle Communications Session Border Controller registers with the gatekeeper indicated in the GCF. When it receives an GRJ message that contains optional information about alternative gatekeepers, the Oracle Communications Session Border Controller attempts to register with an alternate.

If you do not use automatic gatekeeper discovery, the Oracle Communications Session Border Controller registers with the gatekeeper you configure in the gatekeeper parameter. In this case, the gatekeeper identifier you configure is included in to the RRQ. No registration takes place if you do not establish automatic gatekeeper discovery or if you do not configure the gatekeeper and its identifier.

## Automatic Gatekeeper Configuration

To configure automatic gatekeeper discovery:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET (session-router) # h323
```

4. Type **h323-stacks** and press Enter.

```
ACMEPACKET (h323) # h323-stacks  
ACMEPACKET (h323-stacks) #
```

5. **auto-gk-discovery**—Enable this parameter to use automatic gatekeeper discovery. The default value is **disabled**. Valid values are:

- enabled | disabled

- 6. **multicast**—Set this parameter to the address and port where the RAS Multicast Group listens. Your entries in this field will be comprised of an IPv4 address and port values separated by a colon. The default value is 0.0.0.0:0.

### H.323 Alternate Routing

You can configure your Oracle Communications Session Border Controller to try more possible routes within given time constraints and number of retries.

#### Without Alternate Routing Enabled

If you do not enable H.323 alternate routing, the Oracle Communications Session Border Controller tries one possible next hop gateway when routing H.323 calls even if the applicable local policy has multiple next hops configured. If that next hop gateway fails (either because it is busy or out of service), the Oracle Communications Session Border Controller relays the failure back to the caller, who hears a busy tone.

In addition, the call will only be routed to the other available next hops if the first one is:

- A session agent that has gone out of service because its constraints have been exceeded
- A session agent that has gone out of service because it failed to respond to a Oracle Communications Session Border Controller Setup request
- A session agent group

#### With Alternate Routing Enabled

When you enable H.323 Alternate Routing on your Oracle Communications Session Border Controller, you enable the use of the other next hops in addition to the first one. The retry, when the other available next hops are used, is transparent to the caller. However, the number of retries is limited by the value you set for the CLI `connect-tmo` parameter, and this feature works only if there is more than one matching local policy next hop. If there is not more than one match, even if that match is a session agent group, then the call is only attempted once and the caller must retry it.

If the Oracle Communications Session Border Controller receives a Release Complete message before it receives an Alerting message, then it will try the next hop if there are multiple matches. When there is no more than one match, or if the timer or number of retries is exceeded, the Oracle Communications Session Border Controller proxies the most recently received Release Complete message back to the caller.

The following table shows the cause codes and release complete reasons, and either of the two actions the Oracle Communications Session Border Controller takes:

- **Recur**—Means that the Oracle Communications Session Border Controller performs (or continues to perform) alternate routing
- **Proxy**—Means that alternate routing stops, and the Oracle Communications Session Border Controller sends a release complete message back to the caller

H.323 Release Complete Reason	Q.850 Cause Code	Action
No Bandwidth	34—No circuit available	Recur
Gatekeeper Resources	47—Resource unavailable	Recur
Unreachable Destination	3—No route to destination	Recur
Destination Rejection	16—Normal call clearing	Proxy
Invalid Revision	88—Incompatible destination	Recur
No Permission	111—Interworking, unspecified	Recur
Unreachable Gatekeeper	38—Network out of order	Recur
Gateway Resources	42—Switching equipment congestion	Recur

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H.323 Release Complete Reason	Q.850 Cause Code	Action
Bad Format Address	28—Invalid number format	Recur
Adaptive Busy	41—Temporary Failure	Recur
In Conference	17—User busy	Proxy
Undefined Reason	31—Normal, unspecified	Recur
Facility Call Deflection	16—Normal, call clearing	Proxy
Security Denied	31—Normal, unspecified	Recur
Called Party Not Registered	20—Subscriber absent	Recur
Caller Not Registered	31—Normal, unspecified	Recur
New Connection Needed	47—Resource Unavailable	Recur
Non Standard Reason	127—Interworking, unspecified	Recur
Replace With Conference Invite	31—Normal, unspecified	Recur
Generic Data Reason	31—Normal, unspecified	Recur
Needed Feature Not Supported	31—Normal, unspecified	Recur
Tunnelled Signaling Rejected	127—Interworking, unspecified	Recur

### H.323 Alternate Routing Configuration

This section describes how to enable H.323 alternate routing. There is a new parameter, and the behavior of the pre-existing **response-tmo** and **connect-tmo** parameters change when you enable this feature on your system.

To enable this feature, you need to set the new **alternate-routing** parameter in the global H.323 configuration to recur. The other option for this parameter is proxy, which means that the Oracle Communications Session Border Controller performs in the way it did prior to Release 4.1, i.e. try only the first matching local policy next hop that it finds.

You configure H.323 alternate for the global H.323 configuration.

To enable H.323 alternate routing:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
```

4. **alternate-routing**—Enable or disable H.323 alternate routing. If you want to keep the pre-4.1 behavior where the Oracle Communications Session Border Controller only tries one matching local policy next hop, leave this parameter set to its default value **proxy**. Valid values are:

- recur | proxy

5. **response-tmo**—Enter the time in seconds for the response time-out (or T303 timer). This is the amount of time allowed to elapse during which the Oracle Communications Session Border Controller should receive a response to its Setup message. If the first response to the Oracle Communications Session Border Controller's Setup is a callProceeding, then the Oracle Communications Session Border Controller should receive an Alerting or Connect message before this timer (now T303\*2) elapses.

The default for this parameter is **4**. The valid range is:



- Minimum—0
  - Maximum—999999999
6. **connect-tmo**—Enter the time in seconds for the connect time-out (or T301 timer). This is the amount of time allowed to elapse during which the Oracle Communications Session Border Controller should receive a Connect message.

For alternate routing, this parameter is also used to limit the number of next hops that are tried and the length of time they are tried in case the first next hop fails. The call needs to be established before this timer expires; the call will fail after maximum of 5 retries.

The default for this parameter is **32**.

- Minimum—0
- Maximum—999999999

## H.323 LRQ Alternate Routing

There are networks where the Oracle Communications Session Border Controller is positioned so that it needs to send an H.225 LRQ request to one signaling entity, and then fall back to another signaling entity when there are no resources available on the first. This might be the case when network contain elements that have limited amounts of channels and/or ports.

To handle situations like this one, the Oracle Communications Session Border Controller can be configured for H.323 LRQ alternate routing.

Without this feature enabled, the Oracle Communications Session Border Controller performs H.323 alternate routing for an H.323 call by finding the alternate route for a local policy when the call setup using H.225/Q.931 fails. Some network configurations, however, require that an LRQ message be sent to a gatekeeper prior to call setup in order to request the destination call signaling address—meaning that the Oracle Communications Session Border Controller will release the call if it does not receive an LCF for that LRQ.

With H.323 LRQ alternate routing enabled, the Oracle Communications Session Border Controller can route the call even when it does not receive the LCF.

When the Oracle Communications Session Border Controller routes an H.323 call using a local policy and the applicable route specifies gatekeeper/session agent as the next hop, the Oracle Communications Session Border Controller must send that gatekeeper an LRQ to request the destination for the call signaling address. After it sends the LRQ, the Oracle Communications Session Border Controller might receive either an LCF or an LRJ, or it might receive no response at all. Upon failure—either the receipt of an LRJ or no response within a timeout period—the Oracle Communications Session Border Controller tries alternate routes (additional routing policies) until the call is either set up or the routing list ends. For each alternate route, if the next hop is a gatekeeper/session agent, the Oracle Communications Session Border Controller sends an LRQ to the gatekeeper in order to request the destination call signaling address. Otherwise, the Oracle Communications Session Border Controller simply sets up the call.

For a designated period of time, the Oracle Communications Session Border Controller waits for the a response to the LRQ from the gatekeeper. This timeout period is configured by setting two options in the global H.323 configuration: **ras-tmo** (number of seconds the Oracle Communications Session Border Controller waits before retransmitting a RAS message; default is 4) and **maxRasRetries** (maximum number of times the Oracle Communications Session Border Controller retransmits the RAS; default is 1). The Oracle Communications Session Border Controller calculates the LRQ timeout period by multiplying the **ras-tmo** by the **maxRasRetries** and adding one (**ras-tmo** x **maxRasRetries** +1).

If an out of service session agent is part of a route, the Oracle Communications Session Border Controller skips it when using alternate routing and uses other routes for the policy.

A session agent might go out of service when it exceeds the maximum number of consecutive transaction timeouts to the maximum number of allowable transaction timeouts. Applicable session agent constrain parameter of note are:

- **trans-timeouts**—Maximum number of allowable transaction timeouts (default is 5)
- **ttr-no-response**—Dictates when the SA (Session Agent) should be put back in service after the SA is taken OOS (Out Of Service) because it did not respond to the Oracle Communications Session Border Controller

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- **in-service-period**—Amount of time that elapses before a session agent is put back in service after the **ttr-no-response** period has passed

By default, the Oracle Communications Session Border Controller continues to send LRQ messages to a session agent even if the session agent has already sent an LRJ. However, you might want to place a session agent out of service when it has sent a certain number of LRJs; doing so allows alternate routing to take place faster, but this is an optional feature.

To configure an LRJ threshold, you add the **max-lrj** value to an H.323 session agent's **options** parameter; instructions for how to set it and the required syntax appear below. If you do not set this option, then the Oracle Communications Session Border Controller will not put session agents out of service for matters related to LRJs.

If you do set this option (to a non-zero value), then the Oracle Communications Session Border Controller keeps a count of the LRJs received from a session agent. When it receives an LCF from a session agent, the Oracle Communications Session Border Controller resets the counter to zero. This count is used internally only and is not accessible through statistics displays.

If a session agent exceeds the maximum number of LRJs and goes out of service, it remains in that state until the **ttr-no-response** period has passed and it has transitioned through the **in-service-period** time. If the **ttr-no-response** period is zero, then the session agent is never put out of service.

### Caveats

The Oracle Communications Session Border Controller does not support H.323 LRQ alternate routing for these scenarios:

- Calls that require translation between SIP and H.323 (IWF calls)
- For pure H.323 calls where the ingress H.323 interface (stack) is associated with another H.323 interface (stack) that has a valid gatekeeper defined; if there is no valid gatekeeper for the egress interface (stack), this feature may apply.

### H.323 LRQ RAS Retransmission Configuration

There is no configuration for H.323 LRQ alternate routing; it is enabled by default. You do, however, need to set the **ras-tmo** and **maxRasRetries** options to set the timeout period.

If you want to set a maximum number of consecutive LRJs to be received from a session agent, you need to add the **max-lrj** value to an H.323 session agent's **options** parameter.

To configure the number of seconds before the Oracle Communications Session Border Controller retransmits a RAS message:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
ACMEPACKET(h323)#
```

4. **options**—Set the options parameter by typing **options**, a Space, the option name **ras-tmo=x** (where X is number of the seconds that the Oracle Communications Session Border Controller waits before retransmitting a RAS message; default is 4) with a plus sign in front of it, and then press Enter.

Set the **maxRasRetries** option in the same way; here, X is the maximum number of times the Oracle Communications Session Border Controller retransmits the RAS; default is 1).

```
ACMEPACKET(h323-stack)# options +ras-tmo=6
ACMEPACKET(h323-stack)# options +maxRasRetries=2
```

If you type **options** and then the option value for either of these entries without the plus sign, you will overwrite any previously configured options. In order to append the new option to the h323 configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

### H.323 LRJ Limit Configuration

To limit the number of LRJs received from an H.323 session agent before putting it out of service:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # session-agent
```

4. Use the ACLI **select** command so that you can work with the session agent configuration to which you want to add this option.

```
ACMEPACKET (session-agent) # select
```

5. **options**—Set the options parameter by typing **options**, a Space, the option name **max-lrj=X** (where X is the maximum number of allowed LRJs) with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (session-agent) # options +max-lrj=3
```

**If you type options max-lrj=X (without the plus sign), you will overwrite any previously configured options. In order to append the new option to the session-agent's options list, you must prepend the new option with a plus sign as shown in the previous example.**

### H.323 CAC Release Mechanism

When an OLC message is sent to the Oracle Communications Session Border Controller and there is insufficient bandwidth available, the Oracle Communications Session Border Controller will reject the incoming OLC. Normally, endpoints decide whether they want to send new OLCs or if they want to release the call. Some endpoints in this situation do neither. When communicating with the last of endpoints, it is desirable for the Oracle Communications Session Border Controller to take action.

The system supports a option in the H.323 interface called `olcRejectTimer`. When this option is enabled and an OLC is rejected, the stack will:

- If there is another media channel open, the Oracle Communications Session Border Controller will behave as if the release mechanism had not been enabled
- If there are no media channels open, the Oracle Communications Session Border Controller starts a timer for 1 second.
  - If the call is released by the endpoint before the timer expires or another OLC is received from the endpoint before the timer expires, the Oracle Communications Session Border Controller stops the timer and follows expected call handling
  - If the timer expires before either of the above responses from the endpoint occur, the Oracle Communications Session Border Controller releases the call.

### H.323 CAC Release Configuration

To enable the H.323 CAC release mechanism:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the media-related configurations.

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```
ACMEPACKET (configure) # session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET (session-router) # h323
```

4. Type **h323-stacks** and press Enter.

```
ACMEPACKET (h323) # h323-stacks
ACMEPACKET (h323-stacks) #
```

5. Use the ACLI **select** command so can add this feature to an existing H.323 interface.

```
ACMEPACKET (h323-stacks) # select
```

6. Set the **options** parameter by typing **options**, a Space, the option name **olcRejectTimer**, and then press Enter.

```
ACMEPACKET(h323-stacks)# options olcRejectTimer
```

7. If you are adding this service to an H.323 interface that already exists, type **select** to select the interface to which you want to add the service. Then use the options command and prepend the option with a plus (+) sign.
  - If you know the name of the interface, you can type the name of the interface at the name: prompt and press Enter.
  - If you do not know the name of the interface, press Enter at the name: prompt. A list of interfaces will appear. Type the number corresponding to the interface you want to modify, and press Enter.
  - **If are adding service to an existing interface and type in the option without a plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign: options +olcRejectTimer.**


## H.323 Per-Realm CAC

Building on the Oracle Communications Session Border Controller's pre-existing call admission control methods, CAC can be performed based on how many minutes are being used by SIP or H.323 calls per-realm for a calendar month.

In the realm configuration, you can now set a value representing the maximum number of minutes to use for SIP and H.323 session using that realm. Although the value you configure is in minutes, the Oracle Communications Session Border Controller performs CAC based on this value to the second. When you use this feature for configurations with nested realms, the parent realm will have the total minutes for all its child realms (i.e., at least the sum of minutes configured for the child realms).

The Oracle Communications Session Border Controller calculates the number of minutes used when a call completes, and counts both call legs for a call that uses the same realm for ingress and egress. The total time attributed to a call is the amount of time between connection (H.323 Connect) and disconnect (H.323 Release Complete), regardless of whether media is released or not; there is no pause for calls being placed on hold.


If the number of minutes is exhausted, the Oracle Communications Session Border Controller rejects calls with a SIP 503 Service Unavailable message (including additional information "monthly minutes exceeded). In the event that the limit is reached mid-call, the Oracle Communications Session Border Controller continues with the call that pushed the realm over its threshold but does not accept new calls. When the limit is exceeded, the Oracle Communications Session Border Controller issues an alarm and sends out a trap including the name of the realm; a trap is also sent when the alarm condition clears.

 **Note:** The Oracle Communications Session Border Controller does not reject GETS/NSEP calls based on monthly minutes CAC.

You can change the value for minutes-based CAC in a realm configuration at any time, though revising the value downward might cause limits to be reached. This value resets to zero (0) at the beginning of every month, and is checkpointed across both system in an HA node. Because this data changes so rapidly, however, the value will not persist across and HA node if both systems undergo simultaneous failure or reboot.

You can use the ACLI `show monthly minutes <realm-id>` command (where `<realm-id>` is the realm identifier of the specific realm for which you want data) to see how many minutes are configured for a realm, how many of those are still available, and how many calls have been rejected due to exceeding the limit.

### Caveats

 **Note:** this feature is not supported for HA nodes running H.323.

## H.323 Per-Realm CAC Configuration

This section shows you how to configure minutes-based CAC for realms and how to display minutes-based CAC data for a specific realm.

Note that setting the new monthly-minutes parameters to zero (0), or leaving it set to its default of 0, disables this feature.

To configure minutes-based CAC:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type `media-manager` and press Enter.

```
Oracle Communications Session Border Controller(configure)# media-manager
Oracle Communications Session Border Controller(media-manager)#
```

3. Type `realm-config` and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. Select the realm where you want to add SIP per user CAC.

```
ACMEPACKET(realm-config)# select
```

5. `monthly-minutes`—Enter the number of minutes allowed during a calendar month in this realm for SIP and H.323 calls. By default, this parameter is set to zero (0), which disabled monthly minutes-based CAC. You can enter a value as high as 71582788.

6. Save and activate your configuration.

Use the ACLI `show monthly-minutes` command to see the following information:

- How many minutes are configured for a realm
- How many of those are still available
- How many calls have been rejected due to exceeding the limit

To view information about SIP per user CAC using the IP address mode: In either User or Superuser mode, type `show monthly-minutes <realm-id>`, a Space, and the IP address for which you want to view data. Then press Enter. The `<realm-id>` is the realm identifier for the realm identifier of the specific realm for which you want data.

```
ACMEPACKET# show monthly-minutes private_realm
```

## H.323 Bearer-Independent Setup

In Release 4.1, the Oracle Communications Session Border Controller supports a new H.323 option that enables H.323 Bearer-Independent Setup (BIS). When enabled, this feature allows exception to slow-start to fast-start conversion on the Oracle Communications Session Border Controller.

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### H.323 BIS Disabled

Unless you enable this feature, the Oracle Communications Session Border Controller performs slow-start to fast-start conversion when a call entering the system as slow-start was routed to an outgoing H.323 interface (stack) with `call-fast-start` set to enabled and there is a list of valid media-profiles in the configuration.

### H.323 BIS Enabled

There are certain cases in access deployments where the slow-start to fast-start conversion should not be applied. This is the case when the Setup message contains the Bearer Capability information element (IE), which signals BIS.

When you enable this feature and the Oracle Communications Session Border Controller receives an incoming Setup message that does not contain a `fastStart` field, the Oracle Communications Session Border Controller checks for the BIS in the incoming Setup before it starts to perform the slow-start to fast-start conversion. If it finds the BIS, then it does not perform the conversion.

This feature can be enabled on a global or a per-interface basis, meaning that you can apply it to your system's entire H.323 configuration or you can enable it only for the interfaces where you want it applied.

### H.323 BIS Global Configuration

This section explains how to add H.323 BIS support to your global H.323 configuration and to specific H.323 interfaces (stacks).

If you set this option on an H.323 interface (stack), you must set it on the interface (stack) that receives the Setup message with BIS in the Bearer Capability IE.

To enable the H.323 BIS feature globally:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter to access the signaling-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type `h323` and press Enter.

```
ACMEPACKET(session-router)# h323
```

4. Type `options +bearerIndSetup` and press Enter.

```
ACMEPACKET(h323-stacks)# options +bearerIndSetup
```

If you type `options bearerIndSetup` without the plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.

### H.323 BIS Specific Configuration

To enable the H.323 BIS feature for a specific H.323 interface:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter to access the signaling-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type `h323` and press Enter.

```
ACMEPACKET(session-router)# h323
```

4. Type `h323-stacks` and press Enter.

```
ACMEPACKET(h323)# h323-stacks
```

```
ACMEPACKET(h323-stacks)#
```

5. Select the H.323 stack to which you want to add H.323 BIS support.

```
ACMEPACKET(h323-stacks) # select  
<name>:
```

For a list of configured H.323 interfaces (stacks), press Enter at the <name>: prompt. Then enter the number corresponding to the interface where you want to apply this feature.

6. Type **options +bearerIndSetup** and press Enter.

```
ACMEPACKET(h323-stacks) # options +bearerIndSetup
```

**If you type options bearerIndSetup** without the plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.

## TOS Marking for H.323 Signaling

You can configure your Oracle Communications Session Border Controller to perform TOS/DiffServ marking for H.323 signaling packets. This feature enables you to mark H.323 signaling packets so that they receive specific treatment from upstream devices. This feature assists in routing because you can configure the TOS byte inserted in the H.323 packet to mark the traffic for certain destinations. For example, you can prevent unauthorized video transmission through an audio-only session.

The Oracle Communications Session Border Controller also performs TOS/DiffServ marking for media.

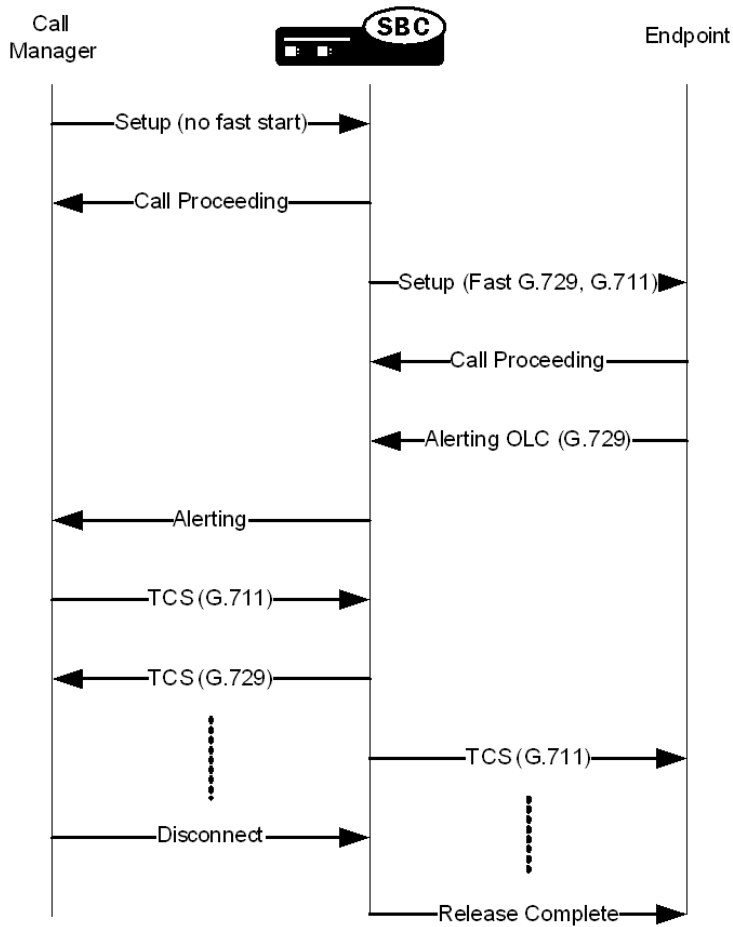
## H.323 Codec Fallback

In the global H.323 configuration, you can enable a parameter that allows the Oracle Communications Session Border Controller to renegotiate—or fallback—to the preferred codec used in an incoming terminal capability set (TCS) from the slow-start side of a slow-start to fast-start H.323 call. When enabled, the Oracle Communications Session Border Controller performs this renegotiation when it detects a mismatch between the codec used in the open logical channel (OLC) opened on the fast-start side of the call, and the codec specified by the slow-start side.

### Codec Fallback Disabled

With codec fallback disabled, the Oracle Communications Session Border Controller opens a channel using the codec specified by the northbound side. Since the call manager had specified another preferred codec, the result is a codec mismatch leading to a dropped call.

The following diagram shows how codec mismatches end in dropped calls.



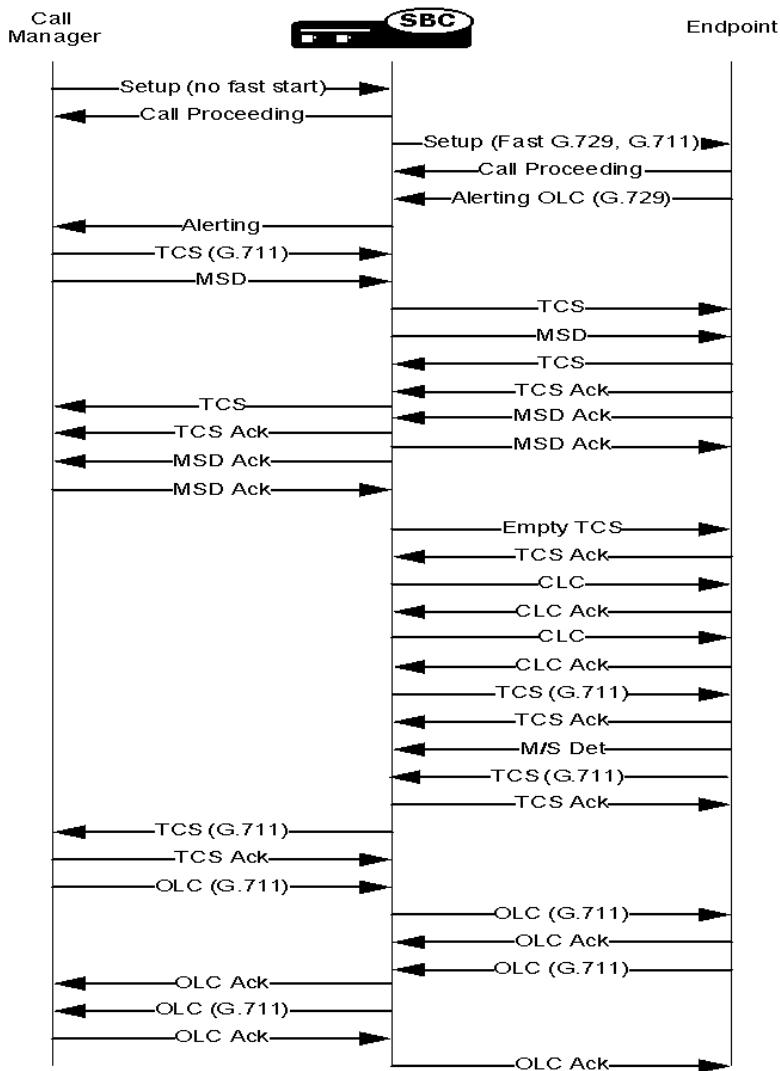
### Codec Fallback Enabled

With H.323 codec fall back enabled, the Oracle Communications Session Border Controller attempts to use the preferred codec that the slow-start side of the call specifies. The Oracle Communications Session Border Controller determines matching based on the incoming TCS from the slow-start side and the OLC on the egress side. If the codecs do not match, the Oracle Communications Session Border Controller sends an empty TCS on the egress side and closes the logical channels on the outgoing side of the call.

To trigger a new capabilities exchange, the Oracle Communications Session Border Controller forwards the TCS from the ingress side of the call to the egress endpoint. Then the TCS from the egress endpoint is propagated to the ingress endpoint, and the logical channels are opened.

The following diagram shows a call scenario using the H.323 codec fallback feature.





### H.323 Codec Fallback Configuration

Note that you configure this feature for your global H.323 configuration, so it has an impact on all H.323 traffic on your system.

To enable H.323 codec fallback:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter. The system prompt will change to let you know that you can configure individual

```
ACMEPACKET(session-router)# h323
```

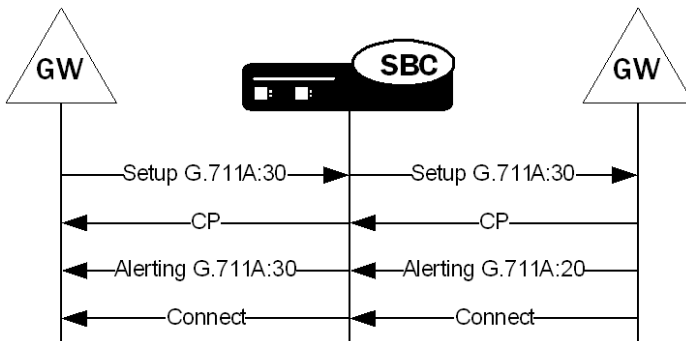
4. **codec-fallback**—Enable or disable the H.323 codec fallback feature. The default value is **disabled**. Valid values are:

- enabled | disabled

### H.323 TCS Media Sample Size Preservation

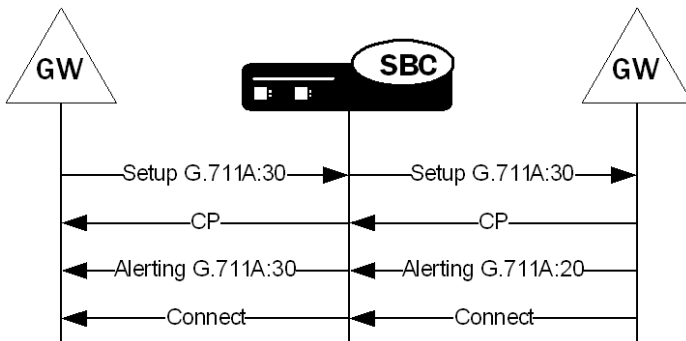
For H.323 fastStart calls, the Oracle Communications Session Border Controller can be configured to preserve the packetization interval from the called gateway if it differs from the one offered in the Setup message the calling gateway sent.

When this feature is disabled and in accordance with the ITU H.323 recommendation, the Oracle Communications Session Border Controller changes the packetization rate to the one used by the calling gateway if the one offered by the called gateway differs. In the following example, this means that the Oracle Communications Session Border Controller replaces the packetization interval of 20 with 30 before it forwards the Alerting message to the calling gateway.



However, not all H.323 elements comply with the ITU recommendation. Since some network elements do modify the packetization rate in the dataType element, this behavior is now configurable.

When you enable media sample size preservation, the Oracle Communications Session Border Controller allows the packetization rate to be modified and forwards on the modified dataType element to the calling gateway. In the following example, you can see that the Oracle Communications Session Border Controller forwards the called gateway's Alerting with the packetization interval of 20 despite the fact that the calling gateway's Setup specified 30.



Note that the calling endpoint might or might not work with the modified dataType.

You can enable this feature for the global H.323 configuration so that it applies to all H.323 fastStart calls, or you can enable it on a per-H.323 interface (stack) basis. When you enable this feature for an individual H.323 interface (stack), the Oracle Communications Session Border Controller performs media sample size preservation for calls egressing on that interface.

#### Media Sample Size Configuration

This section shows you how to configure media sample size preservation for the global H.323 configuration and for an individual H.323 interface (stack).

To enable media sample size preservation for the global H.323 configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
ACMEPACKET(h323)#
```

4. **options**—Set the options parameter by typing **options**, a Space, the option name **forwardFSAcceptedDataType** with a plus sign in front of it. Then press Enter.

```
ACMEPACKET(h323)# options +forwardFSAcceptedDataType
```

If you type **options** and then the option value for either of these entries **without the plus sign**, you will overwrite any previously configured options. In order to append the new option to the **h323** configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

To enable media sample size preservation for an individual H.323 interface:

6. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

7. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

8. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
ACMEPACKET(h323)#
```

9. Type **h323-stacks** and press Enter.

```
ACMEPACKET(h323)# h323-stacks
ACMEPACKET(h323-stack)#
```

If you are adding support for this feature to a pre-existing H.323 interface (stack), then you must select (using the CLI **select** command) the one you want to edit.

10. **options**—Set the options parameter by typing **options**, a Space, the option name **forwardFSAcceptedDataType** with a plus sign in front of it. Then press Enter.

```
ACMEPACKET(h323-stack)# options +forwardFSAcceptedDataType
```

If you type **options** and then the option value for either of these entries **without the plus sign**, you will overwrite any previously configured options. In order to append the new option to the **h323-stack** configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

11. Save and activate your configuration.

## H.323-TCS H.245 Support for H.264 and G722.1

The Oracle Communications Session Border Controller supports the H.264 video codec and the G722.1 audio codec. Especially useful for customer video product offerings in which the Oracle Communications Session Border Controller is deployed, this support further allows the Oracle Communications Session Border Controller to increase ease of use by supporting private addressing. Without this feature enabled (the Oracle Communications Session Border Controller's previous behavior), the Oracle Communications Session Border Controller required deployment for IANA registered IP addresses—despite the fact that IP VPNs allow for RFC 1918 private addressing.

### H.323-TCS Generic Video Configuration

To enable this feature, you need to set up media profile configurations appropriately. Media profiles now allow you to set the configuration either as “generic video” or generic audio.

H.245 provides for defining new capabilities that are described as H.245 generic capabilities (GenericCapability), which the Oracle Communications Session Border Controller now supports using the H.245 GenericCapability

## H.323 Signaling Services

structure. H.264 and G.722.1 are the first codecs the Oracle Communications Session Border Controller offers that use this mechanism.

To set a media profile for generic video support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **media-profile** and press Enter.

```
ACMEPACKET (session-router) # media-profile
ACMEPACKET (media-profile) #
```

4. **name**—Set the name of the generic video media profile to genericVideo. There is no default for this parameter.
5. **media-type**—Set the media type to use for this media profile; for generic video, set this parameter to video.
6. **payload-type**—Set the payload type to use for the generic video media profile.
7. **transport**—Set the transport type to use for the generic video media profile.
8. Complete the rest of the media profile configuration as needed.
9. Save and activate your configuration.

The following is a sample of a generic video media profile configuration:

```
media-profile
  name                genericVideo
  media-type          video
  payload-type        99
  transport            RTP/AVP
  req-bandwidth       0
  frames-per-packet   0
  parameters
  average-rate-limit   0
  sdp-rate-limit-headroom 0
  sdp-bandwidth        disabled
```

## H.323-TCS Generic Audio Configuration

To set a media profile for generic audio support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **media-profile** and press Enter.

```
ACMEPACKET (session-router) # media-profile
ACMEPACKET (media-profile) #
```

4. **name**—Set the name of the generic audio media profile to genericAudio. There is no default for this parameter.
5. **media-type**—Set the media type to use for this media profile; for generic video, set this parameter to audio.
6. **payload-type**—Enter the format in SDP m lines. No payload type number is assigned for newer, dynamic codecs. For RTP/AVP media-profile elements, this field should only be configured when there is a standard payload type number that corresponds to the encoding name. Otherwise, this field should be left blank. This field is used by the system to determine the encoding type when the SDP included with a session identifies the standard payload type on the em line, but does not include an a-rtmpmap entry.

7. **transport**—Set the type of transport protocol to use for the generic audio media profile. The default value is **RTP/AVP**.
  - UPD | RTP/AVP
8. Complete the rest of the media profile configuration as needed.
9. Save and activate your configuration.

The following is a sample of a generic audio media profile configuration:

```
media-profile
  name                genericAudio
  media-type          audio
  payload-type        104
  transport            RTP/AVP
  req-bandwidth       0
  frames-per-packet   0
  parameters
  average-rate-limit   0
  sdp-rate-limit-headroom 0
  sdp-bandwidth       disabled
```

## International Peering with IWF and H.323 Calls

When you do not enable this feature, H.323 calls can default to a National Q.931 Number Type and it is not possible to change it to an International number. This feature allows you to override that behavior by configuring the option **cpnType=X**, where X is an integer that maps to various Q.931 Number Types. When this option is set, Q.931 Number Type for both calling party and called party are updated to the configured value for all outgoing calls on the h323-stack.

The following is a list of possible **cpnType=X** option values for X:

- 0—Unknown public number
- 1—International public number
- 2—National public number
- 3—Specific public network number
- 4—Public subscriber number
- 5—Public abbreviated number
- 6—Private abbreviated number

You configure this feature as an option in the **h323-stack** configuration.

To configure the **cpnType=X** option for H323-H323 calls:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **h323-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# h323-config
ACMEPACKET(h323)#
```

4. Type **h323-stacks** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(h323)# h323-stack
ACMEPACKET(h323-stack)#
```

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5. Set the options parameter by typing **options**, a Space, the option name **cpnType=x** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (h323-stack) # options +cpnType=x
```

If you type **options** without the plus sign, you will overwrite any previously configured options. In order to append the new options to the h323-stack's options list, you must prepend the new option with a plus sign as shown in the previous example.

6. Save and activate your configuration.

## Default OLC Behavior Changed in Upgrade

You can configure the Oracle Communications Session Border Controller to force media profiles in OLC messages when negotiating H.323 calls. This is the default behavior prior to Release S-C6.1.0.

In Release 6.1.0 and forward the default behavior of OLC is to inherit the values received in the signaling message from the remote endpoint.

No license requirements to enable this feature.

To enable forced media profiles in OLC:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **h323-config** and press Enter.

```
ACMEPACKET (session-router) # h323-config
ACMEPACKET (h323-config) #
```

4. **options**—Set the options parameter by typing **options**, a Space, the option-name **forceMediaProfileInOLC** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (h323-config) # options +forceMediaProfileInOLC
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the SIP interface configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

## Options

The options parameter in the global H.323 and H.323 interface configurations allows you to establish the use of specific features; most of those features are customer specific.

You should exercise caution when you apply options because of the fact that many of them are for customer-specific applications. Consult with your Acme Packet systems engineering to find out if using a particular option would be an advantage to you.

Under no circumstance do we recommend that you configure options without Oracle consultation. There is the chance that you could set an option that might harm an otherwise sound configuration.

Some of the options described below are only applicable to IWF calls. However, you need to establish them in your H.323 configuration.

### Global H.323 Options

The following table lists the options that you might want to use in the global H.323 configuration. Again, we recommend that you consult with an Oracle systems engineer about your configuration before using any of these options.

Options	Description
NoDynamicMSD	Oracle Communications Session Border Controller forcefully assumes the “master” role for an outgoing call, and the slave role for an incoming call.
AllowOLCWoMSD	Oracle Communications Session Border Controller sends OLC before master/slave determination is complete.  Causes the Oracle Communications Session Border Controller to be noncompliant with the H.323 recommendation, which does not permit an OLC to be sent prior to MSD completion.
ModifyMediaInAck	Oracle Communications Session Border Controller accepts and propagates changes to media presented in an OLC Ack.  Applies only to Fast Start OLC/OLC Ack messages embedded in H.225/Q.931 messages during call setup.  Causes Oracle Communications Session Border Controller to be noncompliant with the H.323 recommendation, which does not permit media characteristic to be specified in an OLC to be changed in an OLCAck.
MapG729	Oracle Communications Session Border Controller maps H.245 G.729 to SDP G.729 with Annex B and vice versa. Applicable only to IWF calls.
ColonG729	Oracle Communications Session Border Controller uses the : (colon) instead of the = (equal sign) in the media attribute line a=fmtp:18 annexb=yes/no when mapping H.245 G.729 or SDP G.729 with Annex B. Applicable only to IWF calls.
IwfLRQ	Oracle Communications Session Border Controller sends an INVITE (with no SDP) to a redirect server in response to an incoming LRQ received on an H.323 interface. If a 3xx message with a redirected contact header is returned, the Oracle Communications Session Border Controller will send an LCF in response to the LRQ. Otherwise, it will send an LRJ.
NoG729AnnexB	SDP received by the IWF with H.729 and no FMTP will be mapped to G.729 on the H.323 side of the call. Can also be set in the session agent options parameter.
sameT38Port	Oracle Communications Session Border Controller does not allocate separate ports for audio and T.38. Oracle Communications Session Border Controller will send the same audio port in the OLCAck that it sees in a request mode for T.38 and a new OLC for T.38.
pvtStats	Oracle Communications Session Border Controller includes program value tree (PVT) statistics in the show h323d display that are a sum of the PVT statistics for all H.323 interfaces. Used for debugging purposes.
strayARQTimer	Required the syntax "strayARQTimer=x," where x is the number of seconds the system waits before tearing down an unsuccessful call in the case of stray ARQs.
forceMediaProfileInOLC	Reverts to older OLC Behavior.

**H.323 Interface Options**

The following table lists the options that you might want to use in the configuration H.323 interfaces. Again, we recommend that you consult with an Oracle systems engineer about your configuration before using any of these options.

## H.323 Signaling Services

Option	Description
stackAliasWins	Oracle Communications Session Border Controller will replace the sourceAddress of the incoming Setup message with the terminal alias of the egress interface when copying the incoming sourceAddress to the outgoing Setup message.
uniqueRRQRASAddress	Oracle Communications Session Border Controller will generate unique rasAddress for each RRQ that it sends to a gatekeeper in response to an incoming RRQ received on an H.323 interface configured for process registration. The IP address will be the local-ip of the outgoing interface, so the port is the unique portion of the rasAddress.
nonV4AdditiveRRQ	Gatekeeper associated with the H.323 interface support additive registration even though it does not set the additiveRegistration field in the RRQ message. When sending in the additive mode, the H.323 interface only sends with the RRQ new terminal aliases that need to be registered. In non-additive mode, the interface sense all the terminal aliases that have been registered, plus the new aliases.
cachedTerimnalAlias	Oracle Communications Session Border Controller copies the terminal alias(es) of the registered endpoint to the asourceAddress field of the Setup message. Terminal alias(es) are changed after the system successfully processes an RRQ from the endpoint.
proxySrcInfo	Oracle Communications Session Border Controller copies the sourceInfo from the incoming Setup message to the outgoing Setup message. Otherwise, Oracle Communications Session Border Controller uses its own endpointType for the sourceInfo field.
noAliasinRCF	Oracle Communications Session Border Controller does not include any terminal alias in the RCF.
forceH245	Oracle Communications Session Border Controller initiates an H.245 connection after the call is connected. Otherwise, Oracle Communications Session Border Controller listens for an H.245 connection to be initiated by a remote endpoint.
useCPNInRAS	Oracle Communications Session Border Controller uses the calling party number (CPN) IE of the incoming call as the srcInfo of a RAS message sent in the outgoing call (such as an ARQ).
maintenanceProxy	Oracle Communications Session Border Controller registers interfaces on the enterprise side with a gatekeeper on the carrier side, and registers endpoints through the Oracle Communications Session Border Controller with a unique rasAddress. Interfaces on the enterprise side are associated with the carrier interfaces; you set this option on the carrier side.
convertPNTToE164	Oracle Communications Session Border Controller converts the address type partyNumber to dialedDigits (E.164). Conversion applies to sourceAddress, destinationAddress, and destExtraCallInfo aliases in Setup messages.
useCalledPNAsDestInfo	Oracle Communications Session Border Controller uses the H.225 called party number IE as the destinationInfo in ARQ and LRQ requests. Since translation rules can be applied to the Called Party Number, the option enables digit normalization for RAS requests.  When not used, Oracle Communications Session Border Controller derives the destinationInfo field in RAS requests from the DestnationAddress field of the incoming Setup.
waitForIncomingH245	On the incoming leg, the Oracle Communications Session Border Controller does not send out its h245Address, but waits for the calling endpoint to send its H245Address. Applies to the outgoing call led as well: The Oracle Communications



Option	Description
	Session Border Controller does not send out a Facility with startH245 reason and waits for the called endpoint to send its H245Address.
uniqueRRQSrcPort	Enables H.323 RAS Port Mapping. The Oracle Communications Session Border Controller uses the RAS port that it assigned in the rasAddress parameters of an RRQ message as the UDP source port of the outgoing RRQ. Because this feature is linked to the unique RRQ functionality, be aware of the following before you enable the feature: <ul style="list-style-type: none"> <li>Enabling H.323 RAS Port Mapping automatically enables the Oracle Communications Session Border Controller's unique RRQ functionality, eliminating the need for you to configure the latter as a separate option.</li> <li>Enabling the unique RRQ functionality (by setting the uniqueRRQRASAddress option) does not automatically enable H.323 RAS Port Mapping.</li> </ul>
srcCallSignallingPort	Enables use of the Q.931 port value for the port field in the sourceCallSignalAddress parameter in an H.225 Setup message. Useful for customers who configure a separate H.323 interface (stack) on the core side for each external IP-PBX.

## H.323 Stack Monitoring

In releases prior to S-C6.2.0, the Oracle Communications Session Border Controller provides SNMP monitoring of H.323 session agents but not of the H.323 stacks themselves. The H.323 stack/interface configuration now provides a way for you to set alarm thresholds on a per-stack basis. When enabled, this alarm system ties into the max-calls value to send critical, major, or minor alarms when the number of calls approaches the threshold.

Each H.323 stack now has a threshold crossing alert (TCA) where you can set up three severity levels: critical, major, and minor. You can define one severity level or all three for each stack. To prevent the alarm from firing continuously as call volume through the stack varies, each severity level has an has a reset value below the TCA you set. In addition, each threshold value resets when:

- An alarm with a higher severity is triggered, or
- The built-in reset value for the threshold level is 1% less than the parameter value

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## H.323 Stack Monitoring Configuration

This section shows you how to configure H.323 stack monitoring for one H.323 stack configuration. This example shows one instance of the alarm-threshold sub-configuration being established; remember that you can set three—critical, major, and minor. Simply repeat the configuration steps to add more severity levels.

To set up H.323 stack monitoring:

- In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

- Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

- Type **h323** and press Enter to access the global H.323 configuration.

```
ACMEPACKET(session-router)# h323
ACMEPACKET(h323)#
```

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4. Type **h323-stack** and press Enter. If you are adding H.323 stack monitoring to an existing H.323 stack configuration, then remember you must select the stack you want to edit.

```
ACMEPACKET(h323) # h323-stack
ACMEPACKET(h323-stack) #
```

5. Type **alarm-threshold** and press Enter to configure this feature.

```
ACMEPACKET(h323-stack) # alarm-threshold
ACMEPACKET(alarm-threshold) #
```

6. **severity**—Enter the type of severity level for the alarm you want to define. Choose from: **critical**, **major**, or **minor**. This value is required, and defaults to **minor**.
7. **value**—Enter the percentage of the number of calls defined in the **max-calls** parameter that triggers the alarm. For example, if you want to set a minor alarm to fire when the call rate through the stack reaches half the **max-calls** value, enter **50** (meaning 50%). The default value for this parameter is 0, which disables the alarm.

Remember that if the number of calls falls to below 1% of the max-calls threshold you set, the clear trap fires.

8. Save your work. You can see the data related to this feature using the ACLI **display-alarms** and **show h323 stack stack-alarms** commands.

## H.323 Automatic Features

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This section describes H.323 features that are automatically enabled on your Oracle Communications Session Border Controller. You do not have to configure special parameters to turn them on. Even though you do not have to turn these features on, this section describes what they do and how they work.

### Alias Mapping

Alias mapping permits destination addresses to be modified by a gatekeeper.

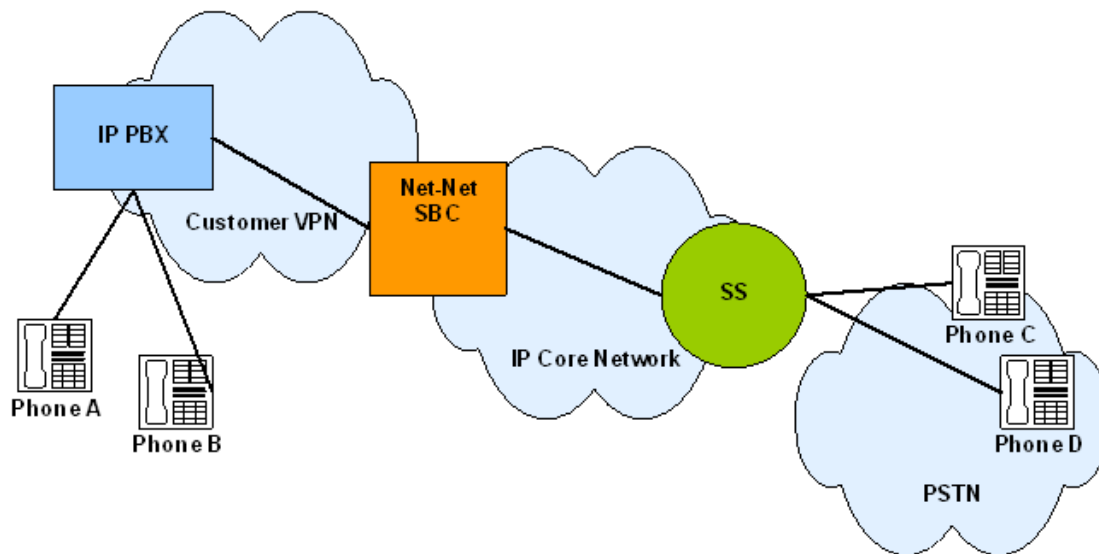
When sending an ARQ or an LRQ message to a gatekeeper, the Oracle Communications Session Border Controller sets the `canMapAlias` field in that message to true. This setting indicates that the Oracle Communications Session Border Controller accepts modified destination information from the gatekeeper. If the resulting ACF or LCF contains `destinationInfo` and/or `destExtraCallInfo` fields, then the Oracle Communications Session Border Controller copies that information respectively to the `destinationAddress` and `destExtraCallInfo` fields of the Setup message. In addition, if the `destinationInfo` is either type `e164` or type `partyNumber`, the Oracle Communications Session Border Controller copies the information into the `calledPartyNumber` information element (IE) of the Setup message, replacing the existing `calledPartyNumber` IE.

You do not need to configure special parameters for this feature; it is enabled automatically.

### Call Hold and Transfer

The Oracle Communications Session Border Controller's H.323 call hold and transfer feature supports consultation in addition to call hold and transfer. This feature uses signaling procedures based on the ITU-T recommendations/H.323 specification for what it calls third party initiated pause and rerouting.

The following diagram shows how the Oracle Communications Session Border Controller is positioned to provide call hold and transfer support for H.323.



### Call Hold and Transfer Basic Call

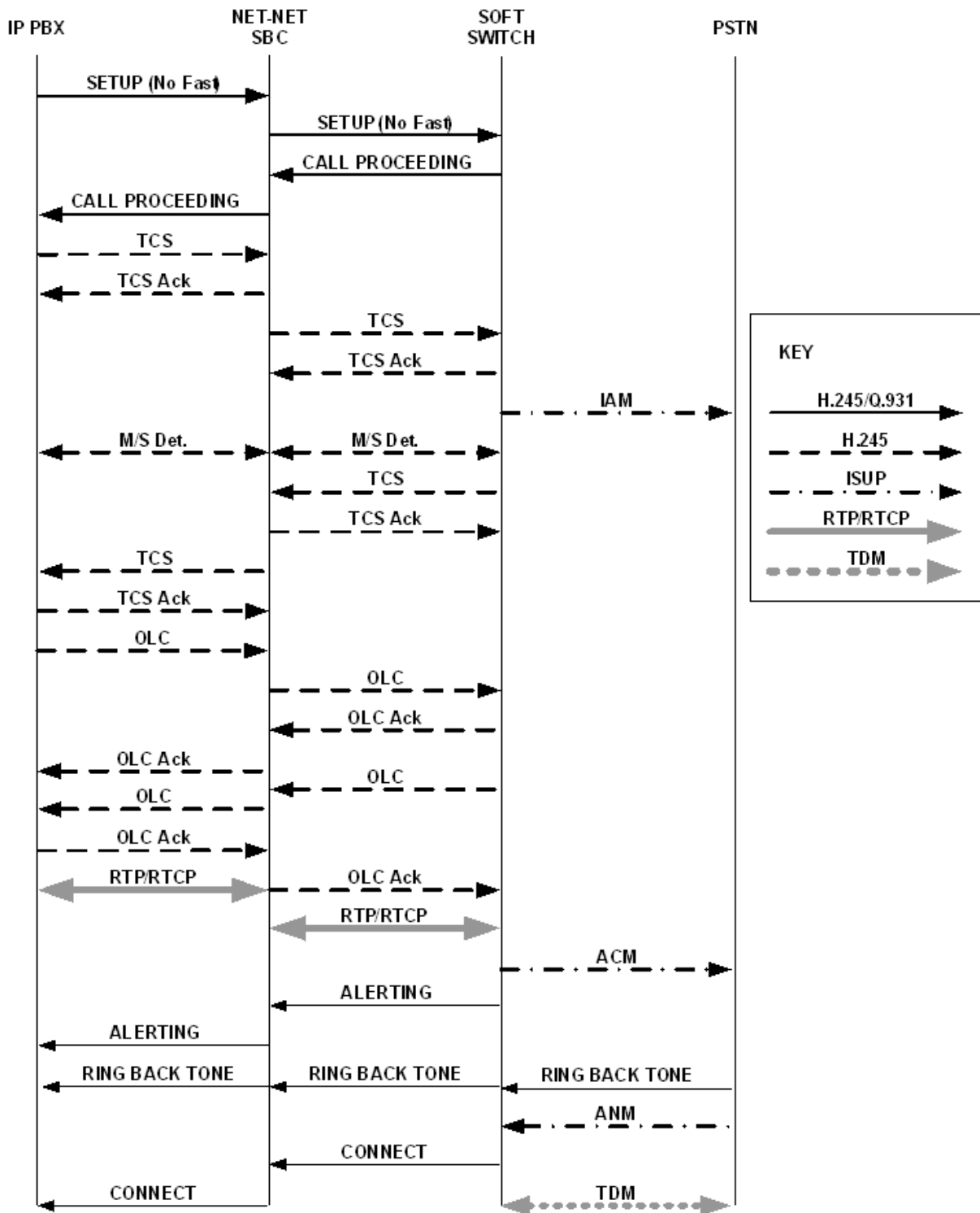
The following diagram show the signaling and media flows between the IP PBX and a softswitch. Note how the Oracle Communications Session Border Controller is position to mediate flows between the two devices.

In the Call Proceeding messages forwarded to the IP PBX, the Oracle Communications Session Border Controller uses a non-zero value to ensure that the IP PBX initiates an H.245 session. A progress indicator does not need to be included if the H.245 address is present in any of the following message types: Alerting, Progress, or Connect.

After the Oracle Communications Session Border Controller receives a Call Proceeding message from the softswitch that contains the H.245 address, the Oracle Communications Session Border Controller sends another Call Proceeding with its own H.245 address.

In the following call flow, the softswitch generates message to the gateway. These messages are:

- Initial Address Message (IAM)
- Address Complete Message (ACM)
- Answer Message (ANM)



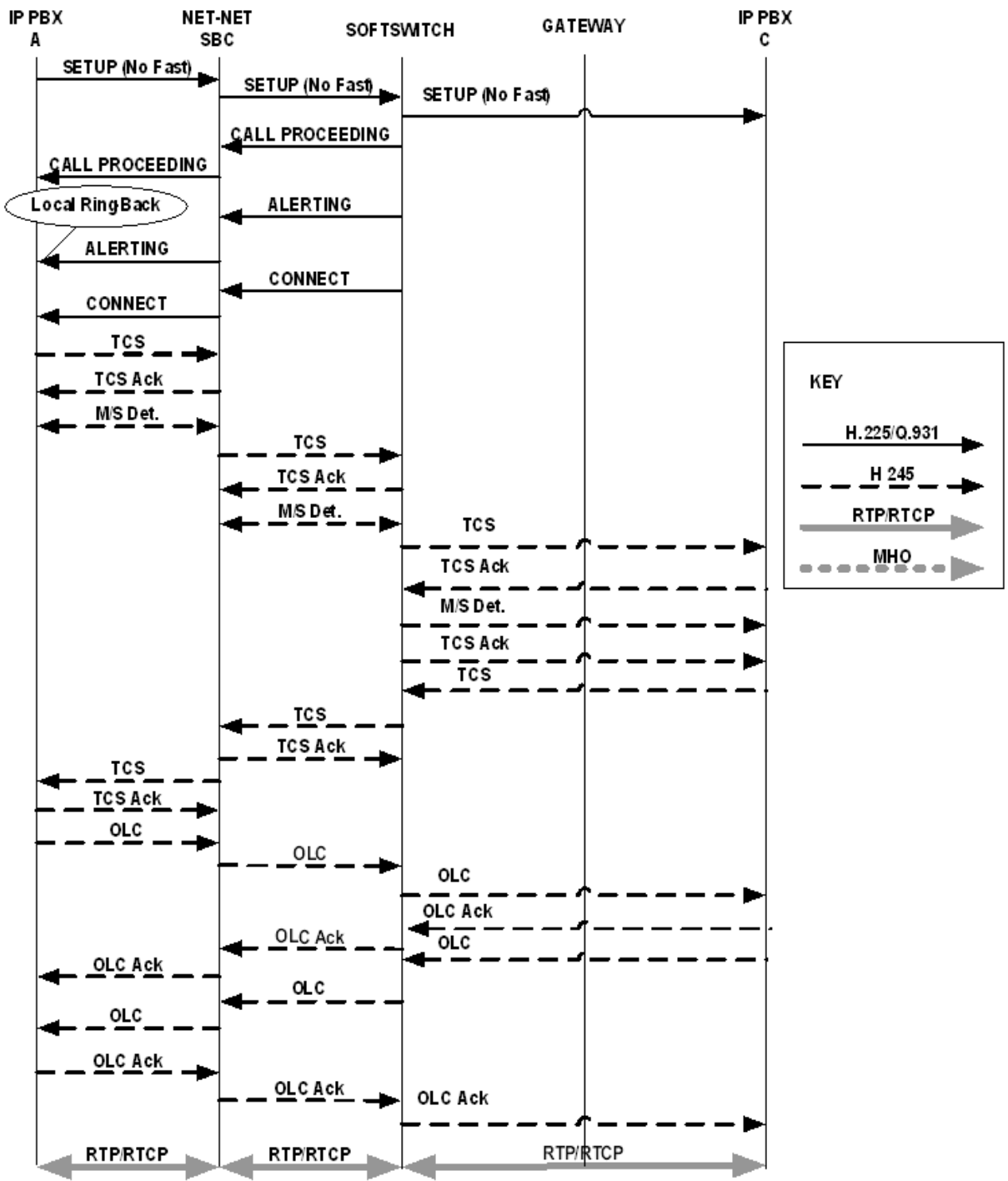
**Call Hold and Transfer Music on Hold**

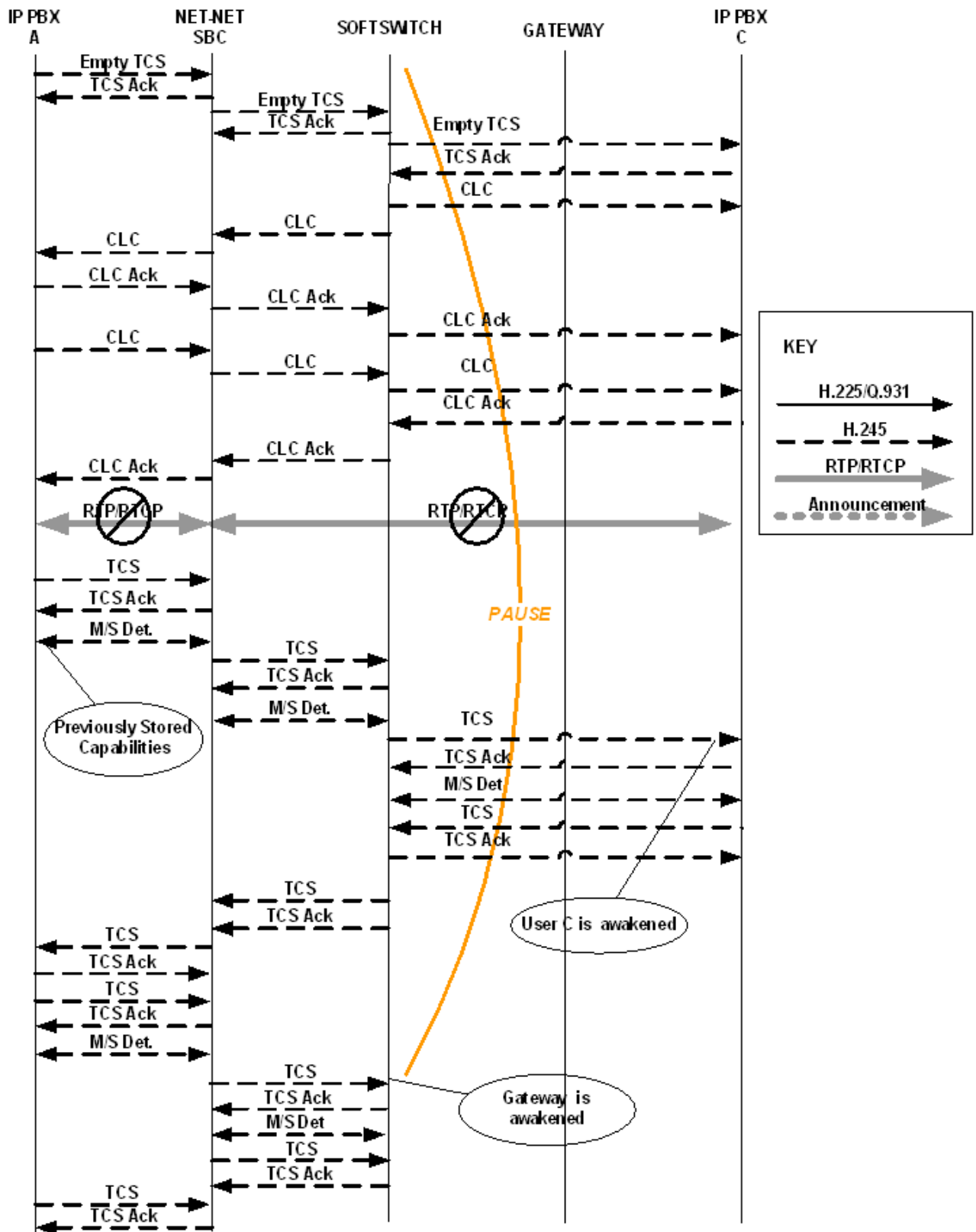
The following diagram begins with the condition that IP PBX A is already connected with a gateway, with the Oracle Communications Session Border Controller and the softswitch positioned between the two.

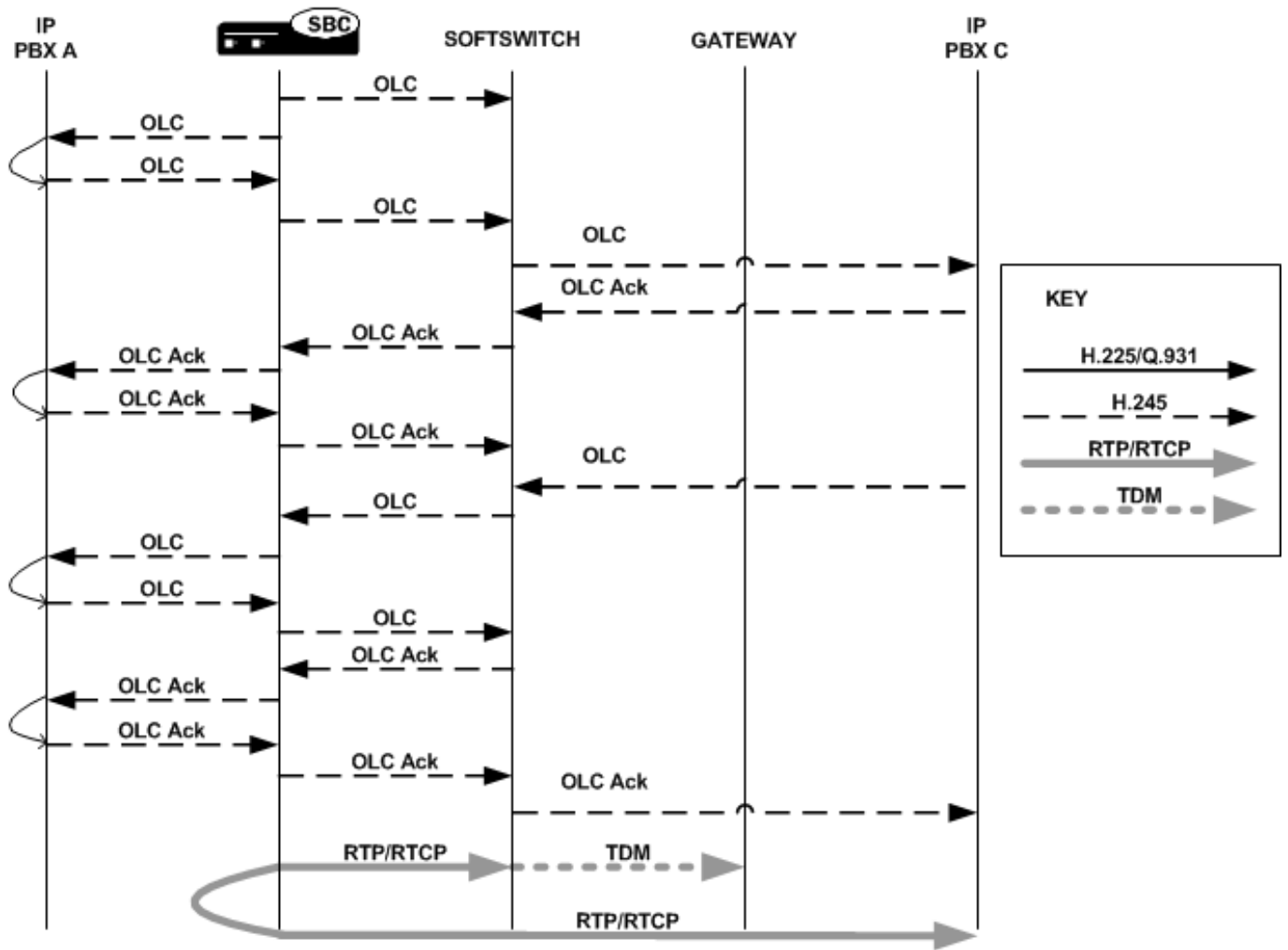
You can see in the call flow where the channels for transporting media are closed, and where the RTP/RTCP is stopped. This creates a pause for the call. With the Oracle Communications Session Border Controller mediating the process, IP PBX A and the softswitch exchange TCS and OLC messages that allow music on hold (MHO) to flow between IP PBX A and the gateway.



### H.323 Signaling Services







### Media Release for SS-FS Calls

When the Oracle Communications Session Border Controller routes a slow-start to fast-start call, it is possible for the same fast-start call to be routed back through the Oracle Communications Session Border Controller making for a hairpin flow. If it does become a hairpin flow, then the Oracle Communications Session Border Controller routes it to its destination as a fast-start to fast-start call. This can result in one-way media if:

- The destination of the hairpin call is in the same realm as the originating slow-start to fast-start call
- The realm reference in the first bullet item is configured to disable in-realm media management
- The called endpoint accepts the proposed fast-start logical channels

The enhancements to the Oracle Communications Session Border Controller's behavior described in this section show how the Oracle Communications Session Border Controller follows additional procedures when setting up a hairpin flow to avoid one-way media when media release occurs.

For H.323 calls, the Oracle Communications Session Border Controller establishes media using the H.245 procedures described in the H.245 ITU-T recommendation: control protocol for multimedia communication. It also uses the Fast Connect procedure defined in the H.323 ITU-T recommendation: packet-based multimedia communication systems.

The latter ITU-T recommendation allows a calling endpoint to send a Setup message that contains a fastStart element, a sequence of OLC structures that describe the calling endpoint's proposed forward/reverse logical channels. If the called endpoint accepts this proposal, then logical channels are established.

When the Oracle Communications Session Border Controller translates a call originating in slow-start to fast-start, it uses a Fast Connect procedure in the outgoing leg by sending an outgoing Setup that includes a fastStart element with one or more OLC structures. But when the Oracle Communications Session Border Controller constructs this



message, it is unaware of whether the call will become hairpinned or if media release will occur. Because it does not yet have this information, the Oracle Communications Session Border Controller sets the Network Address and the TSAP identifier in the OLC structures to the ingress IP address and port of a corresponding media flow allocated for media traveling between the calling and called endpoints. So if the called endpoint accepts the fastStart the Oracle Communications Session Border Controller proposes, the called endpoint would send its media to the Oracle Communications Session Border Controller. After acceptance, the system starts H.245 procedures on the slow-start side of the call to set up logical channels on that side. Then the Oracle Communications Session Border Controller updates the IP address and port of the media flows using OLC and OLCAck messages received from the calling endpoint.

This procedure works well for endpoints that are not in the same realm, or that are in the same realm for which media management is disabled, because each endpoint must send its media through the Oracle Communications Session Border Controller. When the endpoints are in the same realm and when media management is enabled, however, the Oracle Communications Session Border Controller must perform additional steps for media release in slow-start to fast-start calls.

To support media release in slow-start to fast-start calls, the Oracle Communications Session Border Controller performs a hold-and-resume procedure on the fast-start side. After it establishes channels on the slow-start side and if it detects media release being enabled, the Oracle Communications Session Border Controller sends an empty TCS to the fast-start side to put that side on hold. Then the called endpoint closes all the logical channels it previously opened in the Fast Connect procedure and stops transmitting to them. And the Oracle Communications Session Border Controller also closes its logical channels. Once the channels are closed, the Oracle Communications Session Border Controller resumes the call by sending a new, restricted TCS to the fast-start side. The restricted TCS only contains the receive and transmit capabilities of the codec types that the called endpoint accepted in the Fast Connect procedure, and it forces the called endpoint to re-open logical channels of the same codec types accepted in the Fast Connect procedure. Once it receives an OLC from the called endpoint, the Oracle Communications Session Border Controller sends an OLCAck with the Network Address and TSAP identifier for the logical channel from the calling endpoint. Then the Oracle Communications Session Border Controller re-opens logical channels (of the same codec types that it opened in the Fast Connect procedure). If the called endpoint has not changed its Network Address and TSAP identifier for its logical channels, media is re-established after the Oracle Communications Session Border Controller and the called endpoint exit the hold state. The last step is for the Oracle Communications Session Border Controller to re-send the full TCS message from the calling to the called endpoint to inform the called endpoint of the full capabilities of the calling endpoint.

## **Dependencies**

This feature depends on the following assumptions:

- The H.323 endpoint supports the third-party-initiated pause and re-routing feature.
- The H.323 endpoint does not change its Network Address and TSAP identifier when it re-opens the logical channels.
- The H.323 endpoint does not immediately tear down the call when there is not established logical channel in the call.

## **Hold-and-Resume Procedure**

The hold-and-resume procedure has three states:

- **Media Hold**—Starts when the Oracle Communications Session Border Controller sends the empty TCS to the called endpoint to put it on hold.

When it detects media release, the Oracle Communications Session Border Controller puts the called endpoint on hold. It can only do so if it has exchanged the TCS/TCSAck messages and completed master-slave determination with the calling endpoint.

When the Oracle Communications Session Border Controller receives a TCSAck in response to the empty TCS that it sent to the called endpoint, it closes the logical channels it opened as part of the Fast Connect procedure; the called endpoint likewise closes its logical channels. The two then exchange CLC and CLCAck messages, which signals the start of the Media Resume state.

## H.323 Signaling Services

- **Media Resume**—Starts when the Oracle Communications Session Border Controller sends a restricted TCS to resume the call.

The restricted TCS the Oracle Communications Session Border Controller sends contains only the receive/transmit capabilities of the codec types previously accepted by the called endpoint in the Fast Connect procedure. This forces the called endpoint to re-open logical channels of the same codec type that were previously accepted in the Fast Connect procedure.

After sending this TCS, the system is ready (as specified in the ITU-T recommendations) to take part on the master-slave determination (MSD) process. However, the called party and not the Oracle Communications Session Border Controller initiates the MSD if it is required. The MSD is completed if necessary. Alternately, the called endpoint can start to re-open its logical channels. When it receives the first OLC from the called endpoint, the Oracle Communications Session Border Controller also starts to re-open its logical channels.

- **Media Complete**—Starts when all the logical channels that the Oracle Communications Session Border Controller re-opens are acknowledged by the called endpoint.

When it enters the Media Complete state, the Oracle Communications Session Border Controller updates the called endpoint with the full capabilities of the calling endpoint by sending the full TCS.

## H.323 and IWF Call Forwarding

This section describes the Oracle Communications Session Border Controller's H.323 and IWF Call Forwarding feature, which is supported for H.323 calls and for calls initiated in SIP that require interworking to H.323.

### Previous Behavior

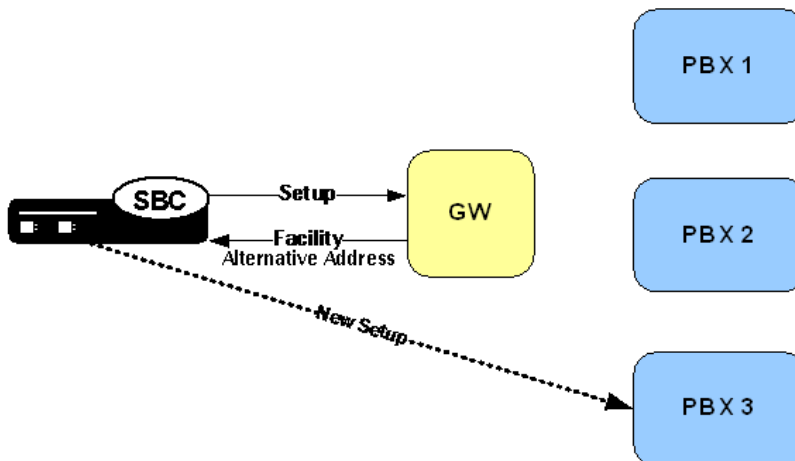
Prior to Release 4.1, the Oracle Communications Session Border Controller did not forward calls when the remote H.323 endpoint sent a Facility message with Call deflection as the reason and an alternate address for forwarding. Instead, it would either:

- Fail to release the initial call and initiate the forwarded call
- Drop the entire call when the remote endpoint for the call tore down the session

### New Behavior

In the diagram below, you can see that the Oracle Communications Session Border Controller sends the initial Setup message to the gateway, and the gateway returns the Facility message with an alternate address for forwarding. Rather than engaging in its former behavior, the Oracle Communications Session Border Controller now releases the call with the gateway and sends a new Setup to the alternate address from the Facility message.

This new Setup up has no effect on the first call leg, which remains connected.



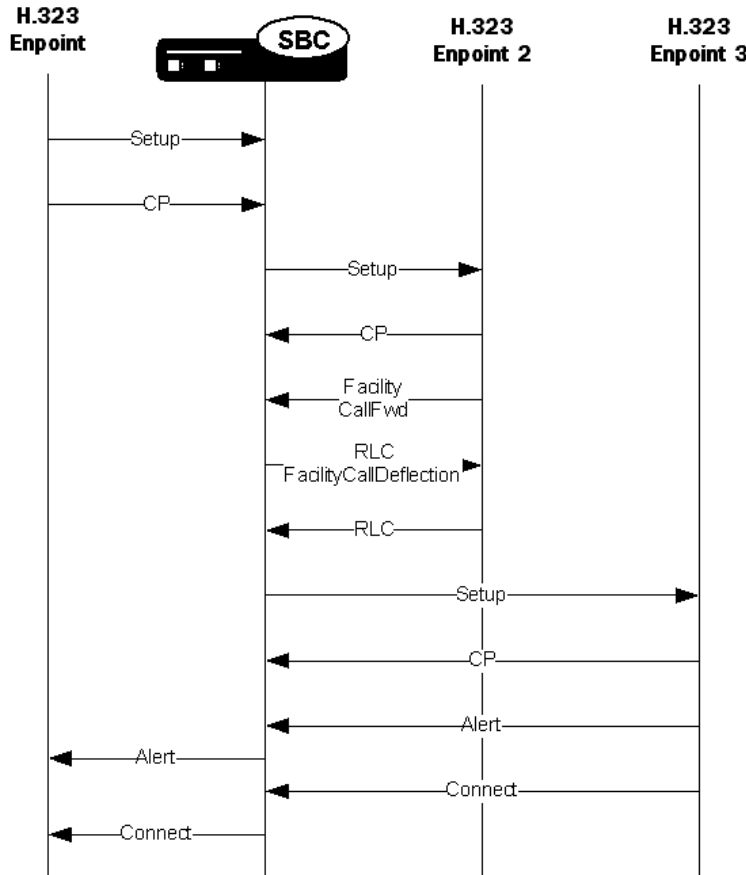
When it receives a Facility message with the reason CallForwarded, the Oracle Communications Session Border Controller looks for an alternate transport address in the Facility's alternativeAddress or alternativeAliasAddress

element. The Oracle Communications Session Border Controller releases the egress call with the reason facilityCallDeflection. Then it takes one of two courses of action:

- If it does not find an alternative address, the Oracle Communications Session Border Controller releases the ingress call (with the reason facilityCallDeflection).
- If it finds an alternative address and the egress call has not been alerted or answered, the Oracle Communications Session Border Controller at this point tries to initiate a new egress call. The Oracle Communications Session Border Controller uses the alternative alias address to populate the calledPartyNumber information element (IE) and the destination address of the new Setup.

### H.323 Sample Call Flow

The following diagram shows how the H.323 Call Forwarding feature works in a purely H.323 environment.



### H.323 NOTIFY Support

To inform another call party of a certain event or communicate information to it, and H.323 entity might send a NOTIFY message. For example, a gateway might send a NOTIFY message to inform the calling party of a display name for a transferee. In previous releases, the Oracle Communications Session Border Controller did not process such a NOTIFY message, blocking the message from reaching its intended destination.


The Oracle Communications Session Border Controller supports the NOTIFY message so that it can pass through and reach its intended destination.

#### Caveats

The Oracle Communications Session Border Controller does not support interworking the NOTIFY message to a SIP message for calls that require interworking between H.323 and SIP; this support is for pure H.323 calls only.

## H.323 H.239 Support for Video+Content

The Oracle Communications Session Border Controller supports multiple media streams for the same payload, generic capabilities, and H.239 generic messages. As a result, these additions broaden the Oracle Communications Session Border Controller’s support for videoconferencing, and free you from have to configure media profiles for H.323 support.

 **Note:** These additions are supported for H.323-H.323 traffic only. These additions do not support SIP-H.323 interworking (IWF), so you still need to configure media profiles for that application.

### Multiple Media Streams with the Same Payload

In releases prior to S-C6.2.0, the Oracle Communications Session Border Controller supports multiple audio-video-data streams only if those streams use different payload types. The Oracle Communications Session Border Controller’s behavior is extended to provide this support as of Release S-C6.2.0. The Oracle Communications Session Border Controller identifies extendedVideoCapability used to establish an additional channel for H.239-compliant endpoints, an OLC that was formerly not supported.

### Support for Generic Capabilities

This feature identifies the OIDs shown in the table below and uses the dynamicPayload type to from the incoming OLC to generate its own OLC. So you no longer need media profiles for: genericAudio, genericVideo, and genericData.

Capability Name	Capability Class	Capability Identifier
H.283	Data protocol	{itu-t (0) recommendation (0) h (8) 283 generic-capabilities (1) 0}
G.722.1	Audio protocol	{itu-t (0) recommendation (0) g (7) 7221 generic-capabilities (1) 0}
G.722.1 Extension	Audio protocol	{itu-t (0) recommendation (0) g (7) 7221 generic-capabilities (1) extension (1) 0}
H.324	Data protocol	{itu-t (0) recommendation (0) h (8) 324 generic-capabilities (1) http (0)}
H.263	Video protocol	{itu-t (0) recommendation (0) h (8) 263 generic-capabilities (1) 0}  Note: Use of this capability to signal H.263 "Profiles and Levels" per Annex X/H.263 should always be accompanied in parallel by the signalling of the same modes in H263VideoCapability. This is necessary to ensure that systems which do not recognize the H.263 generic capabilities continue to interwork with newer systems.
H.224	Data protocol	{itu-t (0) recommendation (0) h (8) 224 generic-capabilities (1) 0}
G.722.2	Audio protocol	{itu-t (0) recommendation (0) g (7) 7222 generic-capabilities (1) 0}
G.726	Audio protocol	{itu-t (0) recommendation (0) g (7) 726 generic-capabilities (1) version2003 (0)}
H.241/H.264	Video protocol	{itu-t (0) recommendation (0) h (8) 241 specificVideoCodecCapabilities (0) h264 (0) generic-capabilities (1)}

Capability Name	Capability Class	Capability Identifier
H.241/H.264	Video protocol	{itu-t(0) recommendation(0) h(8) 241 specificVideoCodecCapabilities(0) h264(0) iPpacketization(0) RFC3984NonInterleaved(1)}
H.241/H.264	Video protocol	{itu-t(0) recommendation(0) h(8) 241 specificVideoCodecCapabilities(0) h264(0) iPpacketization(0) RFC3984Interleaved(2)}

**Support for H.239 Generic Messages**

This section describes the Oracle Communications Session Border Controller’s support for H.239 Generic Messages.

Generic Message	Description
Generic Request	<ul style="list-style-type: none"> <li>flowControlReleaseRequest—Used when a device wants to add a channel toward an MCU that has sent multipointConference, or if the device wants to increase a channel bit rate when the channel is flow-controlled. The message has the channelId, which is the logicalChannelNumber of the channel. The Oracle Communications Session Border Controller proxies this message, replacing the channelId with the logicalChannelNumber of its channel.</li> <li>presentationTokenRequest—Request by the sender to acquire the indicated token. The message has the channelId, which is the logicalChannelNumber of the channel. The Oracle Communications Session Border Controller proxies this message, replacing the channelId with the logicalChannelNumber of its channel.</li> </ul>
Generic Response	<ul style="list-style-type: none"> <li>flowControlReleaseResponse—Sent in response to the flowControlReleaseRequest, either acknowledging or rejecting the request. The “acknowledge” response indicates the far-end device intends to make a best-effort attempt to comply with the request. The exact bit rate requested may not be allocated. The reject response indicates that the far-end device does not intend to comply with the request. The response contains the channelId that was sent in the request. While proxying the response, the Oracle Communications Session Border Controller will replace the channelId with the channelId it received in the request.</li> <li>presentationTokenResponse—Sent in response to the presentationTokenRequest. The response will either confirm or reject the assignment of the indicated token to the sender of the presentationTokenRequest. The response contains the channelId that was received in the request. While proxying the response, the Oracle Communications Session Border Controller will replace the channelId with the channelId it received in the request.</li> </ul>
Generic Command	<ul style="list-style-type: none"> <li>presentationTokenRelease—Sent by the device holding the token in order to relinquish the token. The message has the channelId, which is the logicalChannelNumber of the channel. The Oracle Communications Session Border Controller proxies this message, replacing the channelId with the logicalChannelNumber of its channel.</li> </ul>

## H.323 Signaling Services

Generic Message	Description
Generic Indication	<ul style="list-style-type: none"><li>presentationTokenIndicateOwner—Indicates who owns the token. The message has the channelId, which is the logicalChannelNumber of the channel. The Oracle Communications Session Border Controller proxies this message, replacing the channelId with the logicalChannelNumber of its channel.</li></ul>

### Support for Miscellaneous Indication

An endpoint sends a miscellaneous indication to send (logicalChannelActive) or stop (logicalChannelInactive) live video streams. The message has a channelId, which is the channel's logicalChannelNumber. The Oracle Communications Session Border Controller proxies this message, replacing the channelId with the logicalChannelNumber of its own channel.

## Video Conferencing Support for Polycom Terminals

The Oracle Communications Session Border Controller in a video conferencing environment with Polycom H323 terminals and a Polycom MCU (Multipoint Conferencing Unit) relays H.239/H.245. The Oracle Communications Session Border Controller implements the following messages appropriately:

- Miscellaneous command message with subtype such as multiPointModeCommand, cancelMultipointModeCommand
- Conference Indication message with subtype such as terminalNumberAssign, terminalYouAreSeeing

## ACLI Signaling Mode Configuration Examples

The following ACLI displays provide examples of the Signaling Modes of Operation described earlier in this chapter.

### Configuration Fields and Values for B2BGW Signaling

This example provides is a sample for the Back-to-Back Gateway Signaling mode of operation.

```
h323-config
    state                enabled
    log-level            INFO
    response-tmo        4
    connect-tmo         32
h323-stack
    name                zone1
    state                enabled
    isgateway            enabled
    realm-id            zone1realm
    assoc-stack         zone2
    local-ip            x.x.x.x (IP address of VGW-A)
    max-calls           200
    max-channels        10
    registration-ttl    0
    terminal-alias
    ras-port            h323-ID=private
                       1719
    auto-gk-discovery   enabled
    multicast           224.0.1.41:1718
    gatekeeper          x.x.x.x (IP address of GkZone1)
    gk-identifier       gk-zone1.acme.com
    q931-port           1720
    alternate-transport
    q931-max-calls      200
```

```

h245-tunneling          enabled
fs-in-first-msg         disabled
call-start-fast         disabled
call-start-slow         disabled
media-profiles
process-registration    disabled
anonymous-connection    disabled
proxy-mode
filename
h323-stack
name                    zone2
state                   enabled
isgateway               enabled
realm-id                DomainCrealm
assoc-stack             zone1
local-ip                x.x.x.x (IP address of VGW-C)
max-calls                200
max-channels            10
registration-ttl        0
terminal-alias
ras-port                h323-ID=acme01
                        1719
auto-gk-discovery       enabled
multicast               224.0.1.41:1718
gatekeeper              x.x.x.x (IP address of GkZONE2)
gk-identifier           gk-zone2.acme.com
q931-port               1720
alternate-transport
q931-max-calls          200
h245-tunneling          enabled
fs-in-first-msg         disabled
call-start-fast         disabled
call-start-slow         disabled
media-profiles
process-registration    disabled
anonymous-connection    disabled
proxy-mode
filename
h323-stack
name                    zone3
state                   enabled
isgateway               enabled
realm-id                zone3realm
assoc-stack             zone4
local-ip                x.x.x.x (IP address of VGW-B)
max-calls                200
max-channels            10
registration-ttl        0
terminal-alias
ras-port                h323-ID=private
                        1719
auto-gk-discovery       enabled
multicast               224.0.1.41:1718
gatekeeper              x.x.x.x (IP address of GkZone3)
gk-identifier           gk-zone3.acme.com
q931-port               1720
alternate-transport
q931-max-calls          200
h245-tunneling          enabled
fs-in-first-msg         disabled
call-start-fast         disabled
call-start-slow         disabled
media-profiles
process-registration    disabled

```

## H.323 Signaling Services

```
anonymous-connection      disabled
proxy-mode
filename
h323-stack
name                       zone4
state                     enabled
isgateway                 enabled
realm-id                 DomainCrealm
assoc-stack              zone3
local-ip                 x.x.x.x (IP address of VGW-D)
max-calls                200
max-channels             10
registration-ttl        0
terminal-alias

ras-port                 h323-ID=private
                        1719
auto-gk-discovery       enabled
multicast                224.0.1.41:1718
gatekeeper               x.x.x.x (IP address of GkZone4)
gk-identifier           gk-zone4.acme.com
q931-port                1720
alternate-transport
q931-max-calls           200
h245-tunneling          enabled
fs-in-first-msg         disabled
call-start-fast         disabled
call-start-slow         disabled
media-profiles
process-registration     disabled
anonymous-connection    disabled
proxy-mode
filename
```

## Back-to-Back Gatekeeper Proxy and Gateway

This example provides is a sample for the Back-to-Back Gateway Proxy and Gateway mode of operation.

```
h323-config
state                     enabled
log-level                INFO
response-tmo             4
connect-tmo              32
h323-stack
name                     zone1
state                   enabled
isgateway               disabled
realm-id               zone1realm
assoc-stack            zone2
local-ip               x.x.x.x (IP address of VGW-A/GK-A)
max-calls              200
max-channels           10
registration-ttl      0
terminal-alias

ras-port                 h323-ID=private
                        1719
auto-gk-discovery       disabled
multicast                0.0.0.0:0
gatekeeper              x.x.x.x (IP address of GkZone1)
gk-identifier           gk-zone1.acme.com
q931-port                1720
alternate-transport
q931-max-calls           200
h245-tunneling          enabled
fs-in-first-msg         disabled
```



```

call-start-fast          disabled
call-start-slow         disabled
media-profiles
process-registration     disabled
anonymous-connection    disabled
proxy-mode
filename
h323-stack
name                     zone2
state                    enabled
isgateway                disabled
realm-id                 DomainCrealm
assoc-stack              zone1
local-ip                 x.x.x.x (IP address of VGW-C/GK-C)
max-calls                200
max-channels             10
registration-ttl         0
terminal-alias
ras-port                 h323-ID=acme01
                        1719
auto-gk-discovery        disabled
multicast                0.0.0.0:0
gatekeeper               x.x.x.x (IP address of GkZONE2)
gk-identifier            gk-zone2.acme.com
q931-port                1720
alternate-transport
q931-max-calls           200
h245-tunneling           enabled
fs-in-first-msg          disabled
call-start-fast          disabled
call-start-slow         disabled
media-profiles
process-registration     disabled
anonymous-connection    disabled
proxy-mode
filename
h323-stack
name                     zone3
state                    enabled
isgateway                disabled
realm-id                 zone3realm
assoc-stack              zone4
local-ip                 x.x.x.x (IP address of VGW-B/GK-B)
max-calls                200
max-channels             10
registration-ttl         0
terminal-alias
ras-port                 h323-ID=private
                        1719
auto-gk-discovery        disabled
multicast                0.0.0.0:0
gatekeeper               x.x.x.x (IP address of GkZone3)
gk-identifier            gk-zone3.acme.com
q931-port                1720
alternate-transport
q931-max-calls           200
h245-tunneling           enabled
fs-in-first-msg          disabled
call-start-fast          disabled
call-start-slow         disabled
media-profiles
process-registration     disabled
anonymous-connection    disabled
proxy-mode

```

## H.323 Signaling Services

```
filename
h323-stack
  name                zone4
  state               enabled
  isgateway           disabled
  realm-id            DomainCrealm
  assoc-stack         zone3
  local-ip            x.x.x.x (IP address of VGW-D/GK-D)
  max-calls           200
  max-channels        10
  registration-ttl    0
  terminal-alias
  h323-ID=private
  ras-port            1719
  auto-gk-discovery   disabled
  multicast            0.0.0.0:0
  gatekeeper          x.x.x.x (IP address of GkZone4)
  gk-identifier       gk-zone4.acme.com
  alternate-transport
  q931-port           1720
  q931-max-calls      200
  h245-tunneling      enabled
  fs-in-first-msg     disabled
  call-start-fast     disabled
  call-start-slow     disabled
  media-profiles
  process-registration disabled
  anonymous-connection disabled
  proxy-mode
  filename
```

## Interworking Gatekeeper-Gateway

This example provides is a sample for the Interworking Gatekeeper-Gateway mode of operation.

```
h323-config
  state               enabled
  log-level           INFO
  response-tmo        4
  connect-tmo         32
h323-stack
  name                zone1
  state               enabled
  isgateway           disabled
  realm-id            zone1realm
  assoc-stack         zone2
  local-ip            x.x.x.x (IP address of VGW-A/GK-A)
  max-calls           200
  max-channels        10
  registration-ttl    0
  terminal-alias
  h323-ID=private
  ras-port            1719
  auto-gk-discovery   disabled
  multicast            0.0.0.0:0
  gatekeeper          x.x.x.x (IP address of GkZone1)
  gk-identifier       gk-zone1.acme.com
  q931-port           1720
  alternate-transport
  q931-max-calls      200
  h245-tunneling      enabled
  fs-in-first-msg     disabled
  call-start-fast     disabled
  call-start-slow     disabled
```

```

media-profiles
process-registration          disabled
anonymous-connection         disabled
proxy-mode
filename
h323-stack
name                          zone2
state                          enabled
isgateway                      enabled
realm-id                       DomainCrealm
assoc-stack                    zone1
local-ip                       x.x.x.x(IP address of VGW-C)
max-calls                      200
max-channels                   10
registration-ttl               0
terminal-alias                 h323-ID=acme01
ras-port                       1719
auto-gk-discovery              enabled
multicast                      0.0.0.0:0
gatekeeper                    0.0.0.0:0
gk-identifier                  gk-zone2.acme.com
q931-port                     1720
alternate-transport
q931-max-calls                 200
h245-tunneling                 enabled
fs-in-first-msg                disabled
call-start-fast                disabled
call-start-slow                disabled
media-profiles
process-registration          disabled
anonymous-connection         disabled
proxy-mode
filename
h323-stack
name                          zone3
state                          enabled
isgateway                      disabled
realm-id                       zone3realm
assoc-stack                    zone4
local-ip                       x.x.x.x(IP address of VGW-B/GK-B)
max-calls                      200
max-channels                   10
registration-ttl               0
terminal-alias                 h323-ID=private
ras-port                       1719
auto-gk-discovery              disabled
multicast                      0.0.0.0:0
gatekeeper                    x.x.x.x(IP address of GkZone3)
gk-identifier                  gk-zone3.acme.com
q931-port                     1720
alternate-transport
q931-max-calls                 200
h245-tunneling                 enabled
fs-in-first-msg                disabled
call-start-fast                disabled
call-start-slow                disabled
media-profiles
process-registration          disabled
anonymous-connection         disabled
proxy-mode
filename
h323-stack

```

```
name zone4
state enabled
isgateway enabled
realm-id DomainCrealm
assoc-stack zone3
local-ip x.x.x.x (IP address of VGW-D)
max-calls 200
max-channels 10
registration-ttl 0
terminal-alias

ras-port h323-ID=private
auto-gk-discovery 1719
multicast disabled
gatekeeper 0.0.0.0:0
gk-identifier x.x.x.x (IP address of GkZone4)
gk-identifier gk-zone4.acme.com
```

## Additional Information

This section contains detailed tables to use as a reference when you are learning about H.323 features or when you are configuring them.

### About Payload Types

You set the payload type when you are configuring a media profile to support Slow Start to Fast Start Translation.

When you configure media profiles, you might need set the payload type to identify the format in the SDP m lines. For RTP/AVP, the default transport method of a media profile configuration, this will be the RTP payload type number. Newer codecs have dynamic payload types, which means that they do not have an assigned payload type number.

When you use RTP/AVP as the transport method, you should only set the payload type when there is a standard payload type number for the encoding name; otherwise, leave the payload type blank.

The Oracle Communications Session Border Controller uses the payload type value to determine the encoding type when SDP identifies the standard payload type in the m line, but does not include an `a=rtpmap` entry. These are two equivalent SDPs:

```
c=IN IP4 192.0.2.4
```

```
m=audio 0 RTP/AVP 0
```

```
c=IN IP4 192.0.2.4
```

```
m=audio 0 RTP/AVP 0
```

```
a=rtpmap:0 PCMU/8000
```

The first does not include the RTP map entry, but uses the standard payload type of 0. If the Oracle Communications Session Border Controller receives an SDP like the first, it uses the payload type 0 to locate the corresponding media profiles configuration. When an `a=rtpmap` is present, the Oracle Communications Session Border Controller uses the encoding name in the `a=rtpmap` line to find the media profile configuration and does not consider the payload type number.

### Payload Types for Standard Audio and Visual Encodings

The following is a table of standard audio and visual payload encodings defined in H. Schulzrinne, GND Fokus, RTP Profile for Audio and Visual Conferences with Minimal Control, RFC 1890, and in the *RTP Parameters* document in IANA's Directory of Generally Assigned Numbers.

Payload Type	Encoding Name	Audio (A)/Visual (V)	Clock Rate (Hz)
0	PCMU	A	8000


Payload Type	Encoding Name	Audio (A)/Visual (V)	Clock Rate (Hz)
1	1016	A	8000
2	G721	A	8000
3	GSM	A	8000
4	G723	A	8000
5	DVI4	A	8000
6	DVI4	A	16000
7	LPC	A	8000
8	PCMA	A	8000
9	G722	A	8000
10	L16	A	44100
11	L16	A	44100
12	QCELP	A	8000
13	reserved	A	
14	MPA	A	90000
15	G728	A	8000
16	DVI4	A	11025
17	DVI4	A	22050
18	G729	A	8000
19	reserved	A	
20	unassigned	A	
21	unassigned	A	
22	unassigned	A	
23	unassigned	A	
dyn	GSM-HR	A	8000
dyn	GSM-EFR	A	8000
dyn	L8	A	var.
dyn	RED	A	
dyn	VDVI	A	var.
24	unassigned	V	
25	CelB	V	90000
26	JPEG	V	90000
27	unassigned	V	
28	nv	V	90000
29	unassigned	V	

## H.323 Signaling Services

Payload Type	Encoding Name	Audio (A)/Visual (V)	Clock Rate (Hz)
30	unassigned	V	
31	H261	V	90000
32	MPV	V	90000
33	MP2T	AV	90000
34	H263	V	90000
35-71	unassigned	?	
72-76	reserved for RTCP conflict avoidance	N/A	N/A
77-95	unassigned	?	
96-127	dynamic	?	
dyn	BT656	V	90000
dyn	H263-1998	V	90000
dyn	MP1S	V	90000
dyn	MP2P	V	90000
dyn	BMPEG	V	90000

### About RAS Message Treatment

When you enabled the H.323 Registration Proxy, the Oracle Communications Session Border Controller modifies and deletes certain fields as outlined in the table below. The Oracle Communications Session Border Controller sends on any fields that are not listed in this table without modifying or deleting them.

 **Note:** Although the Oracle Communications Session Border Controller forwards a field, it does not always support the feature related to that field.

Field Name	Message	Deleted	Modified	Value Used in Modification
alternateEndpoints	RRQ, URQ, ACF	X		
alternateGatekeeper	RCF, URQ	X		
altGKInfo	RRJ, URJ, DRJ	X		
alternateTransportAddresses	RRQ, ARQ, ACF	X		
callModel	ARQ		X	direct
	ACF		X	gatekeeperRouted
callSignalAddress	RRQ		X	Mapped virtual CSA allocated by the system for registering the endpoint.
	RCF, ARJ		X	CSA of gatekeeper stack
	URQ		X	If URQ is from an endpoint, endpoint's

Field Name	Message	Deleted	Modified	Value Used in Modification
				mapped virtual CSA. If URQ is from a gatekeeper, real CSA of endpoint.
destCallSignalAddress	ARQ, ACF	X		
destinationInfo.transportID	ARQ, ACF	X		
destExtraCallInfo.trasportID	ARQ, ACF	X		
discoveryComplete	RRQ		X	TRUE
endpointAlias.trasportID	URQ	X		
endpointAliasPattern.Wwildcard.trans portID	URQ			
featureServerAlias.trasportID	RCF	X		
gatekeeperIdentifier	RRQ		X	Gatekeeper identifier of the gateway stack, either configured in the H.323 gateway stack or discovered dynamically.
maintainConnection	RRQ, RCF		X	FALSE
mutipleCall	RRQ, RCF		X	FALSE
preGrantedARQ.alternateTransportAd dresses	RCF	X		
preGrantedARQ.useSpecifiedTranspor t	RCF	X		
rasAddress	RRQ		X	Mapped virtual RAS address allocated by the system for registering endpoint
remoteExtentsionAddress.trasportID	ARQ, ACF	X		
srcCallSignalAddress	ARQ	X		
srcInfo.transportID	ARQ	X		
supportedH248Packages	RRQ	X		
supportsAltGK	RRQ	X		
supportedPrefixes.prefic.transportID	RCF, URQ	X		
terminalAlias.transportID	RRQ	X		
terminalAliasPattern.wilcard.transport ID	RRQ	X		
willRespondToIIRR	RCF, ACF	X		
willSupplyUUIEs	RRQ, ARQ			
uuiesRequested	ACF		X	FALSE

### H.323 Signaling Services

Field Name	Message	Deleted	Modified	Value Used in Modification
setup			X	FALSE
callProceeding			X	FALSE
connect			X	FALSE
alerting			X	FALSE
information			X	FALSE
releaseComplete			X	FALSE
facility			X	FALSE
progress			X	FALSE
empty			X	FALSE
...			X	FALSE
status			X	FALSE
statusInquiry			X	FALSE
setupAcknowledge				
notify				



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## IWF Services

Using the Oracle Communications Session Border Controller's interworking function (IWF), you can interconnect SIP networks with H.323 networks. Considering the large amount of H.323 deployments already in place and the continuing emergence of SIP in new VoIP deployments, the IWF provides a much-needed solution. SIP providers can maintain a single-protocol backbone while exchanging VoIP sessions with H.323 providers.

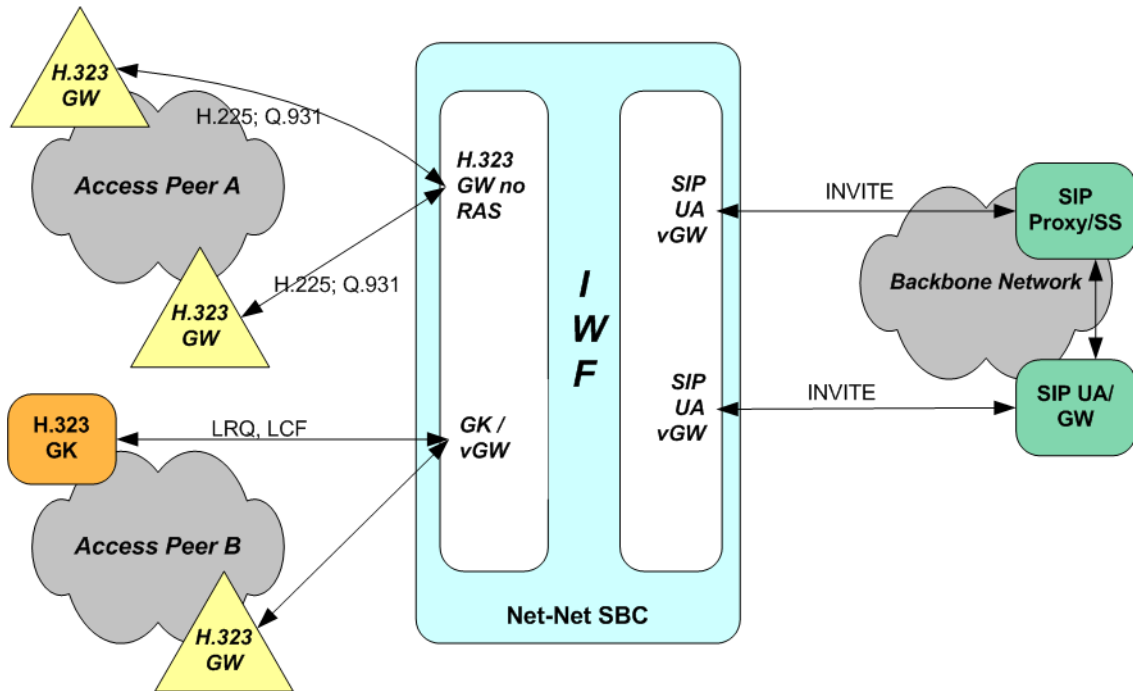
The H.323 Signaling Services section contains information about the H.323 signaling modes of operation that the Oracle Communications Session Border Controller supports. The following H.323 signaling modes of operation can be used when you use the Oracle Communications Session Border Controller's IWF in an access or a peering solution.

- Back-to-back gateway signaling
- Interworking gatekeeper/gateway

## Access Network Application

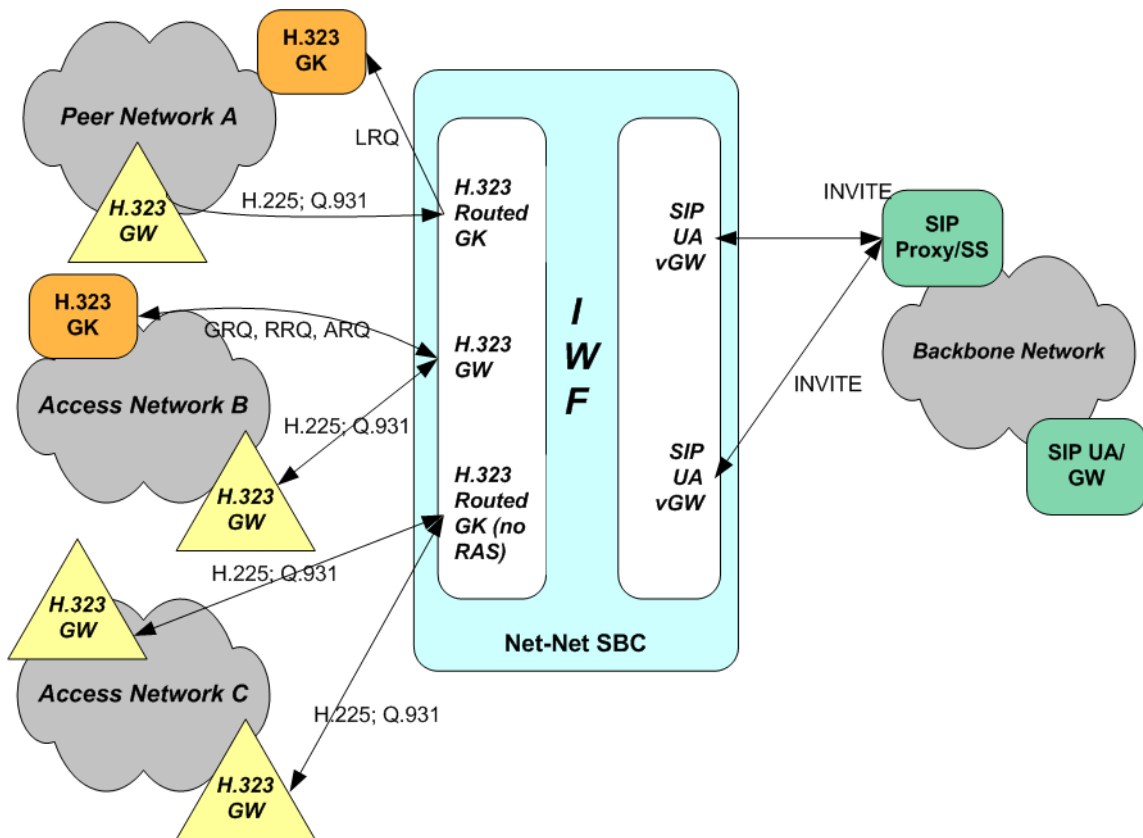
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You can configure your Oracle Communications Session Border Controller so that it provides an access solution for your network. The access solution allows SIP-based hosted communications platforms to be extended to enterprise-based H.323 systems. In the figure below, you can see different types of H.323 signaling modes being interworked with SIP. On the H.323 side, the Oracle Communications Session Border Controller can appear to be a gatekeeper or a gateway, depending on how you configure the H.323 interface. On the SIP side, the Oracle Communications Session Border Controller can appear to be a SIP UA or behave as a virtual gateway.



## Networking Peering Application

In the IWF network peering solution, you can see the same network elements at work. However, the H.323 side of this IWF application shows the use of a gatekeeper controlled gateway for Peer Network B. Because this is a peering solution, the SIP side of the Oracle Communications Session Border Controller communicates with the SIP proxy or softswitch in the backbone network rather than with the SIP UA or SIP gateway.



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## SIP and H.323

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The Oracle Communications Session Border Controller supports interworking between SIP and H.323 for H.323 Slow Start and Fast Start calls. In addition to describing IWF sessions when initiated from the H.323 side and from the SIP side (with sample call flows), this section provides information you will need when you configure SIP and H.323.

### SIP H.323 Negotiation H.323 Fast Start

The Oracle Communications Session Border Controller can perform protocol translations for SIP and H.323 Fast Start, where media capabilities are sent with the Setup request for an H.323 session.

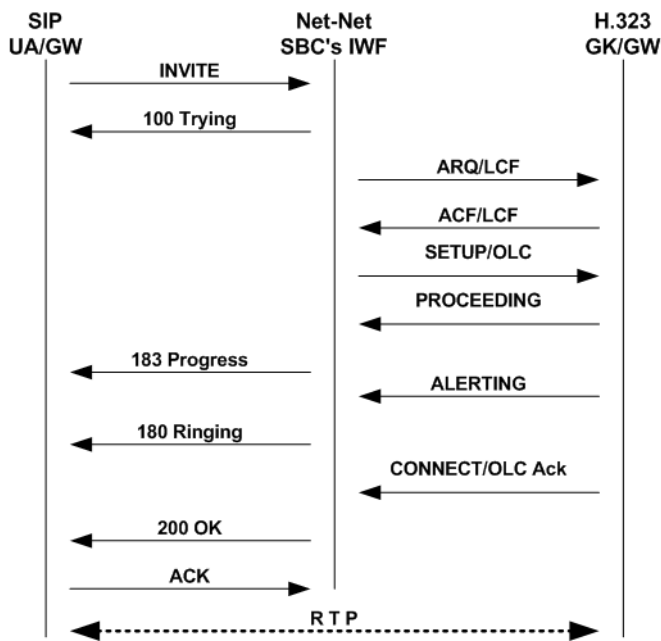
This section's call flow diagrams show how SIP and H.323 messages flow between SIP and H.323 devices, with the Oracle Communications Session Border Controller positioned between the two entities so it can perform translations. The following two sample scenarios with Fast Start appear in the diagrams below, although other scenarios are possible:

- Calls originating in SIP being translated to H.323 Fast Start
- Calls originating in H.323 Fast Start translated to SIP

#### SIP to Fast Start H.323

In the following diagram below, a SIP endpoint (such as a UA or a SIP Gateway) initiates a session by sending an INVITE message destined for an H.323 endpoint (a GK or GW). Between these entities, the system is positioned to perform interworking. The Oracle Communications Session Border Controller recognizes that the INVITE message is destined for an H.323 device, and returns a 100 Trying message to the SIP endpoint as it attempts to negotiate the H.323 side of the session. This negotiation starts when the Oracle Communications Session Border Controller initiates the RAS process with the H.323 endpoint by sending either an ARQ or an LRQ, allowing the Oracle Communications Session Border Controller to determine if the H.323 endpoint will accept the session.

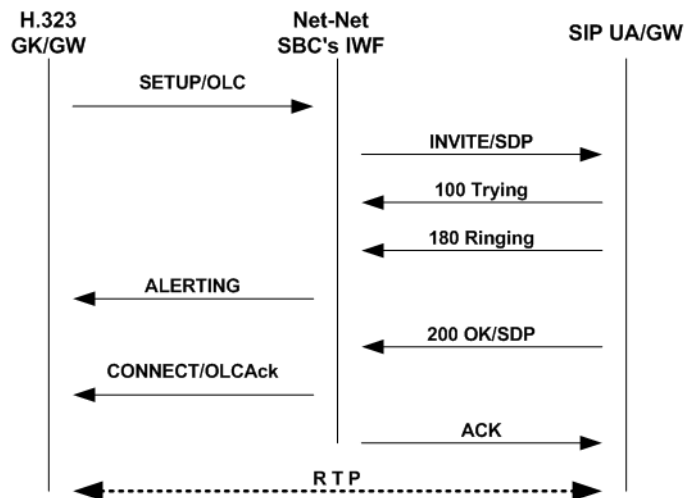
Once the H.323 endpoint responds with an ACF or LCF, the Oracle Communications Session Border Controller reissues the SIP INVITE on the H.323 side as an H.225 Setup, which is sent with the OLC. Then the H.323 endpoint responds with Proceeding and Alerting messages (which correspond respectively to SIP 183 Progress and 180 Ringing messages). At that point, the H.323 endpoint sends a Connect message that includes the OpenLogicalChannel message (OLC), announcing the logical channel for media flows has been set up. The Oracle Communications Session Border Controller converts the H.323 OLC to a SIP 200 OK. After receiving the 200 OK, the SIP endpoint sends an ACK, confirming that the session has been established. Because there is no H.323 equivalent for the SIP ACK, the Oracle Communications Session Border Controller does not generate a corresponding message on the H.323 side. At this point, the session is fully established and RTP flows between the endpoints.



**H.323 Fast Start to SIP**

In the diagram below, an H.323 endpoint (a GK or GW) initiates a session by sending a Setup request destined for a SIP endpoint (such as a UA or a SIP Gateway). Between these entities, the Oracle Communications Session Border Controller is positioned to perform interworking. The H.323 endpoint has completed the RAS process prior to sending the SETUP message.

The Oracle Communications Session Border Controller receives the Setup message and then sends a SIP INVITE on the SIP side. The SIP endpoint responds with a 100 Trying; the Oracle Communications Session Border Controller does not resend this message on the H.323 side. Next, the SIP endpoint issues a 180 Ringing message, which the Oracle Communications Session Border Controller reissues to the H.323 endpoint as an Alerting message. The SIP endpoint then sends a 200 OK, retransmitted by the Oracle Communications Session Border Controller as a Connect message that includes an OLC. Once the Oracle Communications Session Border Controller sends an ACK to the SIP endpoint, RTP flows between the endpoints.



**SIP H.323 Negotiation H.323 Slow Start**

The Oracle Communications Session Border Controller can also perform protocol translations for SIP and H.323 Slow Start, where—unlike the cases with Fast Start described above—media information is not sent with the Setup request for an H.323 session. For H.323 Slow Start, media is negotiated after the session is established.

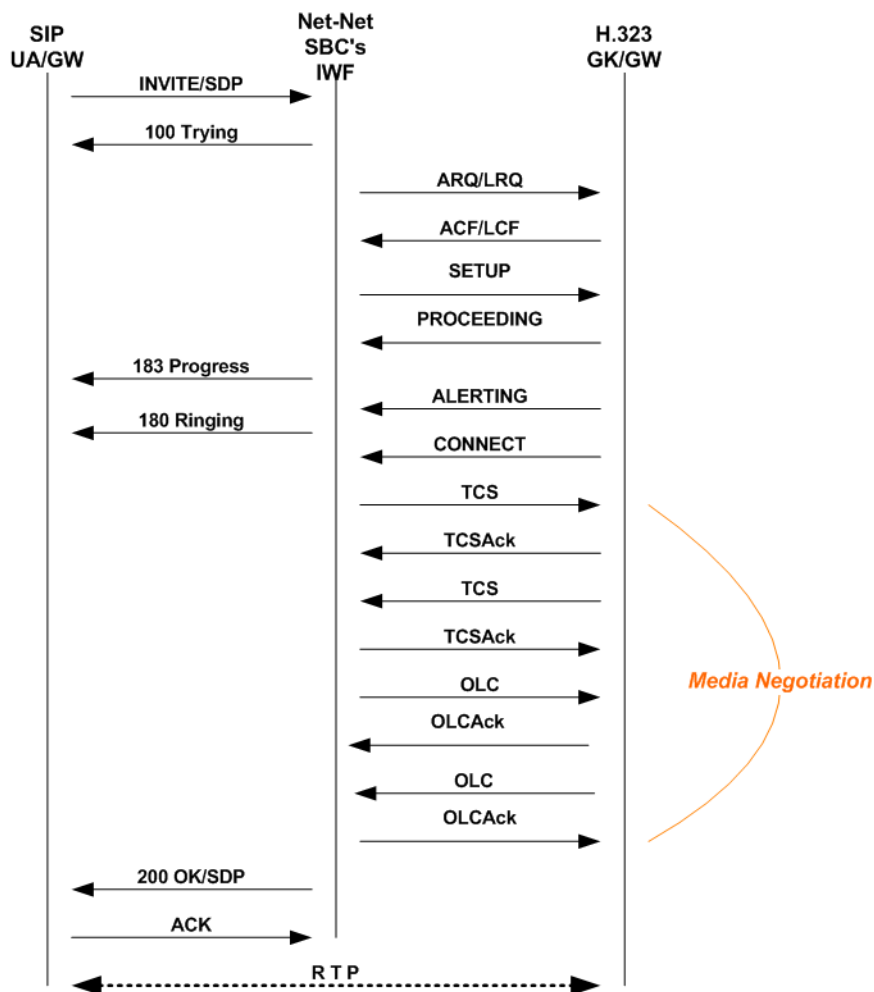
This section's call flow diagrams show how SIP and H.323 messages flow between SIP UA/GW and an H.323 GK/GW, with the Oracle Communications Session Border Controller positioned between the two entities so it can perform translations. Two sample scenarios with Slow Start appear in the diagrams below:

- SIP being interworked to Slow Start H.323
- Slow Start H.323 being interworked to SIP

### H.323 SIP to Slow Start

In the following diagram below, a SIP endpoint (such as a UA or a SIP Gateway) initiates a session by sending an INVITE request destined for an H.323 Slow Start endpoint (a GK or GW). Between these entities, the Oracle Communications Session Border Controller is positioned to perform interworking.

The call flow for this type of translation works fundamentally the same way that the translation does for SIP to Fast Start H.323, with the exception of how the media is established. Media is negotiated through the exchange of TCS and OLC messages after the H.323 Connect and SIP 180 Ringing messages have been sent. The first TCS message is sent from the Oracle Communications Session Border Controller to the H.323 endpoint, and it contains information about media capabilities in SDP. The H.323 endpoint accepts and acknowledges this information with a TCS Ack message. Then the H.323 endpoint sends a second TCS, carrying information about the Gateway's capabilities, that the Oracle Communications Session Border Controller accepts and acknowledges. The H.323 endpoint and the Oracle Communications Session Border Controller then exchange OLC and OLC Ack messages that establish the operating mode and Gateway capability. Finally, the Oracle Communications Session Border Controller completes the 200 OK/ACK sequence on the SIP side, and RTP flows between the two endpoints.

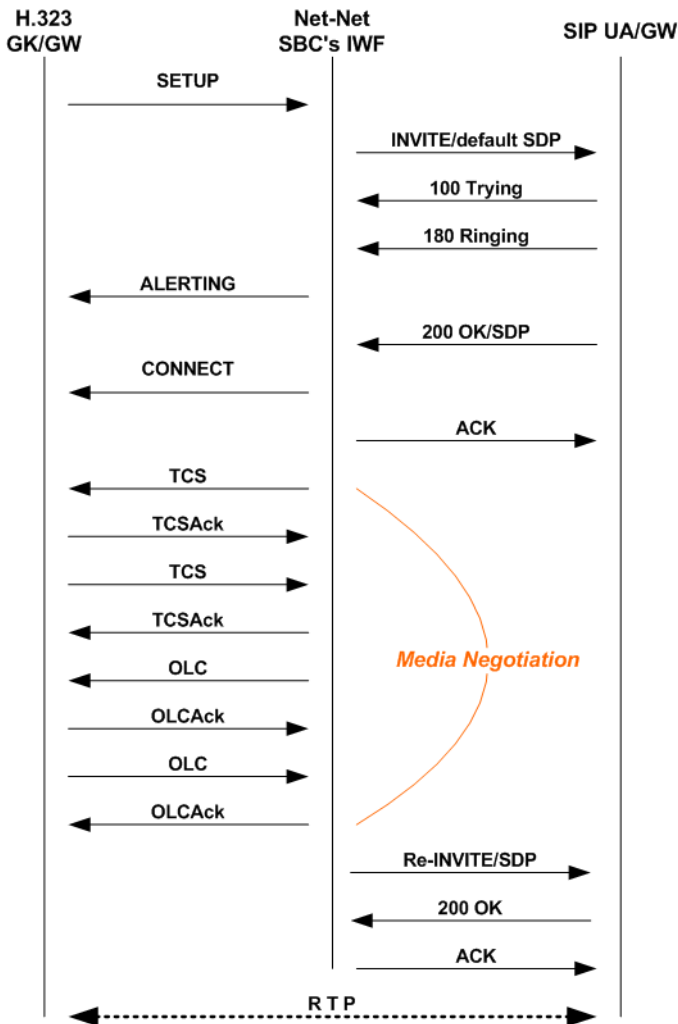


**H.323 Slow Start to SIP**

In the following diagram below, an H.323 endpoint (GW or GK) initiates a session by sending a Setup request destined for a SIP endpoint (such as a UA or a SIP Gateway). Between these entities, the Oracle Communications Session Border Controller is positioned to perform interworking. The H.323 endpoint has completed the RAS process prior to sending the SETUP message.

The call flow for this type of translation works fundamentally the same way that the translation does for H.323 Fast Start to SIP, with the exception of how the media is established. When the Oracle Communications Session Border Controller receives an H.323 message destined for a SIP endpoint, it sends a SIP INVITE message that includes default SDP to that SIP endpoint. The default SDP is constructed using information in the media profiles listed for the IWF configuration; if necessary, this media information is amended later in the sequence. Once the call is set up, the Oracle Communications Session Border Controller negotiates media with the H.323 endpoint through a series of TCS/TCS Ack and OLC/OLC Ack messages that establish the operating mode and Gateway capability.

When the Oracle Communications Session Border Controller completes media negotiation with the H.323 endpoint, it issues a re-INVITE to the SIP endpoint that contains the updated information needed for media transmission. In response, the SIP endpoint sends a 200 OK message that the Oracle Communications Session Border Controller answers with an ACK. Then RTP can flow between the two endpoints.



**Status and Codec Mapping**

The Oracle Communications Session Border Controller maps SIP and H.323 status codes as described in this section. Status and codec mapping do not require configuration; they occur transparently.

### IWF Termination from H.323

When a call that requires the IWF terminates from the H.323 side, the Oracle Communications Session Border Controller uses the mapping scheme in the following table to determine the appropriate SIP status.

H.323 Disconnect Reason	SIP Status
No Bandwidth	480 Temporarily Unavailable
Gatekeeper Resource	404 Not Found
Unreachable Destination	404 Not Found
Destination Rejection	603 Decline
Invalid Revision	505 Version Not Supported
No Permission	401 Unauthorized
Unreachable Gatekeeper	503 Service Unavailable
Gateway Resource	480 Temporarily Unavailable
Bad Format Request	400 Bad Request
Adaptive Busy	486 Busy Here
In Conference	486 Busy Here
Undefined Reason	500 Internal Server Error
Facility Call Deflection	486 Busy Here
Security Denied	401 Unauthorized
Called Party Not Registered	404 Not Found
Caller Not Registered	401 Unauthorized

### IWF Termination During H.323 RAS

When a call that requires the IWF terminates from the H.323 side during RAS and generates an error, the Oracle Communications Session Border Controller uses the mapping scheme in the following table to determine the appropriate SIP status.

H.323 RAS Error	SIP Status
Called Party Not Registered	404 Not Found
Invalid Permission	401 Unauthorized
Request Denied	503 Service Unavailable
Undefined	500 Internal Server Error
Caller Not Registered	401 Unauthorized
Route Call To Gatekeeper	305 User Proxy
Invalid Endpoint ID	500 Internal Server Error
Resource Unavailable	503 Service Unavailable
Security Denial	401 Unauthorized
QoS Control Not Supported	501 Not Implemented
Incomplete Address	484 Address Incomplete

## IWF Services

H.323 RAS Error	SIP Status
Route Call to SCN	302 Moved Temporarily
Aliases Inconsistent	485 Ambiguous
Not Currently Registered	401 Unauthorized

### IWF RAS Registration Failure Code Mapping

For calls that require interworking between H.323 and SIP, the Oracle Communications Session Border Controller supports IWF response code mapping. This feature enables the Oracle Communications Session Border Controller to support configurable SIP response codes for IWF calls that fail during RAS, when the Oracle Communications Session Border Controller has been unable to register with a gatekeeper; this allows a wider range of more accurate response codes to be communicated.

When this feature is not enabled, the Oracle Communications Session Border Controller generates a 404 Not Found when a SIP-to-H.323 call fails as a result of the stack's failure to register with a gatekeeper.

When the condition noted above takes place, the response code can be any of the ones listed in this table. The code values listed in the table are used to specify the code to which you want to map.

Code	Description
403	Forbidden
406	Not Acceptable
408	Request Timeout
410	Gone
420	Bad Extension
480	Temporarily Unavailable
486	Busy Here
487	Request Terminated
500	Server Internal Error
503	Service Unavailable
504	Server Time-out
600	Busy Everywhere
603	Decline

To enable IWF RAS registration failure code mapping:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router  
ACMEPACKET(session-router)#
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323  
ACMEPACKET(h323)#
```



4. **options**—Set the options parameter by typing **options**, a Space, the option name preceded by a plus sign (+) (**iwfRegFailCode=X**), and then press Enter. X is the SIP response code that you want to use; the table above lists the supported response codes that are supported.

```
ACMEPACKET (h323) # options +iwfRegFailCode=503
```

If you type options iwfRegFailCode=X, you will overwrite any previously configured options. In order to append the option to the options list, you must prepend the new option with a plus sign as shown in the previous example.

### IWF Termination from SIP

When a call that requires the IWF terminates from the SIP side, the Oracle Communications Session Border Controller uses the mapping scheme in the following table to determine the appropriate H.323 Release Complete Reason code.

SIP Status	H.323 Release Complete Reason
300 Multiple Choices	Undefined Reason
401 Unauthorized	Security Denied
402 Payment Required	Undefined Reason
403 Forbidden	No Permission
404 Not Found	Unreachable Destination
405 Method Not Allowed	Undefined Reason
406 Not Acceptable	Undefined Reason
407 Proxy Authentication Required	Security Denied
408 Request Timeout	Adaptive Busy
409 Conflict	Undefined Reason
410 Gone	Unreachable Destination
411 Length Required	Undefined Reason
414 Request-URI Too Large	Bad Format Address
415 Unsupported Media Type	Undefined Reason
420 Bad Extension	Bad Format Address
480 Temporarily Unavailable	Adaptive Busy
481 Call/Transaction Does Not Exist	Undefined Reason
482 Loop Detected	Undefined Reason
483 Too Many Hops	Undefined Reason
484 Address Incomplete	Bad Format Address
485 Ambiguous	Undefined Reason
486 Busy Here	In Conference
487 Request Terminated	Undefined Reason
488 Not Acceptable Here	Undefined Reason
500 Internal Server Error	Undefined Reason

SIP Status	H.323 Release Complete Reason
501 Not Implemented	Undefined Reason
502 Bad Gateway	Gateway Resource
503 Service Unavailable	Gateway Resource
504 Gateway Timeout	Adaptive Busy
505 Version Not Supported	Invalid Revision
600 Busy Everywhere	Adaptive Busy
603 Decline	Destination Rejection
604 Does Not Exist Anywhere	Unreachable Destination
606 Not Acceptable	Undefined Reason

**Q.850 Cause to H.323 Release Complete Reason**

When a call that requires the IWF terminates from the H.323 side and no H.323 Release Complete Reason is specified, the Oracle Communications Session Border Controller maps the Q.850 cause to an H.323 Release Complete Reason using the mapping scheme in the following table. This new H.323 status is then mapped to a SIP status as described in the IWF Termination from SIP table.

Q.850 Cause	H.323 Release Complete Reason
No Route To Destination	Unreachable Destination
Normal Call Clearing	Destination Rejection
User Busy	In Conference
Subscriber Absent	Called Party Not Registered
Invalid Number Format	Bad Format Address
Normal Unspecified	Undefined Reason
No Circuit/Channel Available	No Bandwidth
Network Out Of Order	Unreachable Gatekeeper
Temporary Failure	Adaptive Busy
Switching Equipment Congestion	Gateway Resource
Resource Unavailable	Gatekeeper Resource
Incompatible Destination	Invalid Revision
Interworking Unspecified	No Permission

**Codec Mapping**

The Oracle Communications Session Border Controller uses the following mapping scheme when converting media specifications between H.245 (used in H.323) and SDP (used in SIP).

Media coming into the Oracle Communications Session Border Controller one way exits the system in the corresponding way as specified in the following table. For example, media coming into the Oracle Communications Session Border Controller as H.245 type g711Ulaw64k exits the system as media type PCMU.

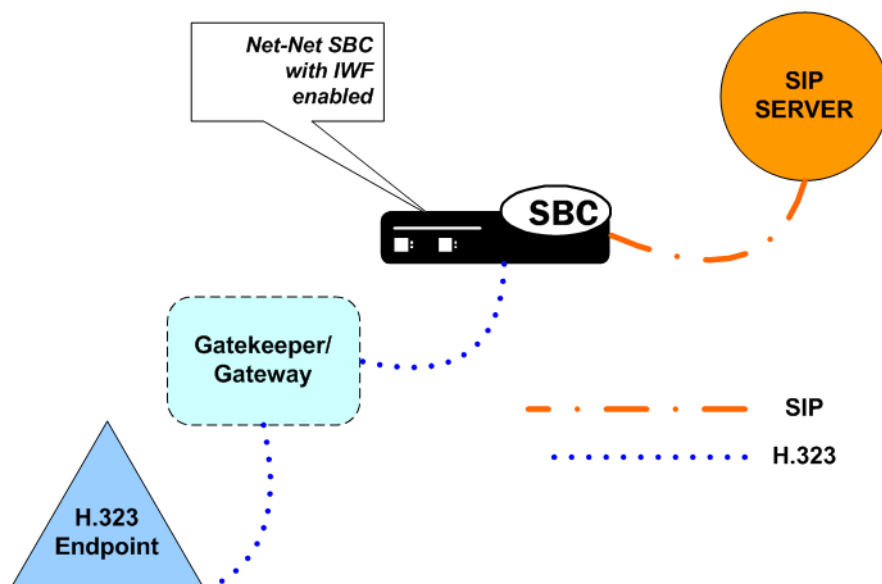
H.245 Type	SDP Media Type
g711Ulaw64k	PCMU
g711Ulaw56k	PCMU
g711Alaw64k	PCMA
g711Alaw56k	PCMA
g726	G726-32
g7231	G723
g722	G722
g728	G728
g729wAnnexB	G729
g729	G729 fmtp:18 annexb=no
h261VideoCapability	H261
h263VideoCapability	H263

## IWF Service Enhancements

This section describes the Oracle Communications Session Border Controller features that are supported for when the Oracle Communications Session Border Controller performs interworking between SIP and H.323. Enabling these enhancements only requires that you set up a fully functional SIP configuration, a fully functional H.323 configuration, and that you enable IWF on your Oracle Communications Session Border Controller. You do not have to set any special configuration because these enhancements happen automatically.

### SIP Redirect—H.323 LRQ Management

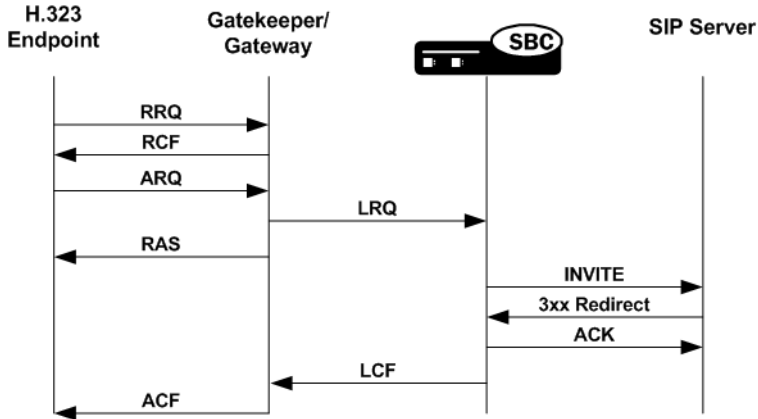
When it needs to interact with a SIP Redirect server, the Oracle Communications Session Border Controller can interpret the SIP messages and manage them on the H.323 side of the session. For IWF sessions, the Oracle Communications Session Border Controller handles SIP Redirect and H.323 LRQ messages.



**Redirect—LRQ Management Sample 1**

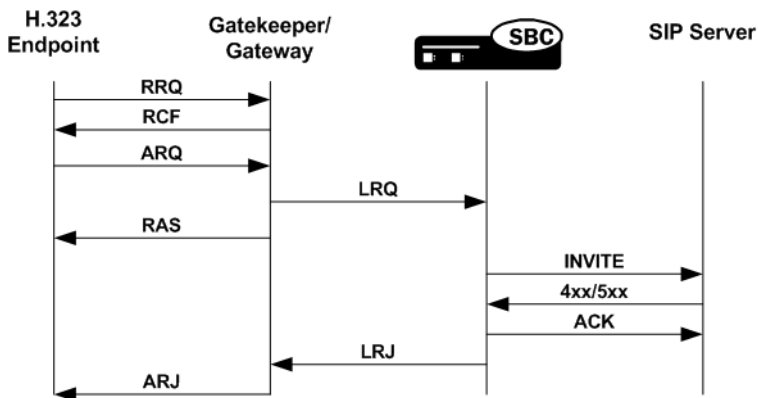
This section presents three possible scenarios for SIP Redirect-H.323 LRQ management.

The following diagram shows an established session that uses SIP Redirect—H.323 LRQ management. Here, the Oracle Communications Session Border Controller sends an INVITE to a SIP Redirect Server that responds with a 3xx Redirection message. The Oracle Communications Session Border Controller then sends the gatekeeper/gateway an LCF message that causes an ACF message to be sent to the H.323 endpoint.



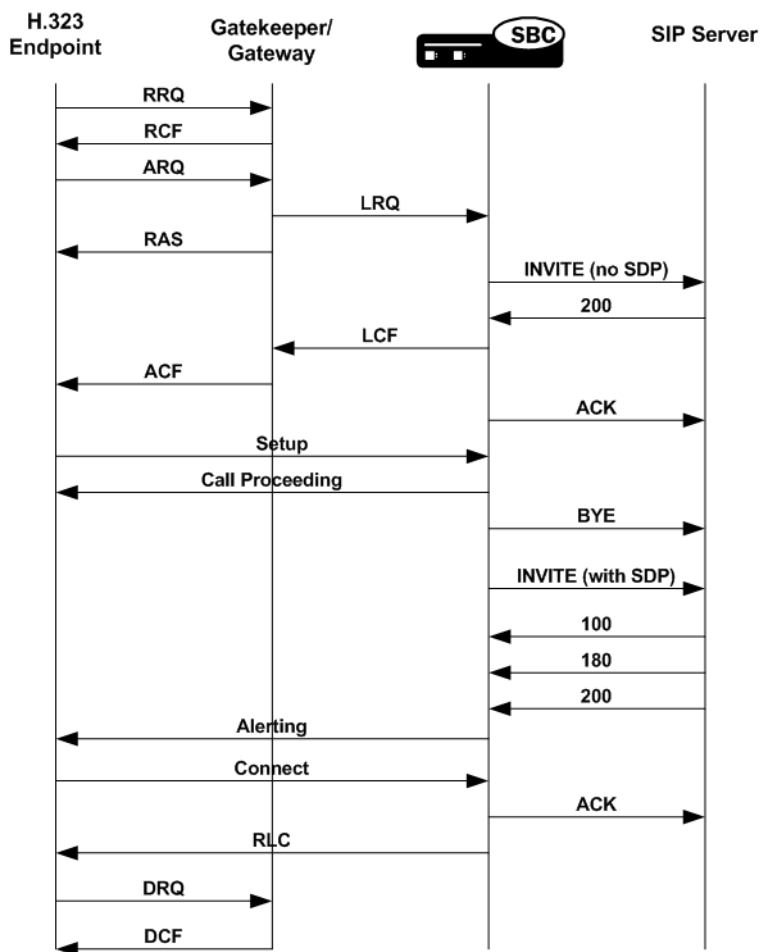
**Redirect—LRQ Management Sample 2**

The following diagram shows how the Oracle Communications Session Border Controller handles the exchange when the SIP Redirect server declares either that there is an error or that there is no such user. These SIP messages come from either the 4xx Request Failure or 5xx Server Failure series. In the example below, the SIP Redirect server returns a 401 Unauthorized message, which the Oracle Communications Session Border Controller interworks and communicates to the H.323 gatekeeper as an LRJ. Then the H.323 gatekeeper/gateway issues an ARJ to the H.323 endpoint.



**Redirect—LRQ Management Sample 3**

In this call flow, the SIP server issues a 2xx Successful message that is not supposed to be sent (because a 3xx, 4xx, or 5xx message should be sent in response to the Oracle Communications Session Border Controller’s INVITE). The Oracle Communications Session Border Controller sends a BYE message to the SIP Redirect Server, but it tries to initiate the session again, this time successfully. The final sample call flow shown rarely occurs.



## SIP INFO and DTMF UII Management

The Oracle Communications Session Border Controller supports DTMF for that require the IWF, enabling features such as keypress, alphanumeric, and hookflash. Because tones are not transmitted as audio, they must pass as out-of-band signaling information, meaning that the Oracle Communications Session Border Controller needs to convert an H.245 UII (User Input Indication) into SIP.

Depending on the capability of the H.323 endpoint, the Oracle Communications Session Border Controller sends either an alphanumeric or DTMF signal in the H.245 UII. The Oracle Communications Session Border Controller sends nothing if the endpoint does not support an alphanumeric or DTMF signal. The SIP INFO message will have a content type of application/dtmf-relay, and the message body will be in the form `Signal=*\r\nDuration=250\r\n`. If the duration is absent in the SIP INFO or the UII received on the H.323 side is alphanumeric, the Oracle Communications Session Border Controller uses the a 250 millisecond default value.

## Mid-Session Media Change

Mid-session media change happens during a call that requires the IWF when the type of media being sent while a session is in progress changes. For example, a fax transmission might require mid-session media change; besides fax, other applications of this feature are possible. To support the transmission of a T.38 fax sent over an IWF session, some media channels must be opened and others closed. In addition, the Oracle Communications Session Border Controller can accommodate a request for media change from, for example, audio to an image type for T.38 fax.

Because the media requirements are driven by endpoints and Gateways, you do not have to configure the Oracle Communications Session Border Controller s mid-session media change support.

### Enhanced Support for FAX Calls

The Oracle Communications Session Border Controller now supports T.38 fax calls in networks containing elements that do not comply with the ITU-T H.323 Annex D recommendation for how to replace an existing audio stream with a T.38 fax stream. This support applies to signaling that requires interworking between SIP and H.323.

In the standard call model following the ITU-T recommendation, the endpoint detecting the fax tone sends an H.245 RequestMode message to its peer with a T.38 data mode. The receiving endpoint returns a RequestMode Ack by way of acknowledgement, triggering the sending endpoint to close its audio channel and open a T.38 fax channel. The receiving endpoint closes and opens the same channels on its end. T.38 fax streams flow upon the acknowledgement of all relevant channels.

However, certain endpoints close their logical channel before sending the H.245 RequestMode message for T.38, leaving the Oracle Communications Session Border Controller with its audio channel still open and without having attempted to open a T.38 fax channel. To overcome this issue, the Oracle Communications Session Border Controller now checks whether or not audio channels have been closed whenever it receives an H.245 RequestMode message for T.38. If it finds a closed audio channel, the Oracle Communications Session Border Controller checks for the presence of a matching outgoing audio channel. A match causes the Oracle Communications Session Border Controller to close the audio channel and continue with the procedure for converting to T.38 fax.

### Removing the T.38 Codec from an H.245 TCS

For SIP-H.323 IWF sessions, H.323 automatically inserts the T.38 FAX codec in the H.246 TCS message. You can stop this insertion using the `remove-t38` parameter in the H.323 global configuration.

## Early Media

For call that require the IWF, the Oracle Communications Session Border Controller supports a cut-through for early media for calls that originate in SIP or H.323.

For a session originating in SIP, the provisional message will contain the SDP information if a Fast Start OLC was received in the Call Proceeding, Alerting, or Progress messages. The same SDP will be sent in the SIP 200 OK.

For a session that starts in H.323, the Oracle Communications Session Border Controller translates the SDP it receives in SIP messages (either a 180 or a 183) into the appropriate H.323 Fast Start elements: Alerting or Progress. If the Alerting or Progress messages contain Fast Start elements, the Progress Indicator Q.931 information element (IE) will also be included in the message with Progress Descriptor 8, indicating that in-band information or an appropriate pattern is now available. This causes the call party to enable end-to-end early media in the reverse direction in accordance with H.323 v4.

In addition, the Oracle Communications Session Border Controller allows early media to flow in the forward direction for a call that requires the IWF starting in H.323 that is being translated to SIP. This happens after the Oracle Communications Session Border Controller has received provisional response with SDP and has sent Alerting or Progress message with Fast Start to the calling party. Similarly, early media in the forward direction is enabled for a call that requires the IWF starting in SIP and being translated to H.323. This happens after the Oracle Communications Session Border Controller received Alerting or Progress messages with Fast Start and maps the Alerting or Progress to SIP 180 or 183 provisional response with the SDP answer.

## Display Name Mapping

The Oracle Communications Session Border Controller displays the full name and number of the calling party (for features such as Caller ID) when it handles calls that require the IWF. The Oracle Communications Session Border Controller takes the display name in the From field of the SIP INVITE and maps it to the display IE so that it can show the full name of the calling party.

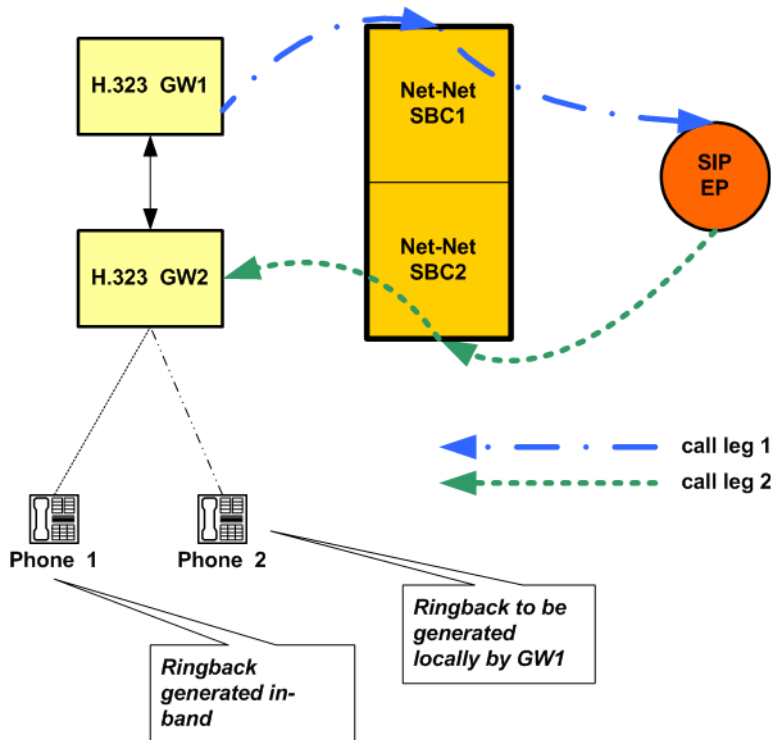
## IWF Ringback Support

When interworking SIP and H.323 to a gateway, PSTN gateway, or other endpoint, the Oracle Communications Session Border Controller uses the mappings shown in the table below. The absence or presence of SDP in the SIP provisional message determines whether the tones are generated in-band or locally.

For each of the mappings listed in the following table, this section provides a sample call flow.

SIP Message	H.323 Message
No Message	CallProceeding
No Message	Progress without PI
183 with SDP	Progress with PI
180 w/o SDP	Alert without PI
180 with SDP	Alert with PI

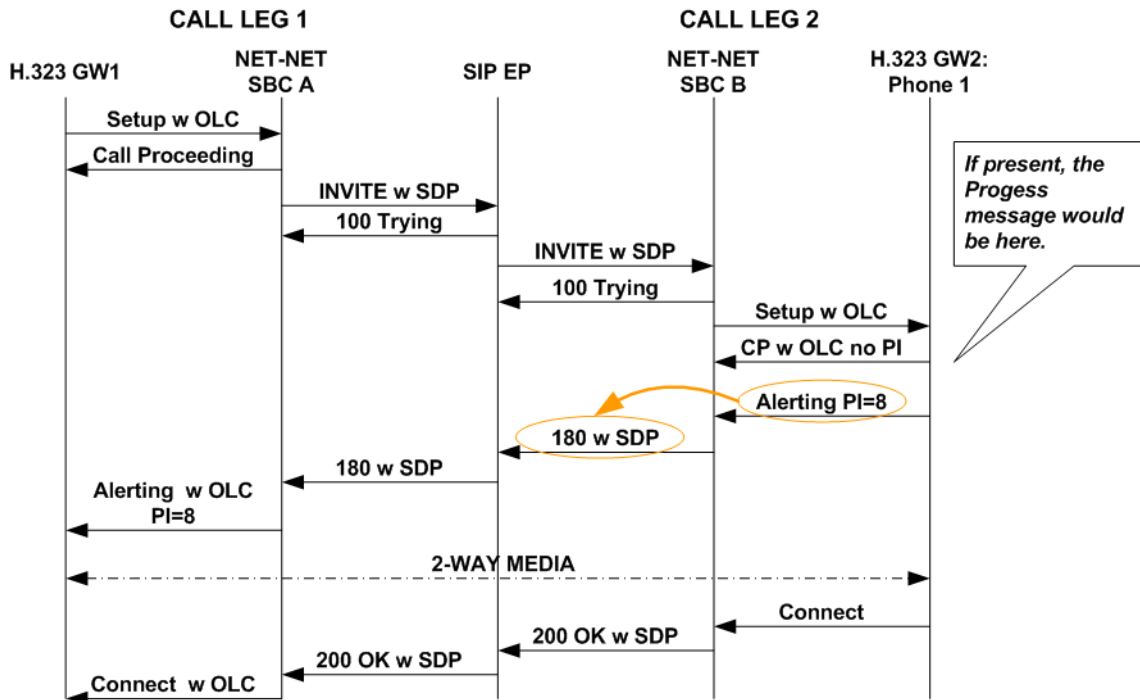
In the following diagram, a call that requires the IWF passes through the Oracle Communications Session Border Controller twice, creating two call legs. The call originates from H.323 GW1 and terminates in Phone 1 or Phone 2.



**Sample 1 In-band Ringback without Progress Message**

This sample flow shows how the Oracle Communications Session Border Controller handles a call that requires the IWF where there is no progress message. In this call flow, there is a progress indicator of eight (8), meaning that ringback is in-band.

In this diagram, you can see that the Oracle Communications Session Border Controller maps the progress indicator included in the Alerting message sent from Phone 1 through H.323 GW2 to a SIP 180 message with SDP. When the Progress message appears, it contains the progress indicator rather than the Alerting message containing it.

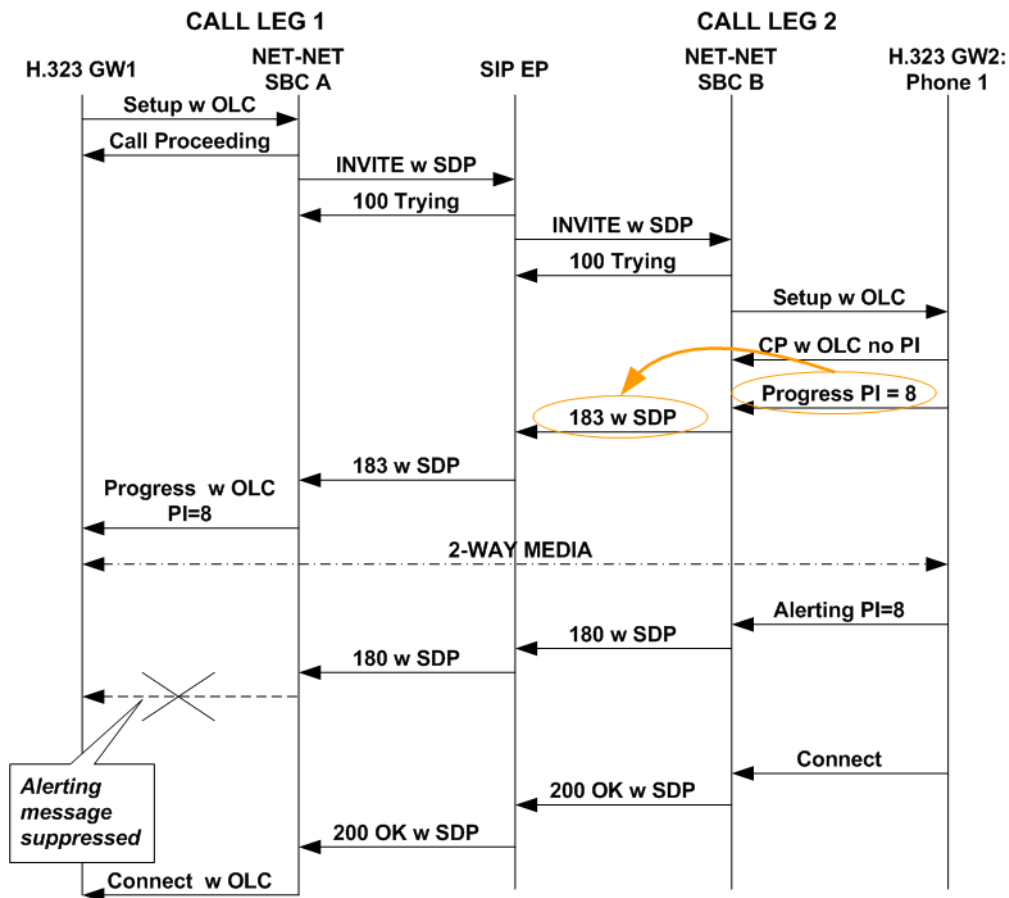


**Sample 2 In-band Ringback with Progress Message**

This sample flow shows how the Oracle Communications Session Border Controller handles a call that requires the IWF where there is a progress message. In this call flow, there is a progress indicator of eight (8), meaning that ringback is in-band.

For this call flow, you can see again that the Oracle Communications Session Border Controller maps the progress indicator included in the alerting message sent from Phone 1 through H.323 GW2 to a SIP 180 message with SDP. Note that now the Progress message contains the progress indicator.

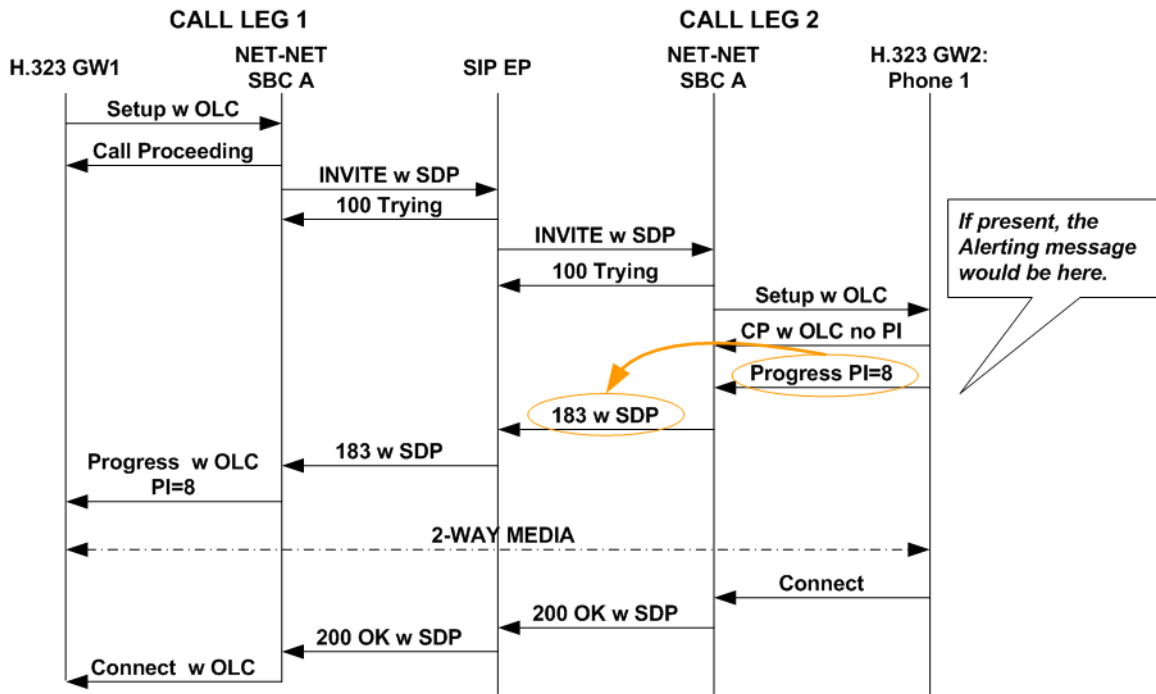




### Sample 3 In-band Ringback without Alerting Message

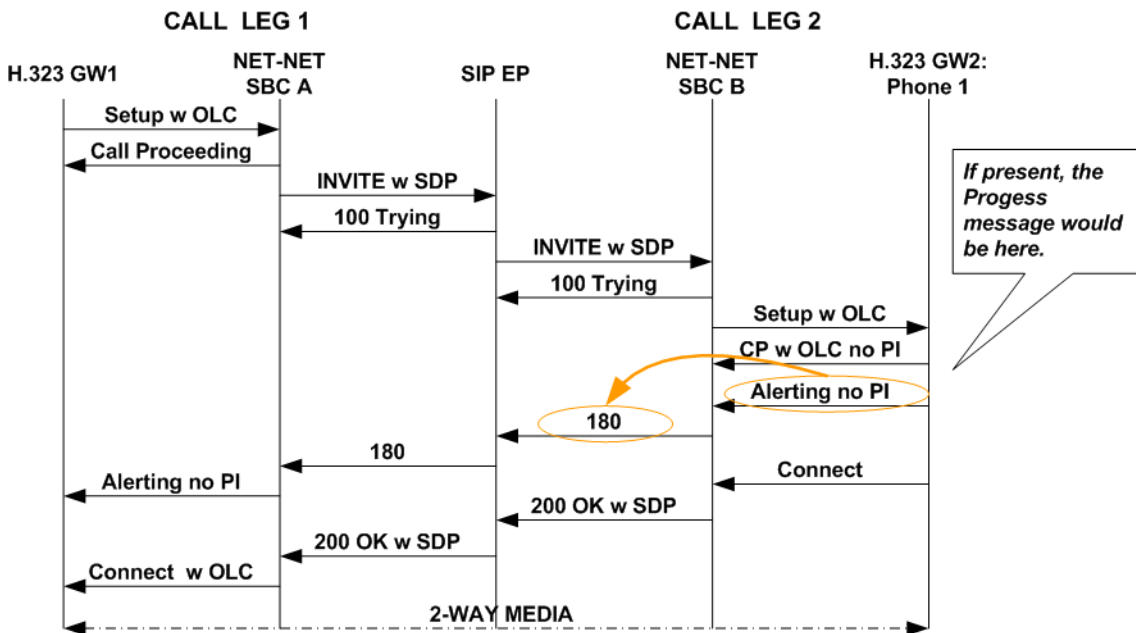
This sample flow shows how the Oracle Communications Session Border Controller handles a call that requires the IWF where there is no progress message. In this call flow, there is a progress indicator of eight (8), meaning that ringback is in-band.

In this diagram, you can see that the Oracle Communications Session Border Controller maps the progress indicator included in the Progress message sent from Phone 1 through H.323 GW2 to a SIP 180 message with SDP. When the Alerting message appears, it contains the progress indicator rather than the Progress message containing it.



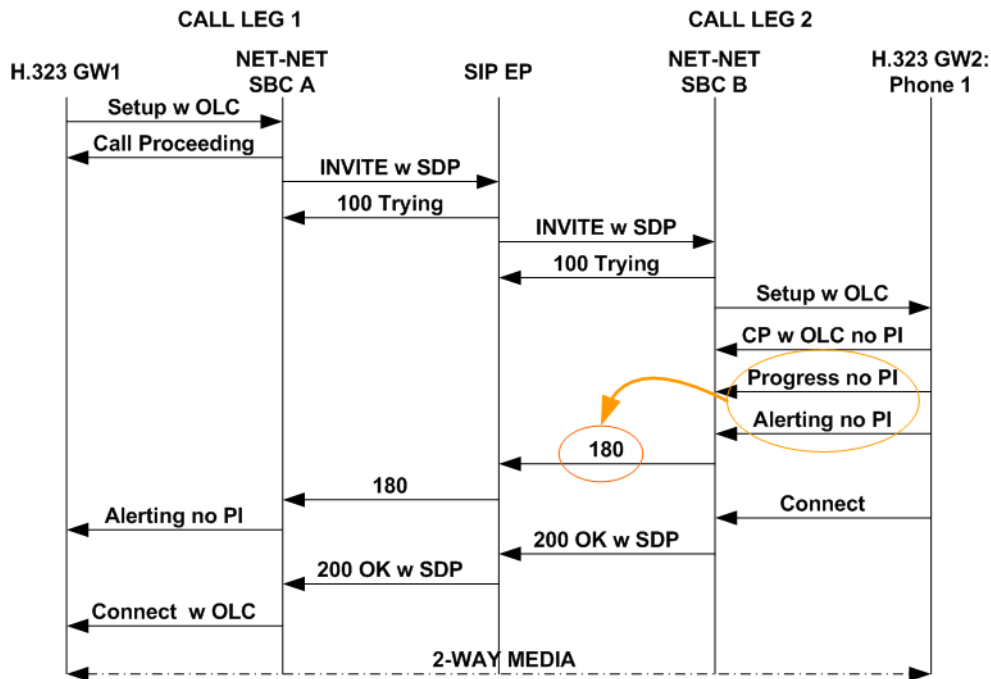
**Sample 4 Out-of-band Ringback without Progress Message**

When there is no progress indicator included in the Alerting message, then there is out-of-band ringback. The system maps the Alerting message to a SIP 180, but it does not include SDP in the SIP 180. This call flow shows that there is no Progress message and that media cannot be set up until after H.323 Connect and SIP messages are sent.



**Sample Flow 5 Out-of-band Ringback with Progress Message**

When there is no progress indicator included in either the Alerting or Progress messages, then there is out-of-band ringback. The system maps the Alerting message to a SIP 180, but it does not include SDP in the SIP 180. This call flow shows includes the Progress message; still, media cannot be set up until after H.323 Connect and SIP messages are sent.

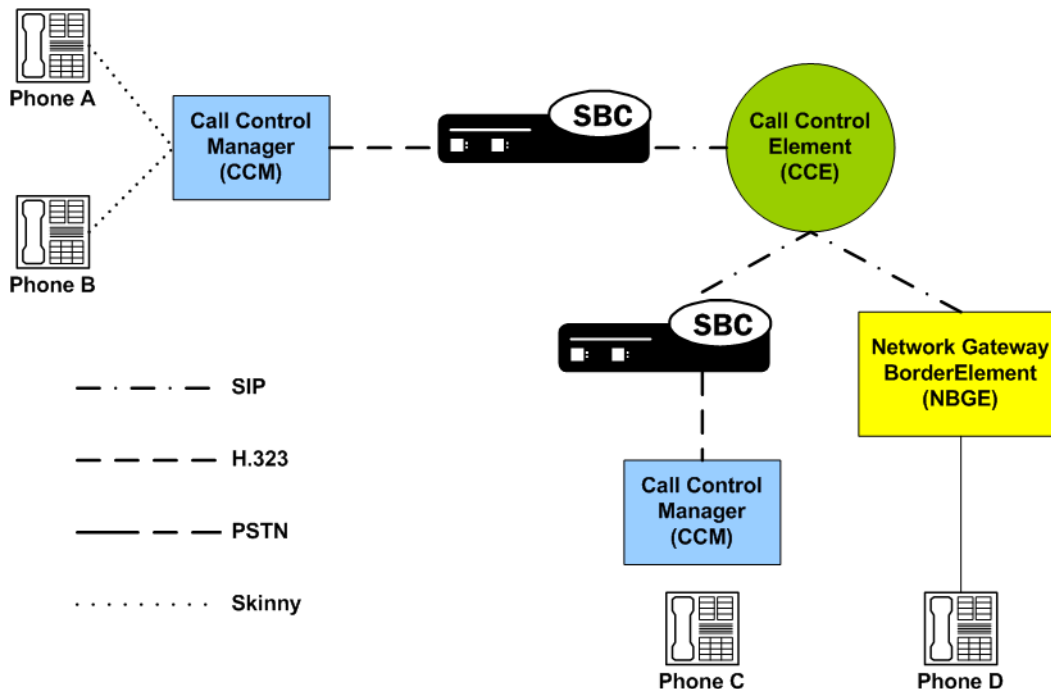


### H.323 Endpoint-Originated Call Hold and Transfer

When calls that require the IWF originating in H.323, the Oracle Communications Session Border Controller supports call hold, transfer, and conference for the H.323 call leg. The call hold and transfer feature uses signaling procedures based on the ITU-T recommendations/H.323 specification for third party initiated pause and rerouting.

You do not have to configure the Oracle Communications Session Border Controller’s call hold and transfer feature.

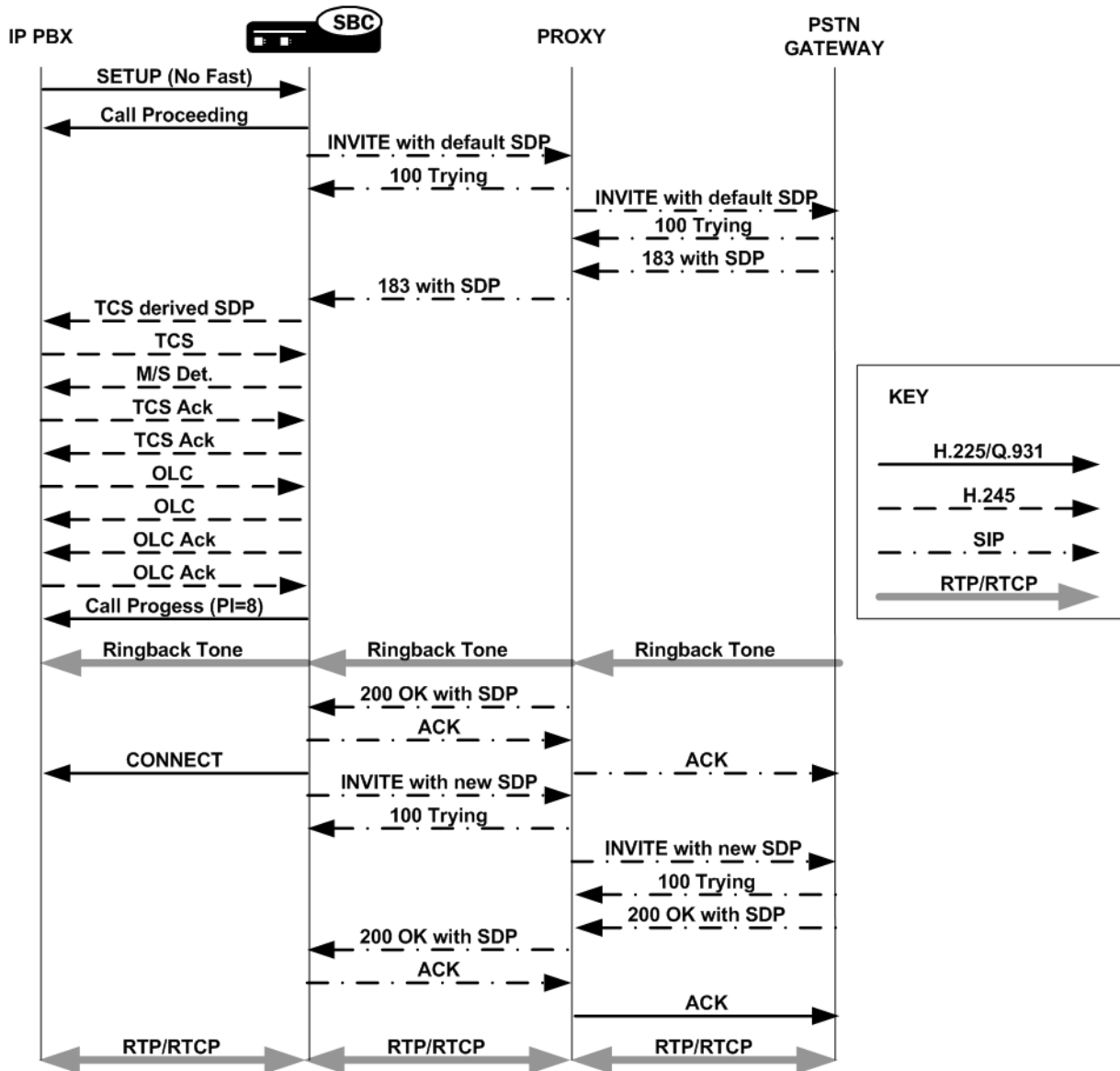
The following diagram shows how the Oracle Communications Session Border Controller provides call hold and transfer support for IWF sessions that originate in H.323. As you review this section’s call flow diagrams, you might want to refer back to the following logical diagram directly below to review the network elements involved, and what protocols they use.



**Basic Call**

In the following sample basic call, IP PBX A sends an H.323 Slow Starts message ultimately destined for the PSTN through the Oracle Communications Session Border Controller . The Oracle Communications Session Border Controller performs translation to SIP and inserts default information about media. Once the PSTN gateway responds with a 183 containing SDP, the Oracle Communications Session Border Controller sends that information to IP PBX A. Then the Oracle Communications Session Border Controller and the IP PBX exchange TCS- and OLC-related messages, and they negotiate master-slave determination. The Oracle Communications Session Border Controller also sends IP PBX A a Call Progress message with a progress indicator of 8.

After the ringback tone, the proxy sends a 200 OK message with SDP to the system. The Oracle Communications Session Border Controller sends a Connect message to the IP PBX A, and then it sends another SIP INVITE to the proxy that contains amended SDP (if that information about media is different from the default). After 200 OK and ACK messages are exchanged, media (RTP/RTCP) flow takes place.

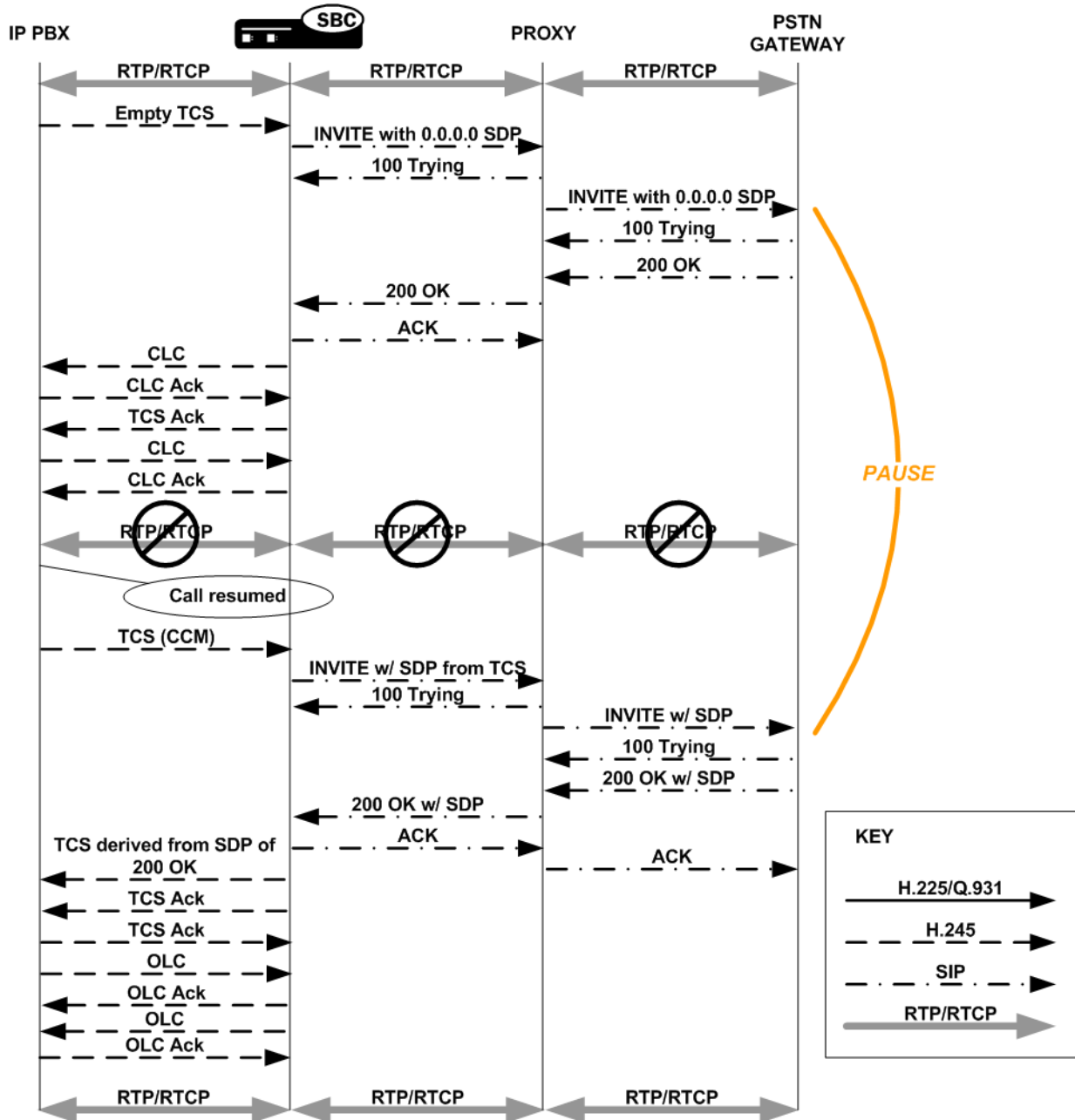


**Hold**

This sample call flow assumes that the IWF call is established and that the RTP/RTCP flow is already in progress. The hold button is pushed, and IP PBX A sends an empty TCS to the Oracle Communications Session Border Controller . The Oracle Communications Session Border Controller puts the called party on hold by sending an INVITE message with 0.0.0.0 SDP to the SIP side of the call. Using 0.0.0.0 as the media address effectively stops the

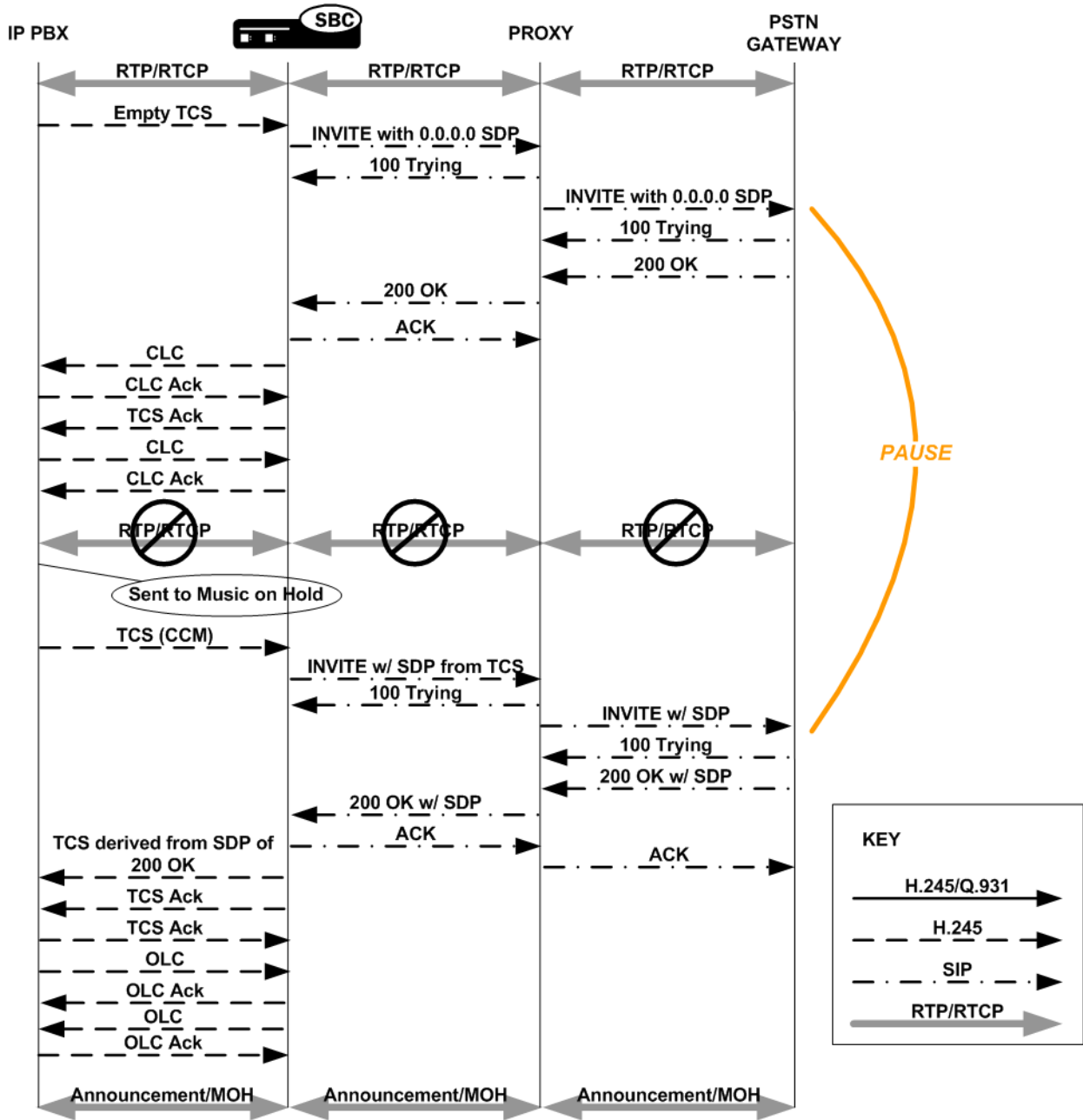
media flow. This INVITE is acknowledged, and the Oracle Communications Session Border Controller closes the channels on the H.323 side, halting the RTP/RTCP flow.

When the caller on the H.323 side takes the call off hold, it resumes with a TCS that the Oracle Communications Session Border Controller receives and then translates on the SIP side as an INVITE with SDP. After that INVITE is acknowledged and received, the Oracle Communications Session Border Controller opens logical channels on the H.323 side and RTP/RTCP flows resume.



### Music On Hold

This scenario is similar to the hold feature enabled for calls that require the IWF, except that after the RTP/RTCP flow between the H.323 and SIP sides stops, the call is sent to music on hold. Before the announcement or music plays, the Oracle Communications Session Border Controller sets up the necessary support for media to be exchanged.



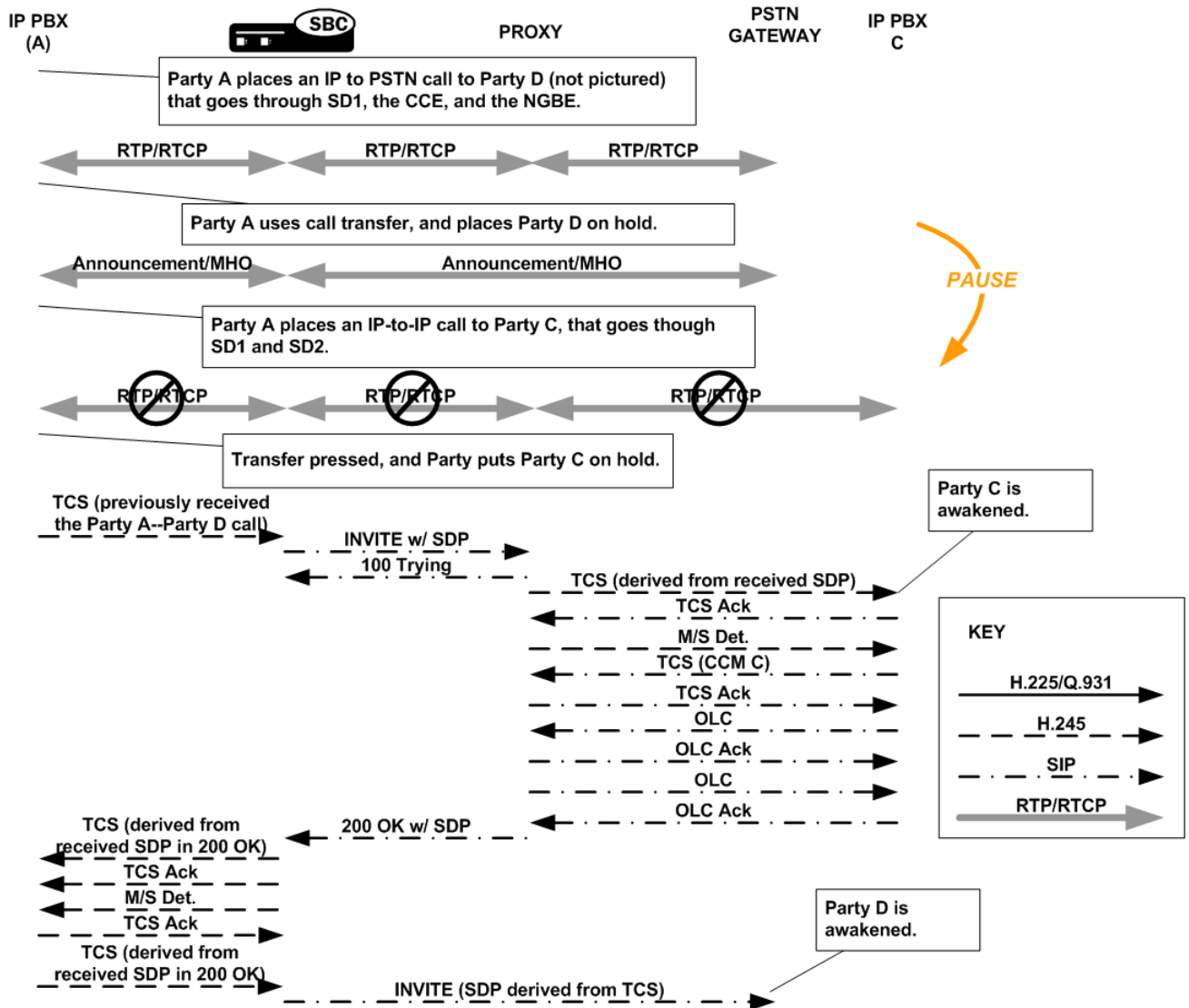
**Transfer**

The call flow described in this section recalls the diagram at the top of the H.323 Endpoint-Originated Call Hold and Transfer section, where endpoints A, B, and C are H.323 devices and endpoint D is a SIP device. When you follow the signaling and media flows, note that there are two Oracle Communications Session Border Controller s in the call transfer and two sets of SIP/H.323 translations that take place. The first Oracle Communications Session Border Controller translates H.323 to SIP, and the second performs the same operations with the protocols reversed.

In the scenario pictured, Party A is on a call with Party D, but wants to transfer Party C to Party D. Party A places Party D on hold, and then makes the call to Party C. Party A then puts Party C on hold, pressing the transfer button. You can see that Oracle Communications Session Border Controller 1 receives a TCS from the IP PBX, which is then translated to SIP. Oracle Communications Session Border Controller 2 receives it, performs the required protocol translations, and then opens a session with Party C via another IP PBX. Once this session is up and Party D is awakened, channels are established for media exchange.

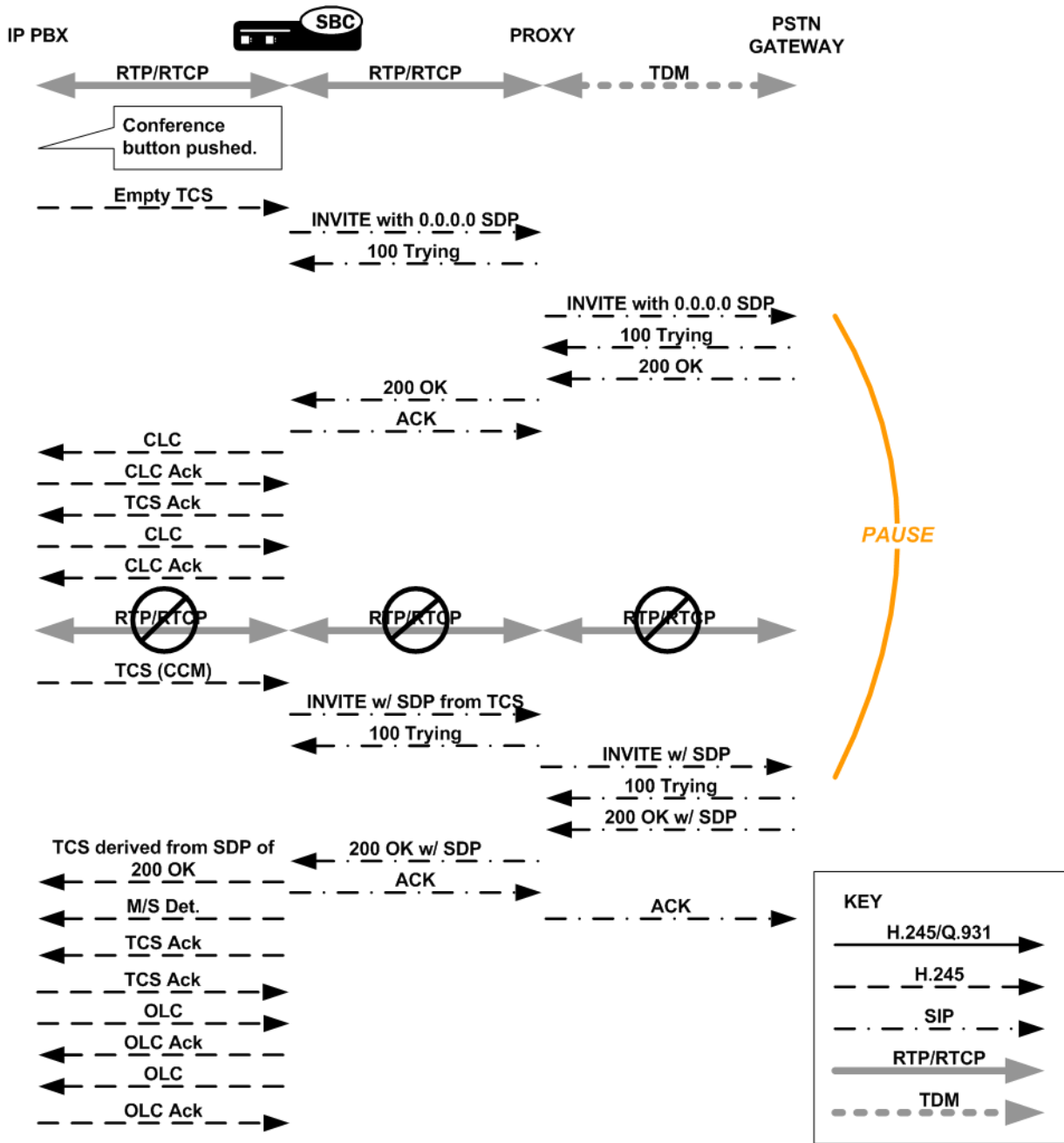
In order to redirect the media so that it flows between Party C and Party D, the Oracle Communications Session Border Controller 1 and IP PBX C exchange OLC and OLC Ack messages that contain address information for Party C and for Party D. Address information for both parties is contained in the OLC Ack messages that the Oracle Communications Session Border Controller exchanges with the IP PBX. IP PBX A does not move forward with the call until it has the necessary address information.

Even though Party A's participation in the call stops early in this scenario, the IP PBX with which it is associated keeps the signaling sessions with the Oracle Communications Session Border Controller alive to manage the transfer.



## Conference

To conference a call that requires the IWF that starts in H.323, the Oracle Communications Session Border Controller uses a scenario much like the one used for holding a call that requires the IWF. Here again, the INVITE with 0.0.0.0 as the media address and the closing of logical channels stops the flow of RTP/RTCP. After signaling and SDP/media information are re-established, RTP/RTCP for the conference flows.



### IWF Call Forwarding

This section describes the Oracle Communications Session Border Controller’s IWF Call Forwarding feature, which is supported for calls initiated in SIP that require interworking to H.323.

Prior to the implementation of this feature, the Oracle Communications Session Border Controller did not forward calls when the remote H.323 endpoint sent a Facility message with Call deflection as the reason and an alternate address for forwarding. Instead, it would either:

- Fail to release the initial call and initiate the forwarded call
- Drop the entire call when the remote endpoint for the call tore down the session

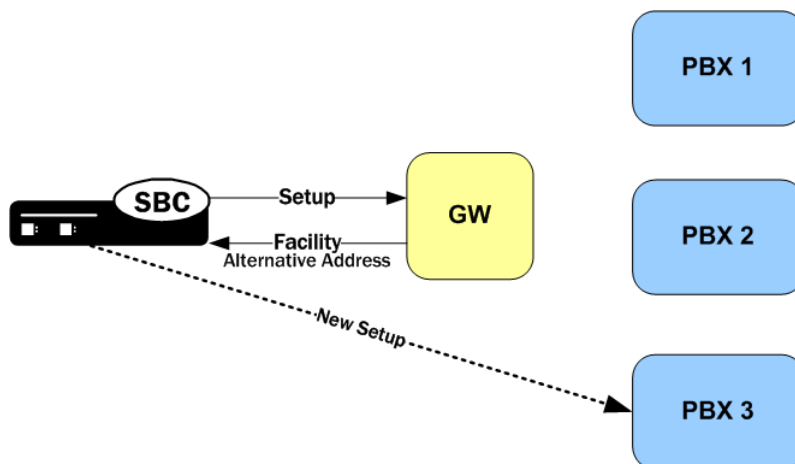
#### New Behavior

In the diagram below, you can see that the Oracle Communications Session Border Controller sends the initial Setup message to the gateway, and the gateway returns the Facility message with an alternate address for forwarding. Rather



than engaging in its former behavior, the Oracle Communications Session Border Controller now releases the call with the gateway and sends a new Setup to the alternate address from the Facility message.

This new Setup up has no effect on the first call leg, which remains connected.



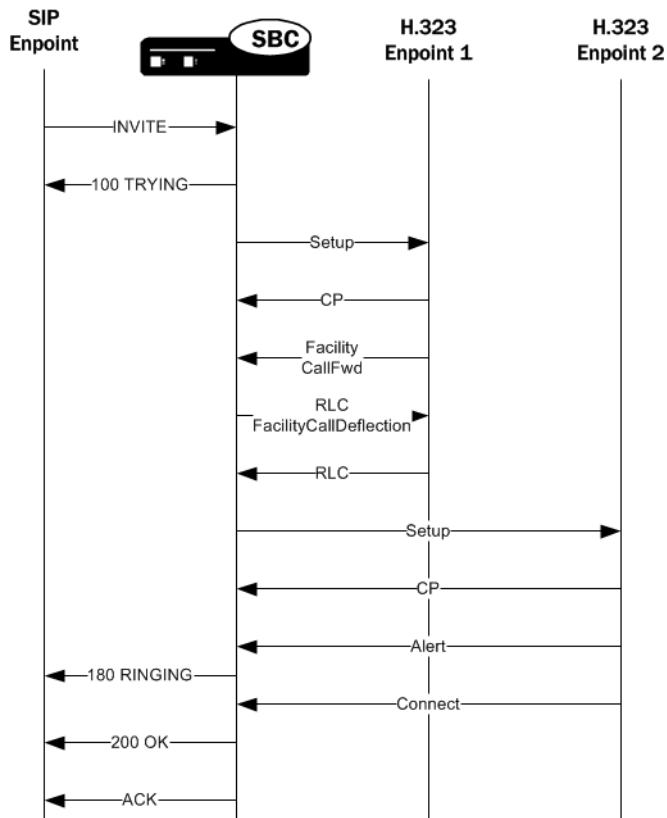
When it receives a Facility message with the reason CallForwarded, the Oracle Communications Session Border Controller looks for an alternate transport address in the Facility's alternativeAddress or alternativeAliasAddress element. The Oracle Communications Session Border Controller releases the egress call with the reason facilityCallDeflection. Then it takes one of two courses of action:

- If it does not find an alternative address, the Oracle Communications Session Border Controller releases the ingress call (with 486 BUSY HERE for a call being interworked from SIP to H.323).

If it finds an alternative address and the egress call has not been alerted or answered, the Oracle Communications Session Border Controller at this point tries to initiate a new egress call. The Oracle Communications Session Border Controller uses the alternative alias address to populate the calledPartyNumber information element (IE) and the destination address of the new Setup.

### H.323 Sample Call Flow

The following diagram shows how the H.323 Call Forwarding feature works in a purely H.323 environment.



### Media Release for H.323 SS-FS Calls for IWF

When the Oracle Communications Session Border Controller routes a slow-start to fast-start call, it is possible for the same fast-start call to be routed back through the Oracle Communications Session Border Controller making for a hairpin flow. If it does become a hairpin flow, then the Oracle Communications Session Border Controller routes it to its destination as a fast-start to fast-start call. This can result in one-way media if:

- The destination of the hairpin call is in the same realm as the originating slow-start to fast-start call
- The realm reference in the first bullet item is configured to disable in-realm media management
- The called endpoint accepts the proposed fast-start logical channels

The enhancements to the Oracle Communications Session Border Controller’s behavior described in this section show how the Oracle Communications Session Border Controller follows additional procedures when setting up a hairpin flow to avoid one-way media when media release occurs.

#### H.323

For H.323 calls, the Oracle Communications Session Border Controller establishes media using the H.245 procedures described in the H.245 ITU-T recommendation: control protocol for multimedia communication. It also uses the Fast Connect procedure defined in the H.323 ITU-T recommendation: packet-based multimedia communication systems.

The latter ITU-T recommendation allows a calling endpoint to send a Setup message that contains a fastStart element, a sequence of OLC structures that describe the calling endpoint’s proposed forward/reverse logical channels. If the called endpoint accepts this proposal, then logical channels are established.

When the Oracle Communications Session Border Controller translates a call originating in slow-start to fast-start, it uses a Fast Connect procedure in the outgoing leg by sending an outgoing Setup that includes a fastStart element with one or more OLC structures. But when the Oracle Communications Session Border Controller constructs this message, it is unaware of whether the call will become hairpinned or if media release will occur. Because it does not yet have this information, the Oracle Communications Session Border Controller sets the Network Address and the TSAP identifier in the OLC structures to the ingress IP address and port of a corresponding media flow allocated for media traveling between the calling and called endpoints. So if the called endpoint accepts the fastStart the Oracle

Communications Session Border Controller proposes, the called endpoint would send its media to the Oracle Communications Session Border Controller. After acceptance, the system starts H.245 procedures on the slow-start side of the call to set up logical channels on that side. Then the Oracle Communications Session Border Controller updates the IP address and port of the media flows using OLC and OLCAck messages received from the calling endpoint.

This procedure works well for endpoints that are not in the same realm, or that are in the same realm for which media management is disabled, because each endpoint must send its media through the Oracle Communications Session Border Controller. When the endpoints are in the same realm and when media management is enabled, however, the Oracle Communications Session Border Controller must perform additional steps for media release in slow-start to fast-start calls.

To support media release in slow-start to fast-start calls, the Oracle Communications Session Border Controller performs a hold-and-resume procedure on the fast-start side. After it establishes channels on the slow-start side and if it detects media release being enabled, the Oracle Communications Session Border Controller sends an empty TCS to the fast-start side to put that side on hold. Then the called endpoint closes all the logical channels it previously opened in the Fast Connect procedure and stops transmitting to them. And the Oracle Communications Session Border Controller also closes its logical channels. Once the channels are closed, the Oracle Communications Session Border Controller resumes the call by sending a new, restricted TCS to the fast-start side. The restricted TCS only contains the receive and transmit capabilities of the codec types that the called endpoint accepted in the Fast Connect procedure, and it forces the called endpoint to re-open logical channels of the same codec types accepted in the Fast Connect procedure. Once it receives an OLC from the called endpoint, the Oracle Communications Session Border Controller sends an OLCAck with the Network Address and TSAP identifier for the logical channel from the calling endpoint. Then the Oracle Communications Session Border Controller re-opens logical channels (of the same codec types that it opened in the Fast Connect procedure). If the called endpoint has not changed its Network Address and TSAP identifier for its logical channels, media is re-established after the Oracle Communications Session Border Controller and the called endpoint exit the hold state. The last step is for the Oracle Communications Session Border Controller to re-send the full TCS message from the calling to the called endpoint to inform the called endpoint of the full capabilities of the calling endpoint.

### Hold-and-Resume Procedure

The hold-and-resume procedure has three states:

- **Media Hold**—Starts when the Oracle Communications Session Border Controller sends the empty TCS to the called endpoint to put it on hold.

When it detects media release, the Oracle Communications Session Border Controller puts the called endpoint on hold. It can only do so if it has exchanged the TCS/TCSAck messages and completed master-slave determination with the calling endpoint.

When the Oracle Communications Session Border Controller receives a TCSAck in response to the empty TCS that it sent to the called endpoint, it closes the logical channels it opened as part of the Fast Connect procedure; the called endpoint likewise closes its logical channels. The two then exchange CLC and CLCAck messages, which signals the start of the Media Resume state.

- **Media Resume**—Starts when the Oracle Communications Session Border Controller sends a restricted TCS to resume the call.

The restricted TCS the Oracle Communications Session Border Controller sends contains only the receive/transmit capabilities of the codec types previously accepted by the called endpoint in the Fast Connect procedure. This forces the called endpoint to re-open logical channels of the same codec type that were previously accepted in the Fast Connect procedure.

After sending this TCS, the Oracle Communications Session Border Controller is ready (as specified in the ITU-T recommendations) to take part on the master-slave determination (MSD) process. However, the called party and not the Oracle Communications Session Border Controller initiates the MSD if it is required. The MSD is completed if necessary. Alternately, the called endpoint can start to re-open its logical channels. When it receives the first OLC from the called endpoint, the Oracle Communications Session Border Controller also starts to re-open its logical channels.

- **Media Complete**—Starts when all the logical channels that the Oracle Communications Session Border Controller re-opens are acknowledged by the called endpoint.

When it enters the Media Complete state, the Oracle Communications Session Border Controller updates the called endpoint with the full capabilities of the calling endpoint by sending the full TCS.

### Additional IWF Steps

For calls originating in slow-start H.323 that require interworking to SIP, the Oracle Communications Session Border Controller also takes additional steps for media release in hairpinned flows that the Oracle Communications Session Border Controller routes as SIP to fast-start H.323.

For such a call, after the Oracle Communications Session Border Controller has established logical channels on the slow-start H.323 side of the call, it sends a reINVITE on the SIP side. This reINVITE has an updated session description to correct the media connection information. The Oracle Communications Session Border Controller performs the hold-and-resume procedure on the fast-start side of the call. This procedure re-establishes the logical channels between the Oracle Communications Session Border Controller and the called endpoint, avoiding the one-way media problem.

When you are configuring H.323 globally on your Oracle Communications Session Border Controller, you might choose to set the noReInvite option. This option stops the Oracle Communications Session Border Controller from sending a reINVITE after the logical channels are established on the slow-start H.323 side of the call. Instead, the Oracle Communications Session Border Controller's H.323 task communicates internally with its own SIP task a SIP UPDATE message that corrects the SDP; then the SIP task updates media flow destinations. But the Oracle Communications Session Border Controller does not send the UPDATE to the next hop, which can result in the one-way media problem if the call is hairpinned and media release occurs. For such cases, the default behavior for the noReInvite option is overridden. When the Oracle Communications Session Border Controller detects media release in an H.323-SIP call, it forwards the UPDATE to the next hop even when you enable the noReInvite option.

### Dependencies

This feature depends on:

- The H.323 endpoint supports the third-party-initiated pause and re-routing feature.
- The H.323 endpoint does not change its Network Address and TSAP identifier when it re-opens the logical channels.
- The H.323 endpoint does not immediately tear down the call when there is not established logical channel in the call.
- The fact that the SIP endpoint supports the UPDATE message if the noReInvite option is enabled.

## Before You Configure

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The Oracle Communications Session Border Controller's IWF requires that there be complete configurations for both SIP and for H.323. These two sets of configurations function together when the interworking is configured and enabled.

You enable the Oracle Communications Session Border Controller's interworking capability when you set the IWF configuration's state parameter to enabled, and all required H.323 and SIP configurations are established. This means that all of the following configurations must be established:

- A full SIP configuration, including SIP interfaces, SIP ports, SIP-NATs (if needed), and SIP features
- A full H.323 configuration, including H.323 global and H.323 interface configurations
- Local policy and local policy attributes (the IWF will not work without these configurations)
- Media profiles
- Session agents and, if needed, session agent groups

## H.323 Configuration

You must have a complete configuration to support H.323 traffic on your Oracle Communications Session Border Controller, including any required support for H.323 Fast Start or Slow Start.

In the H.323 interface configuration, you are able to configure interfaces that enable communication between H.323 devices (for audio, video, and/or data conferencing sessions).

If you know that your Oracle Communications Session Border Controller will be handling traffic originating in Slow Start H.323, you must establish the appropriate list of media profiles in the IWF configuration. Handling Slow Start traffic also requires that you establish appropriate local policy (and local policy attribute) configurations, but configuring session agents and session agent groups is optional.

## SIP Configuration

SIP functionality must also be configured on your Oracle Communications Session Border Controller that will perform IWF translations. You must use appropriate local policy (and local policy attribute) configurations, but configuring session agents and session agent groups is optional. If you use session agents, then you must also configure the information you need for media profiles.

## The Role of Local Policy

You must configure local policies (and local policy attributes, if necessary) in order for translations between SIP and H.323 to take place. These local policies determine what protocol is used on the egress side of a session. Local policy and local policy attribute configurations make routing decisions for the session that are based on the next hop parameter that you set. The next hop can be any of the following:

- IPv4 address of a specific endpoint
- Hostname or IPv4 address of a session agent
- Name of a session agent group

You can use the application protocol parameter in the local policy attributes configuration as a way to signal the Oracle Communications Session Border Controller to interwork the protocol of an ingress message into a different protocol as it makes its way to its egress destination (or next hop).

For example, if you set the application protocol parameter to SIP, then an inbound H.323 message will be interworked to SIP as it is sent to the next hop. An inbound SIP message would travel to the next hop unaffected. Likewise, if you set the application protocol parameter to H.323, then an incoming SIP message will be interworked to H.323 before the Oracle Communications Session Border Controller forwards it to the next hop destination.

The following example shows a configured local policy and its attributes used for IWF traffic.

```
local-policy
  from-address
  to-address
  source-realm
  state
  last-modified-date
  policy-attribute
    next-hop
    realm
    replace-uri
    carrier
    start-time
    end-time
    days-of-week
    cost
    app-protocol
    state
    media-profiles
  *
```

```

  *
  444
  *
  enabled
  2004-04-20 17:43:13
  sag:sag_internal
  internal
  disabled
  0000
  2400
  U-S
  0
  SIP
  enabled
```

### Local Policy in an IWF Session Initiated with H.323

In a session where the Oracle Communications Session Border Controller is interworking H.323 to SIP, it internally forwards the session on for interworking when:

- The next hop in the local policy is configured as a SIP session agent
- The next hop in the local policy is configured as a SIP session agent group
- The next hop in the local policy is not configured as a session agent or session agent group, and the application protocol parameter is set to SIP in the local policy attributes configuration.

### Local Policy in an IWF Session Initiated with SIP

In a session where the Oracle Communications Session Border Controller is interworking SIP to H.323, it internally forwards the session on for interworking when:


- The next hop in the local policy is configured as an H.323 session agent
- The next hop in the local policy is configured as an H.323 session agent group
- The next hop in the local policy is not configured as a session agent or session agent group, and the application protocol parameter is set to H.323 in the local policy attributes configuration

In this case the local policy should also define the egress realm, which you can set in the realm parameter of the local policy attributes configuration.

## SIP-H.323 interworking with Dynamic Payload Types

The SIP and H.323 Protocols use Internet multimedia signaling over IP, and both use the Real-Time Transport Protocol (RTP) for transferring realtime audio/video data. The interworking function (IWF) provides a means of converting translation and signaling protocols and session descriptions between SIP and H.323. However, SIP and H.323 provide different mechanisms when exchanging payload types for media during IWF calls. Therefore, the International Telecommunications Union (ITU) modified the ITU H.245 recommendations in H.245 v16 to include a new “Dynamic Payload Type Replacement” capability that resolves this payload type conflict. This new capability provides a way for an H.323 endpoint to specify the payload type of a media stream for which the endpoint is willing to receive through the OLCacknowledgment (OLC-ACK) message in an audio/video call flow.

The Oracle Communications Session Border Controller supports this new “Dynamic Payload Type Replacement” capability by ensuring interworking of SIP and H.323 when audio/video call flows use dynamic payload types. The Oracle Communications Session Border Controller checks for the presence of this capability in the incoming TCS request. If it finds this capability in the TCS request, it sends an Open Logic Channel Acknowledgement (OLC-ACK) response with the payload type it is willing to receive.

 **Note:** The Oracle Communications Session Border Controller always returns an OLC-ACK with a dynamic payload type value that it received in the incoming Session Description Protocol (SDP) from the SIP endpoint.

For devices that don't support the H.245 v16 recommendations, the Terminal Capability Set (TCS) request from the H.323 endpoint does not have the "Dynamic Payload Type Replacement" capability present. Therefore, the Oracle Communications Session Border Controller rewrites the payload type within the RTP packets when these packets traverse the Oracle Communications Session Border Controller. When devices in a session negotiate different payload types between SIP and H.323 packets, the RTP streams that they receive, always have the expected payload type in the RTP header.


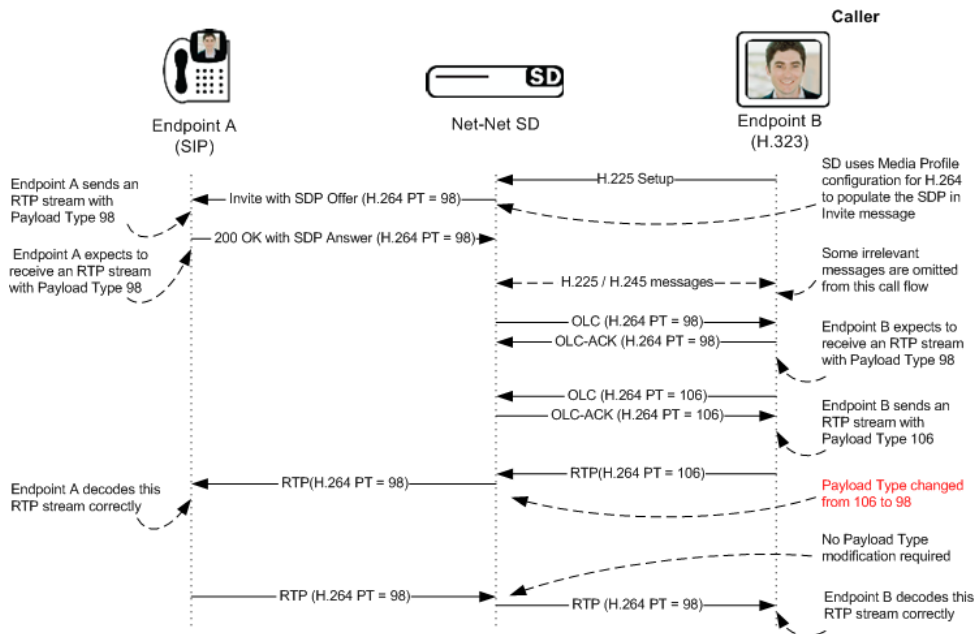
 **Note:** The Oracle Communications Session Border Controller always maps the payload type on the RTP stream received from the H.323 endpoint, and sends it to the SIP endpoint for both audio and video. The Oracle Communications Session Border Controller does not support mapping of payload types in audio streams with 2833 DTMF packets.

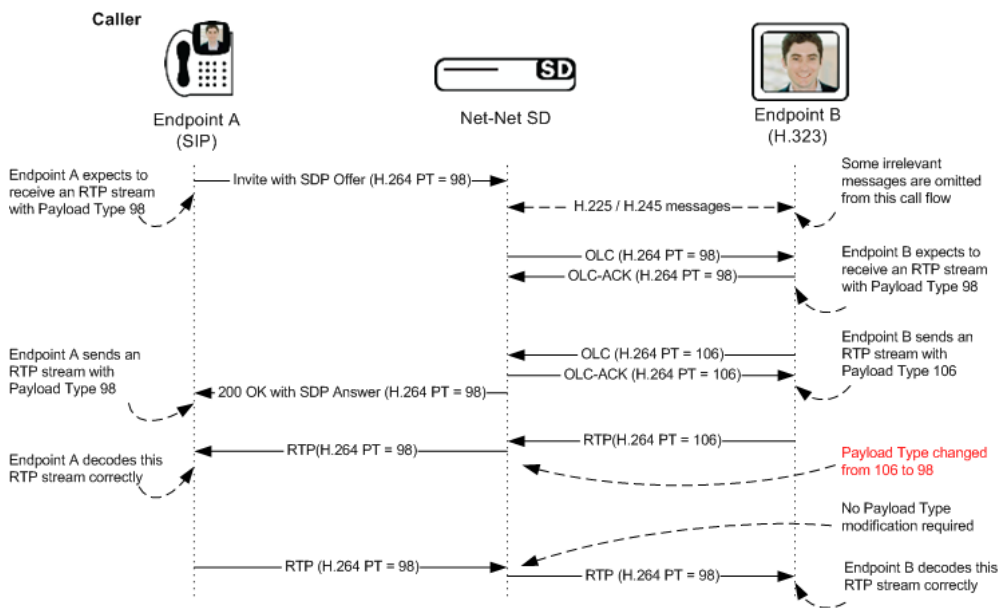
Figure 1a and 1b below shows the call flow from an H.323 Endpoint B to a SIP Endpoint A, and from a SIP Endpoint A to an H.323 Endpoint B, respectively. These illustrations show the negotiation of different dynamic payload types for the video streams but the Codec negotiated is the same. The Oracle Communications Session Border Controller dynamically replaces the payload type in the RTP header of the video stream received from the H.323 endpoint.

Figure 1a Endpoint B calling Endpoint A (H.323 endpoint does not have “Dynamic Payload Type Replacement Capability”)



The H.323 Endpoint B is not H.245 v16 compliant, and hence payload type replacement needs to be done in the RTP packets.

Figure 1b Endpoint A calling Endpoint B (H.323 endpoint does not have Dynamic Payload Type Replacement Capability)

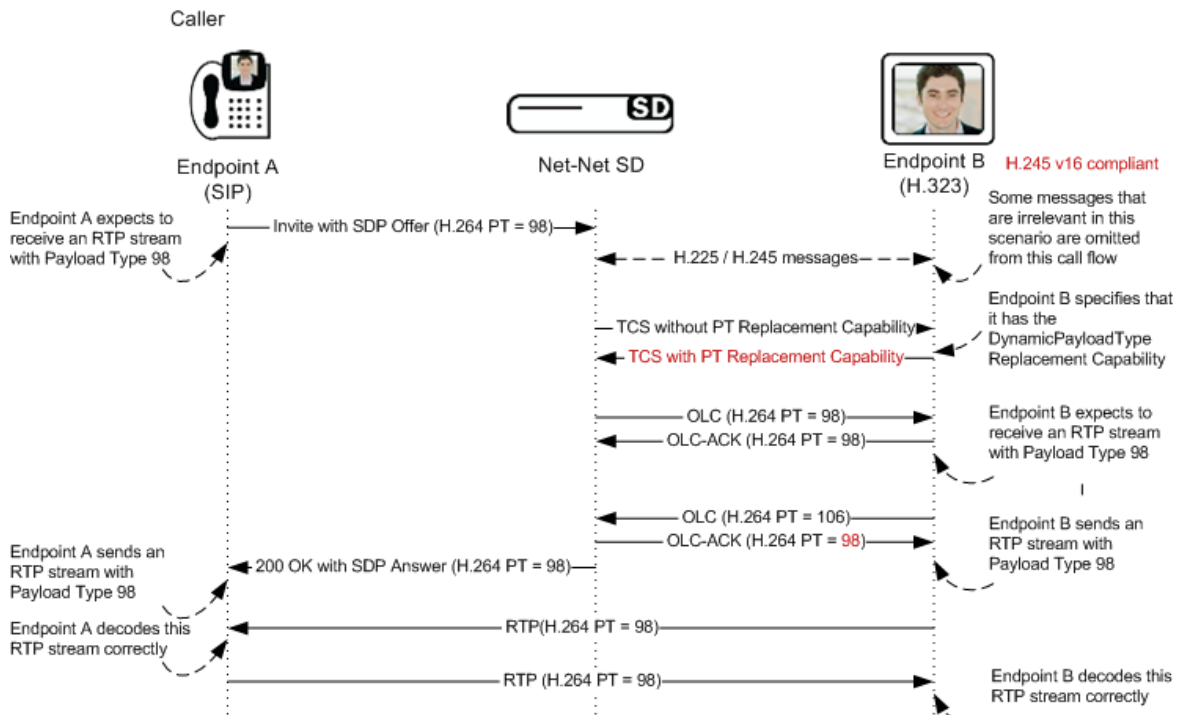


The H.323 Endpoint B is not H.245 v16 compliant, and therefore payload type replacement needs to be done in the RTP packets.

There is no concept of H.245 compliance for the SIP Endpoint A.

Figure 2a shows the call flow of SIP Endpoint A calling an H.323 Endpoint B using slow start. The Oracle Communications Session Delivery Manager modifies the dynamic payload type in the OLC-ACK based on payload type received in the incoming SDP OFFER in the "INVITE" message.

Figure 2a Endpoint A calling Endpoint B (H.323 endpoint has TCS with Dynamic Payload Type Replacement Capability)

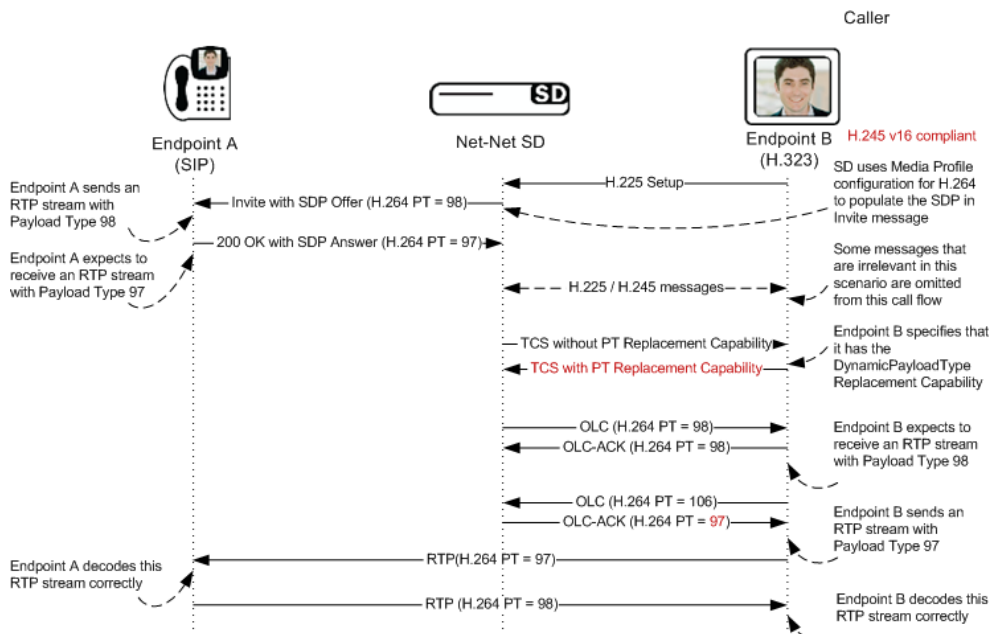


The H.323 Endpoint B is H.245 v16 compliant.

There is no concept of H.245 compliance for the SIP Endpoint A.

Figure 2b shows the call flow an H.323 Endpoint B using slow start, calling a SIP Endpoint A. The Oracle Communications Session Delivery Manager modifies the dynamic payload type in the OLC-ACK based on payload type received in the incoming SDP ANSWER in the "200 OK" message.

Figure 2b Endpoint B calling Endpoint A (H.323 endpoint here has TCS without “Dynamic Payload Type Replacement” Capability)





## Configuring Interworking

If you have already completed the steps outlined in this chapter's IWF Service Enhancements section, then enabling the IWF is a simple process. This section shows you how to enable the IWF, and how to enable certain features that you can use to supplement basic IWF functionality.

An IWF configuration might appear like this in the ACLI:

```
iwf-config
  state                enabled
  media-profiles
                      PCMU
                      telephone-event
  logging              disabled
```

### IWF Configuration

To enable the IWF on your Oracle Communications Session Border Controller :

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **iwf-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # iwf-config
```

From this point, you can configure IWF parameters. To view see all IWF configuration parameters, enter a ? at the system prompt.

4. **state**—Enable this parameter if you want to translate SIP and H.323 sessions on your Oracle Communications Session Border Controller . The default value is **disabled**. Valid values are:
  - enabled | disabled
5. **media-profiles**—Enter the name of the media profiles you want to use for IWF translations. This name is either the name of an SDP codec (such as PCMU), or it can be telephone-event if you are configuring your system for DTMF support.

If you want to use more than one media profile for SIP/H.323 translations, enter the names in quotation marks with a space between each one.

```
ACMEPACKET (iwf-config) # media-profiles "PCMU telephone-event"
```

6. **logging**—Enable this parameter if you want the Oracle Communications Session Border Controller to log SIP messages that are related to the IWF. The default value is **disabled**. Valid values are:
  - enabled | disabled

## Topology Hiding for IWF with an Internal Home-Realm

You can configure the Oracle Communications Session Border Controller to mask the IP address of the originating caller in the SIP From and/or P-Asserted-Identity headers when calls are placed from H.323 to SIP endpoints.

The option **NoPAssertedID** checks for incoming SETUP messages have the presentation indicator set to restricted and instructs the Oracle Communications Session Border Controller to send a Privacy header without the P-Asserted-Identity and not to make the From header anonymous.

The option **replace-call-origin-ip** removes the calling party's IP address in the SIP From header. The IP address from the internal home realm is used instead.

## IWF Services

The topology hiding feature uses the presentation indicator field from an inbound H.323 setup message to determine if/how the headers will be masked. The following table summarizes the configurable Oracle Communications Session Border Controller parameters and the values for the From and P-asserted-identity headers.

H.255 Presentation Indicator Setting	P-Asserted-Identity Header Value	From Header Value	SD Session Agent Option
Allow	IP address from home realm of SD SIP Config	H.255 Calling Party IP/Port	No Option Set
Allow	IP address from home realm of SD SIP Config	IP address of Home Realm SIP-Interface	NoPAssertedID
Allow	IP address from home realm of SD SIP Config	IP address of Home Realm SIP-Interface	replace-call-origin-ip
Allow	IP address from home realm of SD SIP Config	IP address of Home Realm SIP-Interface	replace-call-origin-ip, NoPAssertedID
Restricted	PAI Header not present	Anonymous	No Option Set
Restricted	PAI Header not present	Anonymous	NoPAssertedID
Restricted	PAI Header not present	Anonymous	replace-call-origin-ip
Restricted	PAI Header not present	Anonymous	NoPAssertedID, replace-call-origin-ip

## IWF Topology Hiding Configuration

To enable IWF topology hiding on your Oracle Communications Session Border Controller :

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **sip-config** and press Enter. The system prompt changes.

```
ACMEPACKET (session-router) # session-agent  
ACMEPACKET (session-agent) #
```

4. Use the ACLI **select** command so that you can work with the session agent configuration to which you want to add these options.

```
ACMEPACKET (session-agent) # select
```

5. **options**—Set the options parameter by typing **options**, a Space, the option name preceded by a plus sign (+) (**replace-call-origin-ip**), and then press Enter. Follow the same steps to add the **NoPAssertedID** option.

```
ACMEPACKET (session-agent) # options +replace-call-origin-ip  
ACMEPACKET (session-agent) # options +NoPAssertedID
```

If you type either of these options without the plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign as shown in the example above.

## DTMF Support

For calls that require the IWF, you can enable support for the relay of RFC 2833 DTMF digits. The availability of this feature means that the Oracle Communications Session Border Controller is compliant with RFC 2833, which defines two payload formats for DTMF digits. To learn more about this RFC, refer to <http://www.ietf.org/rfc/rfc2833.txt>.

Until the exchange of TCS messages with the H.323 endpoint, the Oracle Communications Session Border Controller has no information about the endpoint's RFC 2833 capabilities. The Oracle Communications Session Border Controller adds telephone-event to the SDP on the SIP side of the call.

For calls that require SIP/H.323 translation, you can enable support for the relay of RFC 2833 DTMF digits.

To use this feature, you need to configure a media profile called telephone-event and set relevant parameters for it. Application of the media profile can happen either in a session agent configuration or in the IWF configuration.

- The **name** parameter in the media profiles configuration
- The **media-profiles** list in the IWF configuration
- The **media-profiles** list in the session agent configuration

All of the scenarios outlined here assume that you have established a telephone-event media profile configuration.

You can configure DTMF support using the following parameters. The way that the Oracle Communications Session Border Controller uses these values is described below. The payload type, part of the media profiles configuration, is dynamic and varies with different endpoints, so there is no default configuration for the telephone-event media profile.

The telephone-event media profile is used as follows in these types of IWF sessions:

- **Calls that require the IWF originating in H.323 Slow Start**—There is no channel (media) information available on the H.323 side.
  - If the incoming H.323 endpoint is configured as a session agent on the Oracle Communications Session Border Controller, then the telephone-event parameter in the media profiles set for that session agent configuration will be used in the SDP on the SIP side of the session.
  - If the H.323 endpoint is not a session agent or the telephone-event media profile is not configured in the session agent configuration corresponding to the endpoint, then the Oracle Communications Session Border Controller refers to the media profile information configured for the IWF configuration.
- **Calls that require the IWF originating in SIP**—If the TCS was not exchanged before a 200 OK was sent on the SIP side, the Oracle Communications Session Border Controller will behave in one of these two ways.
  - If the outbound H.323 endpoint is configured as a session agent, then the media profiles from that session agent configuration will be used.
  - If the outbound H.323 endpoint is not configured as a session agent, the media profile configured within the IWF configuration with the telephone-event value will be used.

As mentioned above, DTMF support is configured by using a combination of the telephone-event media profile and either the session agent or IWF configuration. First you set up the media profile, then you apply it to a session agent or to the IWF configuration.

## DTMF Configuration

DTMF support requires you to configure a media profile named telephone-event. This section shows you how to set it up.

To configure a telephone-event media profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **media-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # media-profile
```

From this point, you can configure parameters for media profiles. To view see all media profile configuration parameters, enter a ? at the system prompt.

4. **name**—Enter the name telephone-event and press Enter.
5. **parameters**—Enter the parameters to be applied for the codec; these are the digits that endpoints can support.
6. **media-type**—Leave the default media type set to audio.
7. **payload-type**—Set the payload type to **101**, which is the dynamic payload type needed to support this feature.
8. **transport**—Leave the default transport protocol set to RTP/AVP.
9. **frames-per-packet**—You can leave this parameter set to **0** (default).
10. **req-bandwidth**—You can leave this parameter set to **0** (default).

### Applying the Media Profile

After you have configured the telephone-event media profile, you need to apply it either to a H.323 session agent or the global IWF configuration.

### DTMF for all IWF translations

To use DTMF for all IWF translations:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **iwf-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # iwf-config
```

From this point, you can configure IWF parameters. To view see all IWF configuration parameters, enter a ? at the system prompt.

4. Add the telephone-event media profile to the media profiles list and save your work. If you already have a media profiles for the IWF configuration set up and want to keep them (adding telephone-event to the list), then you must type in all of the media profiles that you want to use.

```
ACMEPACKET (iwf-config) # media-profiles "PCMU telephone-event"  
ACMEPACKET (iwf-config) # done
```

### DMTF Support on a Per-Session-Agent Basis

To use DMTF support on a per-session-agent basis:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # session-agent
```

From this point, you can configure IWF parameters. To view see all IWF configuration parameters, enter a ? at the system prompt.

4. When you configure a new H.323 session agent, you can configure DTMF support by simply adding the telephone-event media profile to the list of media profiles. You can add it along with the other media profiles you might want to use for that session agent.

```
ACMEPACKET (session-agent) # media-profiles "telephone-event g711Ulaw64k"
```

5. When you want to add DTMF support to an H.323 session agent that you have already configured, you need to select that session agent, add the media profile, and save your work.

```
ACMEPACKET (session-agent) # select
<hostname>:
1: 192.168.1.48 realm=          ip=
2: 192.168.1.49 realm=          ip=
3: 192.168.1.50 realm=external ip=
selection:3
ACMEPACKET (session-agent) # media-profiles "telephone-event g711Ulaw64k"
ACMEPACKET (session-agent) # done
```

## RFC 2833 DTMF Interworking

This section explains the Oracle Communications Session Border Controller's support of transporting Dual Tone Multi-Frequency (DTMF) in Real-Time Transport Protocol (RTP) packets (as described in RFC 2833) to H.245 User Input Indication (UII) or SIP INFO method interworking.

Multimedia devices and applications must exchange user-input DTMF information end-to-end over IP networks. The Oracle Communications Session Border Controller provides the interworking capabilities required to interconnect networks that use different signaling protocols. Also, the Oracle Communications Session Border Controller provides DTMF translation to communicate DTMF across network boundaries.

The Oracle Communications Session Border Controller supports:

- RFC 2833 to H.245 UII translation for H.323-to-H.323 calls, when one side is a version 4 H.323 device requiring RFC-2833 DTMF event packets, and the other side is a pre-version 4 H.323 device that only uses H.245 UII.
- RFC 2833 to H.245 UII or INFO translation of H.323 to SIP (and SIP to H.323) IWF calls, when one side is a version 4 H.323 device requiring RFC 2833 DTMF event packets and the SIP endpoint only supports INFO messages. Or when one side is a pre-version 4 H.323 device that only uses H.245 UII and the SIP endpoint supports RFC-2833 DTMF event packets.

### About RFC 2833

RFC 2833 specifies a way of encoding DTMF signaling in RTP streams. It does not encode the audio of the tone itself, instead a signal indicates the tone is being sent. RFC 2833 defines how to carry DTMF events in RTP packets. It defines a payload format for carrying DTMF digits used when a gateway detects DTMF on the incoming messages and sends the RTP payload instead of regular audio packets.

### About H.245 UII

H.245 provides a capability exchange functionality to allow the negotiation of capabilities and to identify a set of features common to both endpoints. The media and data flows are organized in logical channels. H.245 provides logical channel signaling to allow logical channel open/close and parameter exchange operations. The H.245 signaling protocol is reliable, which ensures that the DTMF tones will be delivered.

H.245 User Input Indication (UII) plays a key role in all the services that require user interaction. For video messaging, typical uses of UII include selection of user preferences, message recording and retrieval, and typical mailbox management functions. H.245 UII provides two levels of UII, alphanumeric and signal.

### About RFC 2833 to H.245 UII Interworking

The Oracle Communications Session Border Controller provides 2833 to H.245-UII interworking by checking 2833-enabled RTP streams for packets matching the payload type number for 2833. It then sends the captured packet to the host for processing and translation to H.245 UII messages. A H.245 UII message received by the Oracle Communications Session Border Controller is translated to 2833 packets and inserted into the appropriate RTP stream.

### About DTMF Transfer

DTMF transfer is the communication of DTMF across network boundaries. It is widely used in applications such as interactive voice response (IVR) and calling card applications.

The multiple ways to convey DTMF information for packet-based communications include:

- In-band audio: DTMF digit waveforms are encoded the same as voice packets. This method is unreliable for compressed codecs such as G.729 and G.723
- Out-of-band signaling events:

H.245 defines out-of-band signaling events (UII) for transmitting DTMF information. The H.245 signal or H.245 alphanumeric methods separate DTMF digits from the voice stream and send them through the H.245 signaling channel instead of through the RTP channel. The tones are transported in H.245 UII messages.

All H.323 version 2 compliant systems are required to support the H.245 alphanumeric method, while support of the H.245 signal method is optional.

SIP INFO – uses the SIP INFO method to generate DTMF tones on the telephony call leg. The SIP INFO message is sent along the signaling path of the call. Upon receipt of a SIP INFO message with DTMF content, the gateway generates the specified DTMF tone on the telephony end of the call.

- RTP named telephony events (NTE): uses NTE to relay DTMF tones, which provides a standardized means of transporting DTMF tones in RTP packets according to section 3 of RFC 2833.

Of the three RTP payload formats available, the Oracle Communications Session Border Controller supports RTP NTE. NTE is most widely used for SIP devices but is also supported in H.323 version 4 or higher endpoints.

RFC 2833 defines the format of NTE RTP packets used to transport DTMF digits, hookflash, and other telephony events between two peer endpoints. With the NTE method, the endpoints perform per-call negotiation of the DTMF transfer method. They also negotiate to determine the payload type value for the NTE RTP packets.

The NTE payload takes the place of codec data in a standard RTP packet. The payload type number field of the RTP packet header identifies the contents as 2833 NTE. The payload type number is negotiated per call. The local device sends the payload type number to use for 2833 telephone event packets using a SDP or H.245 Terminal Capability Set (TCS), which tells the other side what payload type number to use when sending the named event packets to the local device. Most devices use payload type number 101 for 2833 packets, although no default is specified in the standard.

The 2833 packet's RTP header also makes use of the timestamp field. Because events often last longer than the 2833 packets sending interval, the timestamp of the first 2833 packet an event represents the beginning reference time for subsequent 2833 packets for that same event. For events that span multiple RTP packets, the RTP timestamp identifies the beginning of the event. As a result, several RTP packets might carry the same timestamp.

See RFC 2833 and draft-ietf-avt-rfc2833bis-07.txt for more information.

### Preferred and Transparent 2833

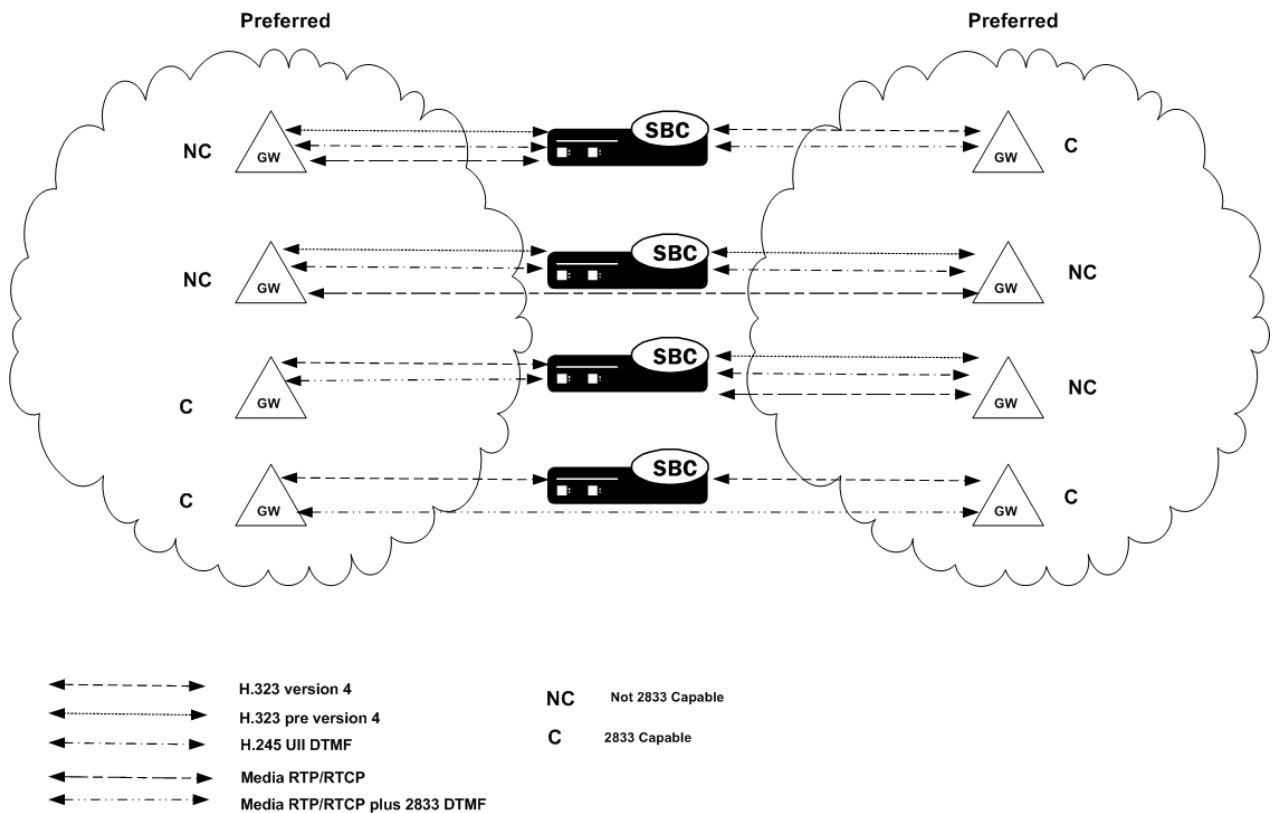
To support preferred (signaled) 2833 and transparent 2833, the Oracle Communications Session Border Controller provides 2833 detection and generation (if necessary) when the endpoint signals support for 2833.

- Preferred: the Oracle Communications Session Border Controller only generates and detects 2833 for endpoints if they negotiate support for 2833 through signaling
- Transparent: the Oracle Communications Session Border Controller offers and answers based on end-to-end signaling and transparently relaying 2833

### Preferred 2833 Support

If one side of the call, or a SIP interface, or a session agent, is configured for preferred 2833, the Oracle Communications Session Border Controller only generates and detects 2833 for endpoints if they signal support for 2833. The Oracle Communications Session Border Controller will offer 2833 in the TCS SDP, even if the originating caller did not.

- When the Oracle Communications Session Border Controller manages calls originating from a preferred source going to a preferred target, it:
  - Performs 2833 translation for an endpoint when the originating side requests 2833 but the target does not negotiate 2833
  - Allows 2833 to pass through if the originating side and target of the call are configured as preferred and negotiate 2833
- When the Oracle Communications Session Border Controller manages calls originating from a preferred source going to a transparent target, it:
  - Performs 2833 translation when the originating side requests 2833 but the target is configured as transparent and does not negotiate 2833.
  - Allows 2833 to pass through if the originating side and the target of the call are configured as transparent and negotiate 2833. The Oracle Communications Session Border Controller does not perform active translation because both ends support 2833.



If one SIP endpoint does not signal 2833 capability, but the other SIP or H.323 endpoints do, the Oracle Communications Session Border Controller does not perform 2833 translation.

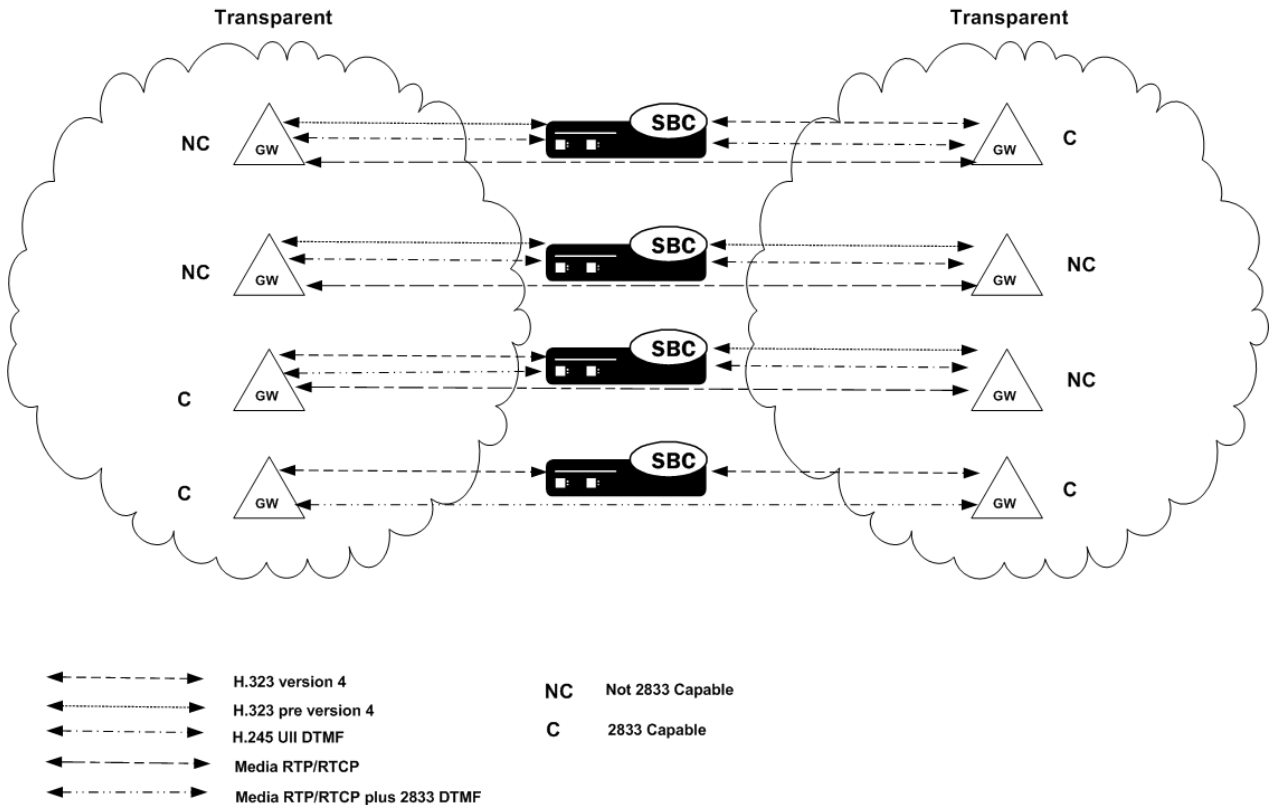
### Transparent 2833 Support

The default configuration of the Oracle Communications Session Border Controller for H.323 is transparent 2833. The Oracle Communications Session Border Controller passes on the offered capabilities to the next-hop signaling element. If the next-hop endpoint is for a transparent 2833 target, typical capability negotiation determines the DTMF

method. The Oracle Communications Session Border Controller transparently relays the DTMF as it has in previous releases.

With transparent 2833, the Oracle Communications Session Border Controller acts as a typical B2BUA or B2BGW/GK. However when the target of the call is configured as preferred 2833, the Oracle Communications Session Border Controller:

- Relays the 2833 packets if the originating endpoint signals 2833 and the next-hop endpoint for the preferred target signals 2833
- Performs 2833 translation if the originating endpoint does not signal 2833 and the next-hop endpoint for the preferred target does signal 2833
- Does not perform 2833 translation or transparently relay 2833 if the originating endpoint signals 2833 and the next-hop endpoint for the preferred target (or even a transparent 2833 target) does not signal 2833.



## Payload Type Handling

The Oracle Communications Session Border Controller supports the RTP NTE for telephony events such as transport of DTMF digits and hook flash. Using RTP NTE, endpoints perform per-call negotiation of the DTMF transfer method and negotiate payload type value for the RTP NTE packets.

Although most endpoints use payload type number 101, the RTP payload type formats can become asymmetrical when being interworked between SIP and H.323 because there is no default standard and endpoints use different types. This means that the payload type negotiated on one side of the Oracle Communications Session Border Controller, and that ends up being used for the call, might not be the same payload type negotiated on the other side of the Oracle Communications Session Border Controller. And while certain endpoints handle the asymmetry well, others do not.

Consider the simplified example of an IWF call initiated in SIP and translated to H.323. In this scenario, the SIP endpoint negotiates the payload type 106 with the Oracle Communications Session Border Controller. And despite the fact that the H.323 endpoint negotiates payload type 101, the Oracle Communications Session Border Controller returns type 106 and the call proceeds using type 106.



However, you can enable forced symmetric payload type handling so the Oracle Communications Session Border Controller changes the payload type of RFC 2833 packets to avoid using asymmetrical payload types.

For H.323 session agents and H.323 interfaces (stacks), you can configure an option that forces symmetric payload type use. The Oracle Communications Session Border Controller can detect when the payload types negotiated by the SIP and H.323 endpoints are symmetrical and when they are not. When it detects asymmetrical payload type use, the Oracle Communications Session Border Controller forces the remote endpoint to use the RFC 2833 payload type you configure in the SIP interface.

## Basic RFC 2833 Negotiation Support

If H.323, SIP, or session agents on either side of the call are configured for preferred 2833 support, the Oracle Communications Session Border Controller supports end-to-end signaled negotiation of DTMF on a call-by-call basis. If the calling party is not configured for preferred support but sends 2833, the Oracle Communications Session Border Controller sends 2833 to the next-hop called party. If the calling party sends H.245 signals or alphanumeric UII, the Oracle Communications Session Border Controller sends H.245 signals or alphanumeric UII to the next-hop called party (if it is an H.323 next-hop).

The Oracle Communications Session Border Controller also supports hop-by-hop negotiation of DTMF capability on a call-by-call basis, if the signaling protocols or session agents on either side of the call are configured for preferred 2833 support.

### H.323 to H.323 Negotiation

The Oracle Communications Session Border Controller serves as the H.323 called gateway. It answers RFC 2833 audio telephony event capability in the version 4 H.323/H.245 TCS when it receives a call from an H.323 endpoint configured for preferred RFC 2833.

If the Oracle Communications Session Border Controller is the answering device, configured for preferred support, and the calling device sends 2833, the Oracle Communications Session Border Controller accepts the 2833 regardless of the next-hop's DTMF capabilities. The received dynamic RTP payload type is used for detecting 2833 packets, while the response dynamic payload type is used for generating 2833 packets.

The Oracle Communications Session Border Controller supports:

- RFC-2833 audio telephony events in the version 4 H.323/H.245 TCS as the H.323 calling gateway, when the Oracle Communications Session Border Controller calls an H.323 endpoint configured for preferred RFC 2833 support. The Oracle Communications Session Border Controller sends 2833 to the called party regardless of whether the calling party sends it.
- H.245 UII and RFC-2833 packets sent at the same time, to the same endpoint, even if only half of the call is being provided 2833 support by the Oracle Communications Session Border Controller.

If one half of the call supports H.245 UII, and the other half is being provided 2833 translation by the Oracle Communications Session Border Controller, the Oracle Communications Session Border Controller can also forward the H.245 UII it receives to the 2833 endpoint. For example, when the signaling goes through a gatekeeper or third party call control, sending the H.245 UII in the signaling path allows those devices to learn the DTMF digits pressed.

## Signal and Alpha Type Support

The Oracle Communications Session Border Controller supports:

- H.245 signal and alpha type UII in the H.323/H.245 TCS as the H.323 calling gateway when the:

Oracle Communications Session Border Controller calls an H.323 endpoint configured for transparent 2833 support

calling endpoint's target is configured as preferred

If the originating preferred side also sends 2833, the Oracle Communications Session Border Controller forwards it to the transparent side. The Oracle Communications Session Border Controller sends signal and alpha UII support to the called party regardless of whether the calling party sends it, if the call originates from a preferred side to a transparent side.

## IWF Services

- H.245 alphanumeric UII for DTMF for H.323 endpoints that do not signal 2833 or contain explicit H.245 UII capability, for stacks configured for transparent 2833 support.

When the other half of the call is an H.323 endpoint of a stack configured for preferred 2833, the Oracle Communications Session Border Controller translates incoming H.245 UII on the transparent side, to 2833 packets on the preferred side, and vice versa. If the other half of the call is an H.323 endpoint of a transparent stack, the Oracle Communications Session Border Controller relays the H.245 UII messages.

- H.245 signal type UII for DTMF for H.323 endpoints that do not signal 2833, but do signal explicit H.245 UII capability, for stacks configured for transparent 2833 support.

When the other half of the call is an H.323 endpoint of a stack configured for preferred 2833, the Oracle Communications Session Border Controller translates incoming H.245 signaled UII on the transparent side, to 2833 packets on the preferred side, and vice versa. If the other half of the call is an H.323 endpoint of a transparent stack, the Oracle Communications Session Border Controller relays the H.245 UII messages if both sides support it.

## H.323 to SIP Calls

This section explains DTMF interworking specific to H.323 to SIP calls.

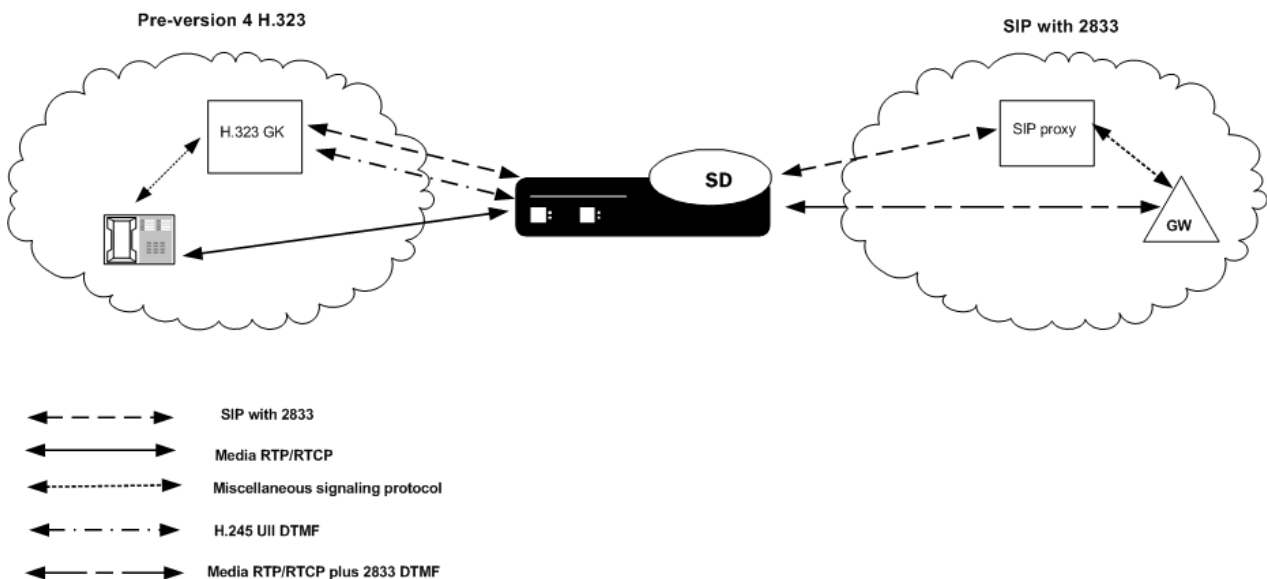
### SIP Endpoints

SIP endpoints include those that support:

- RFC 2833
- SIP INFO method

## H.323 Non-2833 Interworking with SIP

RFC 2833 and the SIP INFO method can be used for conveying DTMF information for SIP based-services. (RFC 2833 is the most widely used.) To provide end-to-end DTMF for SIP devices supporting RFC-2833 interworking with H.323 devices that do not, an RFC 2833 to H.323 UII interworking function is provided.



## How H.323 to SIP Calls Work

For H.323 to SIP IWF calls, if 2833-related information is to be sent in the INVITE, the SIP interface of the SIP session agent has to be configured with the `rfc2833-mode` parameter set to preferred.

The following example shows an INVITE without 2833 in the SDP:

```

Apr  5 04:28:50.073 On 127.0.0.1:5070 sent to 127.0.0.1:5060
INVITE sip:780@192.168.200.6:5060 SIP/2.0
Via: SIP/2.0/UDP
127.0.0.1:5070;branch=z9hG4bKIWF0000g12018604agg71c0;acme_irealm=external;acme
_sa=192.168.1.6
Contact: "jdoe"<sip:127.0.0.1:5070>
GenericID: 1144211330000000@000825010100
Supported: 100rel^M
From: "msmith"<sip:192.168.200.68:5060>;tag=000000ab00011940
To: <sip:780@192.168.200.6:5060>
Call-ID: 7f00000113ce000000ab000101d0@127.0.0.1
CSeq: 2 INVITE
Content-Length: 225
Content-Type: application/sdp
v=0
o=IWF 3 3 IN IP4 192.168.1.6
s=H323 Call
c=IN IP4 192.168.1.6
t=0 0
m=audio 5214 RTP/AVP 0 18
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
a=fmtp:18 annexb=no
m=video 5216 RTP/AVP 31
a=rtpmap:31 H261/9000/1

```

## SIP INFO—RFC 2833 Conversion

The Oracle Communications Session Border Controller can perform SIP INFO—RFC 2833 conversion. The Oracle Communications Session Border Controller also provides a way for you to enable a dual conversion mode, where the Oracle Communications Session Border Controller:

- Inserts telephone-event in the SDP offer
- Generates both RFC 2833 event packets and SIP INFO messages regardless of whether or not the SDP offer indicates RFC 2833

You can enable this feature either for SIP interfaces or session agents. The following apply:

- If the next hop SIP interface or session agent's **rfc2833-mode** is set to preferred, then the SD inserts RFC 2833 into the SDP offer/answer. This occurs regardless of whether:
  - The original SDP on the opposite side of the call does not support RFC 2833
  - The opposite side's SIP interface or session agent is set to transparent mode
- If the next hop SIP interface or session agent is set to transparent, then the behavior of the Oracle Communications Session Border Controller depends on the previous hop.
  - If the previous hop is a SIP interface or session agent configured for transparent mode, then the S Oracle Communications Session Border Controller does not perform any conversion.
  - If the previous hop is a SIP interface or session agent configured for preferred mode, the Oracle Communications Session Border Controller does not insert RFC-2833 into the SDP on the transparent side. It does, however, translate from RFC 2833 to SIP INFO if the originating endpoint supports RFC 2833.

## IPv6 SIP INFO to RFC 2833 Telephone Event Interworking

The Oracle Communications Session Border Controller can interwork SIP INFO and RFC Telephone Event messages for IPv4, IPv6—or for any session requiring interworking between IPv4 and IPv6. Other than installing the applicable licences on your Oracle Communications Session Border Controller and enabling IPv6 support in your system configuration (**system-config**), you do not have to perform any configuration steps for this capability to work.

## RFC 2833 Interworking Configuration

This section explains how to configure RFC 2833 to H.245 User Input Indication (UII) or SIP INFO method interworking.

### RFC 2833 Mode for H.323 Stacks

To configure RFC 2833 mode for H.323 stacks:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET (session-router) # h323
```

4. Type **h323-stacks** and press Enter.

```
ACMEPACKET (h323) # h323-stacks  
ACMEPACKET (h323-stack) #
```

From this point, you can configure H.323 stack parameters. To view all H.323 stack parameters, enter a ? at the system prompt.

5. **rfc2833-mode**—Set the RFC2833 mode. The default value is **transparent**. Valid values are:
  - **transparent**—The Oracle Communications Session Border Controller and H.323 stack behave exactly the same way as before and the 2833 or UII negotiation is transparent to the Oracle Communications Session Border Controller.
  - **preferred**—The H.323 stack uses 2833 for DTMF transfer, which it signals in its TCS. However, the remote H323 endpoint makes the decision. If the endpoint supports 2833, 2833 is used. If not, the H.323 stack reverts back to using UII. You configure the payload format by configuring the h323-config element.

### RFC 2833 Payload for H.323

To configure the RFC 2833 payload in preferred mode:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **h323** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # h323
```

From this point, you can configure global H.323 parameters. To view all H.323 configuration parameters, enter a ? at the system prompt.

4. **rfc2833-payload**—Enter a number that indicates the payload type the Oracle Communications Session Border Controller will use for RFC 2833 packets while interworking 2833 and UII. The default value is **101**.
  - Minimum—96
  - Maximum—127

### Configuring the SIP Interface

You configure the 2833 mode and payload for the SIP interface. You must configure the payload the Oracle Communications Session Border Controller will use for RFC 2833 packets, while interworking 2833 and INFO/UII.

To configure the SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-interface) #
```

From this point, you can configure SIP interface parameters. To view all sip-interface parameters, enter a ? at the system prompt.

4. **rfc2833-payload**—Enter a number that indicates the payload type the SIP interface will use for RFC 2833 packets while interworking 2833 and UII. The default value is **101**. The valid range is:

- Minimum—96
- Maximum—127

5. **rfc2833-mode**—Set the RFC 2833 mode for the SIP interface. The default value is **transparent**. Valid values are:

- **transparent**—The SIP INFO and RFC 2833 translation is transparent to the Oracle Communications Session Border Controller.
- **preferred**—The RFC 2833 transfer method is the preferred method for sending DTMF, and a telephone event is inserted in the SDP of the outgoing offer. The actual method of transfer, however, depends on the SDP offer/answer exchange that occurs between the Oracle Communications Session Border Controller and remote endpoint. If the remote endpoint supports RFC 2833, the Oracle Communications Session Border Controller performs SIP INFO—RFC 2833 conversion.
- **dual**—The Oracle Communications Session Border Controller behaves the same as it does when set to preferred mode, and it forwards both the original DTMF mechanism and the translated one to the remote endpoint.

## Configuring Session Agents

You configure session agents with:

- payload type the Oracle Communications Session Border Controller wants to use for RFC 2833 packets while interworking 2833 and UII.

The default value for this attribute is 0. When this value is zero, the global rfc2833-payload configured in the h323-configuration element will be used instead. For SIP session agents, the payload defined in the SIP interface is used, if the SIP interface is configured with the preferred RFC 2833 mode.

- 2833 mode

A value of transparent or preferred for the session agent's 2833 mode will override any configuration in the h323-stack configuration element.

To configure session agents:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # session-agent
ACMEPACKET (session-agent) #
```

4. **rfc2833-mode**—Set the RFC 2833 mode you want the session agent to use. The default value is **none**. Valid values are:
  - **none**—RFC 2833 to UII interworking is based on the H.323 stack configuration.
  - **transparent**—The RFC 2833 or UII negotiation is transparent to the Oracle Communications Session Border Controller. This overrides the H.323 stack configuration, even if the stack is configured for preferred mode.
  - **preferred**—RFC 2833 for DTMF transfer is preferred, which is signaled in the TCS. If the endpoint supports 2833, 2833 is used. If not, the H.323 stack configured as preferred will revert back to using UII. This overrides any configuration in the h323-stack even if the stack is configured for transparent mode.

For SIP INFO—RFC 2833 conversion, you can choose:

  - **none**—The 2833-SIP INFO interworking will be decided based on the sip-interface configuration.
  - **transparent**—The session agent behaves the same as it did without the SIP INFO—RFC 2833 conversion feature. The SIP INFO and RFC 2833 translation is transparent to the Oracle Communications Session Border Controller.
  - **preferred**—The RFC 2833 transfer method is the preferred method for sending DTMF, and a telephone event is inserted in the SDP of the outgoing offer. The actual method of transfer, however, depends on the SDP offer/answer exchange that occurs between the Oracle Communications Session Border Controller and remote endpoint. If the remote endpoint supports RFC 2833, the Oracle Communications Session Border Controller performs SIP INFO—RFC 2833 conversion.
  - **dual**—The Oracle Communications Session Border Controller behaves the same as it does when set to preferred mode, and it forwards both the original DTMF mechanism and the translated one to the remote endpoint.
5. **rfc2833-payload**—Enter a number that indicates the payload type the session agent will use for RFC 2833 packets while interworking 2833 and UII. The default value is 0. The valid range is:
  - Minimum—0, 96
  - Maximum—127

### Enabling Payload Type Handling

You can configure H.323 session agents and H.323 interfaces (stacks) with an option that forces symmetric payload type use. For Payload Type Handling to work properly, you must set the following SIP interface and the global H.323 configuration parameters with these values:

- **rfc2833-mode**—Set this parameter to **preferred**; the default is **transparent**.
- **rfc2833-payload**—Set this parameter to the payload type you want forced for the remote endpoint. Your entry will be between **96** and **127**, with **101** as the default.

To enable forced symmetric payload type handling for an H.323 session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET(session-router) # session-agent
ACMEPACKET(session-agent) #
```

If you want to add this option to a pre-existing H.323 session agent, select the one you want to edit.

4. **options**—Set the options parameter by typing options, a Space, the option name **Map2833ForceRemotePT** with a plus sign in front of it. Then press Enter.

```
ACMEPACKET(session-agent) # options +Map2833ForceRemotePT
```

If you type options and then the option value for either of these entries without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

To enable forced symmetric payload type handling for an H.323 interface:

6. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

7. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

8. Type **h323-config** and press Enter.

```
ACMEPACKET(session-router)# h323-config
ACMEPACKET(h323-config)#
```

9. Type **h323-stacks** and press Enter.

```
ACMEPACKET(h323-config)# h323-stacks
ACMEPACKET(h323-stack)#
```

10. **options**—Set the options parameter by typing options, a Space, the option name **Map2833ForceRemotePT** with a plus sign in front of it. Then press Enter.

```
ACMEPACKET(h323-stack)# options +Map2833ForceRemotePT
```

If you type **options** and then the option value for either of these entries without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

11. Save and activate your configuration.

## DTMF Transparency for IWF

In certain vendors' implementations of DTMF during SIP/H.323 IWF, there have been discrepancies between the RFC 2833 and UI/INFO negotiations and what type of messages actually get sent. Instead of correcting these errors on its own end, the Oracle Communications Session Border Controller has perpetuated these inaccuracies.

To ensure that the Oracle Communications Session Border Controller always sends the correctly negotiated protocols, a **media-manager-config** parameter called **translate-non-rfc2833-event** has been created. When **translate-non-rfc2833-event** is enabled, the Oracle Communications Session Border Controller always sends the type of messages that were initially negotiated, regardless of the type of messages it may be receiving.

### DTMF Transparency Configuration

To enable DTMF transparency for SIP/H.323 IWF:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **media-manager** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# media-manager
ACMEPACKET(media-manager-config)#
```

4. **translate-non-rfc2833-event**—To enable this feature, set this parameter to **enabled**. If you do not want to use the feature leave it set to its default behavior, disabled.

5. Save and activate your configuration.

### RFC 2833 Packet Sequencing

You can configure your Oracle Communications Session Border Controller to generate either the entire start-interim-end RFC 2833 packet sequence or only the last three end 2833 packets for non-signaled digit events.

#### RFC 2833 Packet Sequencing Configuration

To send only the last three end 2833 packets for non-signaled digits events:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **media-profile** and press Enter.

```
ACMEPACKET(media-manager)# media-manager
ACMEPACKET(media-manager-config)#
```

4. **rfc2833-end-pkts-only-for-non-sig**—By default, this parameter is enabled—meaning that only the last three end 2833 packets are used for non-signaled digits events. Set this parameter to disabled if you want the entire start-interim-end RFC 2833 packet sequence for non-signaled digit events

5. Save and activate your configuration.

### Enhanced H.245 to 2833 DTMF Interworking

Enhanced H.245 to 2833 and SIP INFO to 2833 DTMF interworking addresses issues experienced where the way the Oracle Communications Session Border Controller timestamps audio RTP packets result in dropped digits and digits with a stutter pattern. These occurrences can cause other network devices to deem the packets unrecoverable (due to jitter), meaning that they will never render the digit.

The Oracle Communications Session Border Controller offers the following:

- Timestamp is based on the current time—The Oracle Communications Session Border Controller can compute the timestamp of the egress 2833 packets using the actual time elapsed in milliseconds since the last RTP packet (rather than incrementing the time by 1 sample). Not only does the Oracle Communications Session Border Controller fill out the timestamp field more accurately, but it also recalculates the checksum.
- End-event 2833 messages default behavior—The Oracle Communications Session Border Controller's new default behavior is to send three end-event 2833 packets only if the DTMF event is received for:
  - An alphanumeric UII or SIP INFO with no duration
  - A signaled UII or SIP INFO with a duration less than the minimum signal duration (the value you configure using the new media manager configuration **min-signal-duration** option)

For a signaled UII or SIP INFO with a duration greater than the minimum signal duration, the Oracle Communications Session Border Controller behaves as it does in prior releases: It sends the initial event packets, any interim packets (if they exist), and the three end packets.

- Configurable duration for the 2833 event—Without the enhancements being configured, the Oracle Communications Session Border Controller uses a 250 millisecond duration for the 2833 event when it receives an alphanumeric UII or a SIP INFO with no specified duration. The result is that 2833 packets are sent at 50-millisecond intervals until the 250 millisecond time expires; then the three end-event packets are sent.

Now the Oracle Communications Session Border Controller allows you to set the duration of these 2833 events using a new **default-2833-duration** parameter (with a 100 millisecond default) in the media manager configuration. In addition, the Oracle Communications Session Border Controller uses this configured value (instead of the duration sent in the signaling message) when it receives an UII or SIP INFO with a duration less



than the minimum signal duration. It checks to make sure that the value for the **default-2833-duration** parameter is greater than the minimum signal duration.

- Configurable minimum signal duration value—Without this configured, the Oracle Communications Session Border Controller accepts and uses the duration it receives in the UII or SIP INFO for the 2833 event. However, you can configure this value using the **min-signal-duration** option in the media manager configuration. If the duration the Oracle Communications Session Border Controller receives is less than the threshold, it uses the value configured in the **default-2833-duration** parameter.

If you do not configure this option, then there is no signaling duration threshold.



**Note:** Timestamp changes and duration changes only take effect when the 2833 timestamp (**rfc-2833-timestamp**) is enabled in the media manager configuration.

## Enhancements Configuration

This section shows you how to configure enhancements for H.245 UII/SIP INFO—2833 DTMF interworking.

To enable the Oracle Communications Session Border Controller to calculate the timestamp based on the current time:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
```

3. Type **media-profile** and press Enter.

```
ACMEPACKET(media-manager)# media-manager
ACMEPACKET(media-manager-config)#
```

4. **rfc-2833-timestamp**—Enable this parameter to use a timestamp value calculated using the actual time elapsed since the last RTP packet. The default is **disabled**. Valid values are:

- enabled | disabled

5. Save and activate your configuration.

To configure a duration for the 2833 event:

6. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

7. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
```

8. Type **media-profile** and press Enter.

```
ACMEPACKET(media-manager)# media-manager
ACMEPACKET(media-manager-config)#
```

9. **default-2833-duration**—Set this parameter to the time value in milliseconds for the Oracle Communications Session Border Controller to use when it receives an alphanumeric UII or a SIP INFO with no specified duration; then the three end-event packets are sent. The default value is 100. The valid range is:

- Minimum—50
- Maximum—5000

10. Save and activate your configuration.

## Setting the Minimum Signal Duration

To configure the minimum signal duration value:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
```

3. Type **media-profile** and press Enter.

```
ACMEPACKET (media-manager) # media-manager
ACMEPACKET (media-manager-config) #
```

4. **options**—Set the options parameter by typing **options**, a Space, the option name **min-signal-duration=x** (where **x** is the value in milliseconds you want to use for the threshold) with a plus sign in front of it. Then press Enter.

```
ACMEPACKET (media-manager-config) # options +min-signal-duration=200
```

If you type **options** and then the option value for either of these entries **without the plus sign**, you will overwrite any previously configured options. In order to append the new option to the configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

## SIP Tel URI Support

---

The Oracle Communications Session Border Controller maps H.323 addresses to either SIP URIs or Tel URIs. You can configure the Oracle Communications Session Border Controller to include Tel URIs in the following SIP headers for calls that require the IWF:

- Request Line
- To
- From

When Tel URI support is not used on a Oracle Communications Session Border Controller performing IWF translations, the SIP INVITE is formatted like it is in the following example. This example uses 192.168.5.5 as the external proxy address, or the next hop (as configured in the local policy).

```
INVITE sip:602@192.168.5.5:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.5.58:5060;branch=z9hG4bKIWF0aqoqg001g11a7kos4g0
Contact: <sip:603@192.168.5.58:5060>
From: <sip:603@192.168.5.58:5060>;tag=4069ac210018a0
To: <sip:602@192.168.5.5:5060>
```

In the example above, the session needs to be routed to another SIP proxy that can resolve an E.164 number to a SIP address. However, the next SIP proxy must be informed that the message will be routed based on the included E.164 number; the SIP address of the Request URI does not have a routable SIP address. To devise a routable address, the Request URI must be reconstructed as a Tel URI.

Without Tel URI support configured, the terminating SIP user would be required to have an address of 602@192.168.5.5, where the IPv4 address portion is the same as the address for the proxy. If it were not the same, then the session would terminate at the proxy. However, the proxy would be unable to handle the session because the SIP address it received would be unknown/unroutable.

Because it is not desirable to have an IPv4 address be the user-identity and rely on the configuration of the IP network, the SIP INVITE generated by the Oracle Communications Session Border Controller and sent to the proxy must have the following format if it is sent to an H.323 entity.

```
INVITE tel:2345 SIP/2.0
Via: SIP/2.0/UDP 192.168.5.52:5060;branch=z9hG4bKIWFaqoqg00cobgf9so10o0
Contact: <sip:1234@192.168.5.58:5060>
From: <tel:1234>;tag=4069ac35000c5ff8
To: <tel:2345>
Call-ID:7f0000113ce4069ac35000c5440
```

```
CSeq: 1 INVITE
Content-Length: 155
Content-Type: application/sdp
```

## SIP Interface Configuration

You enable this feature in the SIP interface configuration.

To configure SIP Tel URI support for calls that require the IWF:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET(session-router)# sip-interface
```

From this point, you can configure SIP interface parameters. To view see all SIP interface parameters, enter a ? at the system prompt.

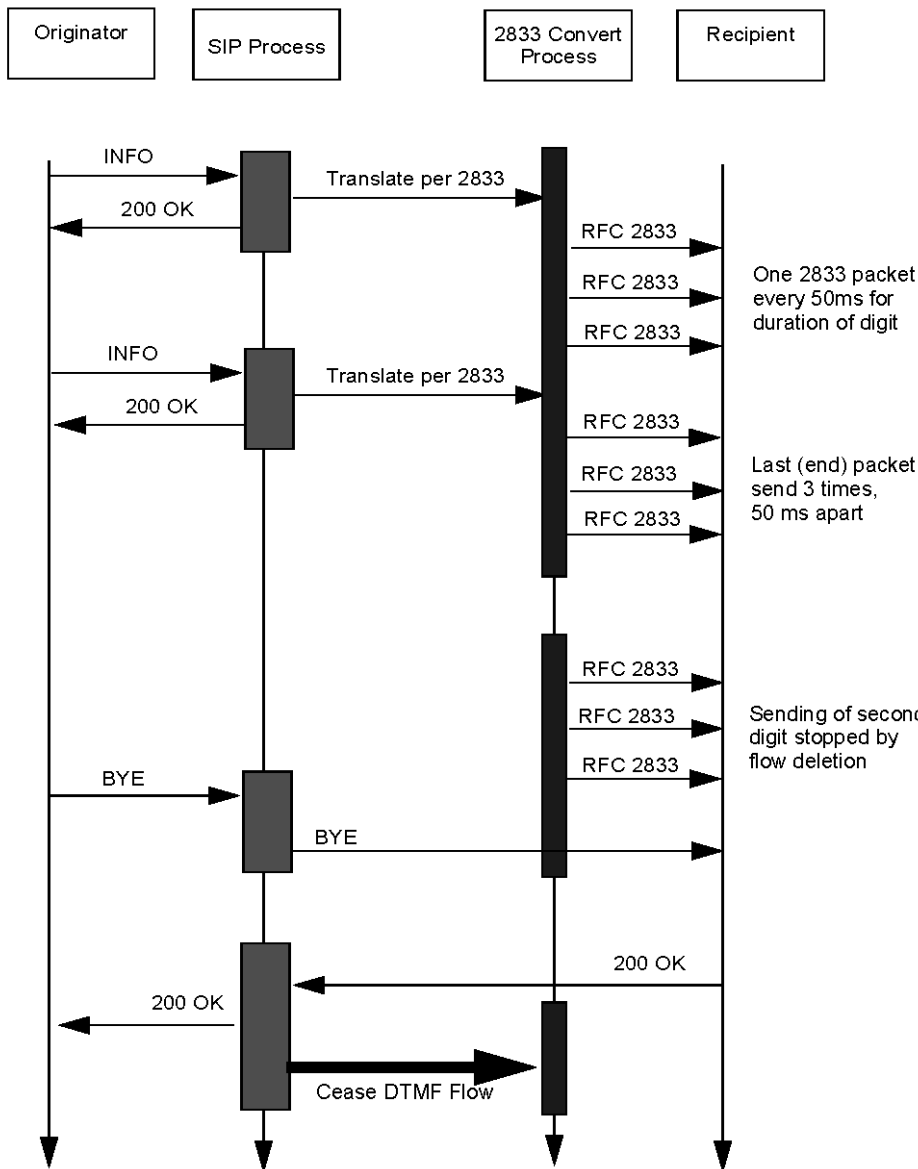
4. **teluri-scheme**—Enable or disable the conversion of SIP URIs to Tel URIs. The default value is **disabled**. Valid values are:

- enabled | disabled

```
ACMEPACKET(sip-interface)# teluri-scheme enabled
ACMEPACKET(sip-interface)# done
```

## Graceful DTMF Conversion Call Processing

The default implementation for SIP INFO to RFC2833 events performs well in most network topologies is shown below.



When the Oracle Oracle Communications Session Border Controller receives an INFO DTMF request, the SIP process determines whether or not it needs to perform DTMF-to-RFC2833 translation. If translation is required, the process forwards the DTMF to the 2833 convert process for translation and transmission to the recipient. Immediately after off-loading the DTMF, the SIP process sends a 200 OK response for the INFO. As shown in the figure, the 2833 convert process generates a number of RFC2833 packets to represent received DTMF digits.

Specifically, the 2833 convert process generates one RFC 2833 packet every 50 milliseconds for the duration of the DTMF digit, whose length is specified in the INFO request, and two retransmits of the last packet (known as the end packet) 50 milliseconds apart.

Consequently, the time interval between the 200 OK and the actual transmission of the RFC 2833 translation is the sum of the DTMF duration and 100 ms.

**Note:** This time interval can be shortened to 100 ms by enabling the `rfc2833-end-pkts-only-for-non-sig` parameter in `media-manger` which results in Oracle Communications Session Border Controller only generating the last packet and its two retransmits.

The early 200 OK allows the endpoint to send the next DTMF digit before the SD has sent all the RFC2833 packets, resulting in the next digit being queued internally by the 2833 convert process before being sent.

A problem arises if the SIP process receives a BYE request from the DTMF originator while queued digits are awaiting translation and transmission. In such an event, the SIP process immediately forwards the BYE request to the recipient, ending the session with DTMF digits awaiting translation and transmission.

An alternative DTMF conversion model provides for a feedback mechanism from the 2833 convert process to the SIP process. With this model enabled, the SIP process buffers a received BYE until it obtains confirmation that all queued DTMF digits have been translated and transmitted. Only after obtaining confirmation, does it forward the BYE to terminate the session.

This processing model is enable by a SIP option, **sync-bye-and-2833**, and requires that **rfc2833-mode** parameter on the egress interfaces is NOT set to dual, any value other than dual, is supported.

1. From superuser mode, use the following command sequence to access sip-config configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

2. Use the SIP option **sync-bye-and-2833** to delay BYE processing until DTMF-to-RFC2833 translation has been completed.

```
ACMEPACKET(sip-config)# options +sync-bye-and-2833="enabled"
ACMEPACKET(sip-config)#
```

3. Use the **done** and **exit** commands to complete configuration.

```
ACMEPACKET(sip-config)# done
ACMEPACKET(sip-config)# exit
ACMEPACKET(session-router)#
```

## IWF Inband Tone Option

This option enables the Oracle Communications Session Border Controller to send a progress indicator (PI)=8 in an H.225 message when an SDP is received in a provisional message. In effect, this option sends network announcements inband. It is also applicable because in some networks H.323 endpoints support early H.245.

The H.323 inband tone option is enabled by adding the **inbandTone** as an option in a configured H.323 stack.

When this option is not used, the ringtone is generated locally (NO PI=8 in PROGRESS OR ALERTING) is the default behavior.

## IWF Inband Tone Configuration

To configure the IWF inband tone option:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the media-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
```

4. Type **h323-stacks** and press Enter.

```
ACMEPACKET(h323)# h323-stacks
ACMEPACKET(h323-stacks)#
```

5. Use the ACLI **select** command add this feature to an existing H.323 interface.

```
ACMEPACKET(h323-stacks)# select
```

6. If you are adding this service to a new H.323 interface, type **option inbandTone** and press Enter.

ACMEPACKET(h323-stacks)# **option inbandTone**

7. If you are adding this service to an H.323 interface that already exists, type **select** to select the interface to which you want to add the service. Then use the options command and prepend the option with a plus (+) sign.
  - If you know the name of the interface, you can type the name of the interface at the name: prompt and press Enter.
  - If you do not know the name of the interface, press Enter at the name: prompt. A list of interfaces will appear. Type the number corresponding to the interface you want to modify, and press Enter.
  - **If are adding service to an existing interface and type in the option without a plus (+) sign, you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign: options +inbandTone.**


## AMR AMR-WB Payload Type Mapping

When the Oracle Communications Session Border Controller connects endpoints that choose AMR with different dynamic payload types but equivalent AMR parameters, payload type remapping can be handled by the local network processor rather than using transcoding resources. In prior releases, the system transcoded all AMR-to-AMR calls or all AMR-WB-to-AMR-WB calls when the Offer/Answer differed only in the assignment of dynamic payload types, or in the values assigned to certain AMR/AMR-WB parameters.

The decision to transcode is based entirely upon a comparison of the SDP Offer presented by the session initiator and the SDP Answer returned by the session responder. Transcoding is NOT REQUIRED when all of the following conditions are met.

- The offer/answer codecs are identical.
- The offer/answer payload formats (specified by the AMR/AMR-WB octet-align parameter) are identical.
- The offer/answer mode-sets (specified by the AMR/AMR-WB mode-set parameter) are identical or intersect.

To use this feature you must create appropriate media profiles (examples given below) and codec policies. You must also enable the audio-payload-type-mapping option in the media-manager-config.

 **Note:** On a per-call basis, if AMR/AMR-WB payload mapping is invoked, the system can not also perform RFC 2833 to SIP INFO/H.323 UII payload translation. If a call starts with RFC 2833 translation and per a call change AMR/AMR-WB payload mapping, the system prioritizes the AMR/AMR-WB action and ceases RFC 2833 translation.

## AMR AMR-WB octet-align Parameter

AMR/AMR-WB sessions can be established in either bandwidth-efficient or octet-aligned format. Octet-aligned format increases processing efficiency, and is required for AMR's robust sorting, interleaving and frame CRC capabilities. With octet-aligned format, all fields in the AMR/AMR-WB header are individually aligned to octet boundaries by the addition of padding bits. With bandwidth-efficient format, only the full payload is octet aligned resulting in the addition of fewer padding bits.

Format usage is specified by the optional AMR/AMR-WB octet-align parameter, which assumes one of two values: 0 (the default), indicating bandwidth-efficient format, or 1 indicating octet-aligned format. In the absence of the octet-align parameter, bandwidth-efficient format is employed.

A mismatch of offer/answer octet-align values requires transcoding as shown in the following table, where N/P indicates the absence of the octet-align parameter.

octet-align Values		Format	Transcoding Required
Offer	Answer		
0	0	bandwidth-efficient	No
1	1	octet-aligned	No
0	1	mismatch	Yes
1	0	mismatch	Yes

N/P	N/P	bandwidth-efficient	No
N/P	0	bandwidth-efficient	No
N/P	1	mismatch	Yes
0	N/P	bandwidth-efficient	No
1	N/P	mismatch	Yes

## AMR AMR-WB mode-set Parameter

AMR/AMR-WB codecs support 8 standard encoding modes as shown in the following table.

Frame Content	Mode Indicator	Frame Content
AMR 4.75 kbit/sec	0	AMR-WB 6.60 kbit/sec
AMR 5.15 kbit/sec	1	AMR-WB 8.85 kbit/sec
AMR 5.90 kbit/sec	2	AMR-WB 12.65 kbit/sec
AMR 6.70 kbit/sec	3	AMR-WB 14.25 kbit/sec
AMR 7.40 kbit/sec	4	AMR-WB 15.85 kbit/sec
AMR 7.95 kbit/sec	5	AMR-WB 18.25 kbit/sec
AMR 10.2 kbit/sec	6	AMR-WB 19.85 kbit/sec
AMR 12.2 kbit/sec	7	AMR-WB 23.05 kbit/sec

The optional AMR/AMR-WB mode-set parameter can be used to restrict the session mode set to a sub-set of the 8 standard modes. If a mode-set is specified, it must be honored, and frames encoded with modes not specified within the sub-set must not be sent in any RTP payload or used in codec mode requests. In the absence of a mode-set parameter, the inclusive mode-set (0,1,2,3,4,5,6,7) provides the default value.

## Other AMR AMR-WB Parameters

AMR/AMR-WB support the following additional optional parameters. These parameters are irrelevant in the transcoding decision process. However, they can change the SDP in the answer returned to the session initiator.

channels	mode-change-neighbor
crc	mode-change-period
interleaving	mode-set
max-red	octet-align
maxtime	ptime
mode-change-capability	robust-sorting

## Examples and Explanations

The following sections illustrate scenarios of various levels of complexity.

Basic Scenarios illustrate default-based offer/answer exchanges. In the absence of explicitly configured AMR/AMR-WB octet-align parameter in the original offer or answer, both the initiator and the responder opt for bandwidth efficient payload format. In a similar fashion, the absence of an AMR/AMR-WB mode-set parameter indicates mutual acceptance of the inclusive mode-set (0,1,2,3,4,5,6,7). This section also includes examples of intersecting mode-sets.

Advanced Scenarios illustrate offer/answer exchanges that contain AMR or AMR-WB mode-set parameters.

SDP examples contain rtpmap and fmpm attributes whose syntax is described in the following two sections.

### rtpmap Attribute

The rtpmap attribute takes the form:

```
a=rtpmap:<payloadType> <encodingName>/<clockRate>[/audioChannels]
```

payloadType: contains a dynamically-assigned RTP payload type as specified in RFC 3551, RTP Profile for Audio and Video Conferences with Minimal Control. The Internet Assigned Numbers Authority (IANA) has designated RTP payload types 96 through 127 as available for dynamic assignment.

encodingName: contains the codec identifier (AMR or AMR-WB in the following tables).

clockRate: contains the sampling rate.

audioChannels: (used only for audio streams, and optional) contains the number of audio channels — not used in transcoding decisions.

### fmtp Attribute

The fmtp attribute shown in the following tables takes the form:

```
a=fmtp:<amrParameter> <parameterValues>
```

amrParameter contains one of the AMR/AMR-WB RTP-specific parameters listed below.

```
channels
crc
interleaving
max-red
maxtime
mode-change-capability
mode-change-neighbor
mode-change-period
mode-set
octet-align
ptime
robust-sorting
```

parameterValue: contains the value of the specified parameter.

### Basic Scenarios

Examples provided in this section are supported by the following codec policy and media profiles.

```
codec-policy
  name      net192
  allow-codecs      *
  add-codecs-on-egress  AMR::PT96 AMR-WB::PT97 AMR::PT98
  AMR-WB::PT99
  order-codecs
  force-ptime      disabled
  packetization-time  30
  dtmf-in-audio    disabled
```

```
media-profile
  name          AMR
  subname       PT96
  media-type    audio
  payload-type  96
  transport     RTP/AVP
  req-bandwidth 0
  frames-per-packet 0
  parameters
  average-rate-limit 0
  peak-rate-limit    0
  max-burst-size     0
  sdp-rate-limit-headroom 0
  sdp-bandwidth      disabled
  police-rate        0
  standard-pkt-rate  0
```

```
media-profile
  name          AMR-WB
  subname       PT97
  media-type    audio
  payload-type  97
  transport     RTP/AVP
  req-bandwidth 0
  frames-per-packet 0
  parameters
  average-rate-limit 0
```



```

    peak-rate-limit          0
    max-burst-size          0
    sdp-rate-limit-headroom 0
    sdp-bandwidth           disabled
    police-rate             0
    standard-pkt-rate       0

media-profile
    name                    AMR
    subname                 PT98
    media-type              audio
    payload-type            98
    transport               RTP/AVP
    req-bandwidth           0
    frames-per-packet       0
    parameters              mode-set="0,1,2,3"
    average-rate-limit      0
    peak-rate-limit         0
    max-burst-size          0
    sdp-rate-limit-headroom 0
    sdp-bandwidth           disabled
    police-rate             0
    standard-pkt-rate       0

media-profile
    name                    AMR-WB
    subname                 PT99
    media-type              audio
    payload-type            99
    transport               RTP/AVP
    req-bandwidth           0
    frames-per-packet       0
    parameters              mode-set="0,1,2,3"
    average-rate-limit      0
    peak-rate-limit         0
    max-burst-size          0
    sdp-rate-limit-headroom 0
    sdp-bandwidth           disabled
    police-rate             0
    standard-pkt-rate       0

```

The following SDP reflects the case where the session responder answers with the same codec and payload type as presented by the session initiator.

Offer received from session initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000
	INVITE SDP (partial, codec information only)
Offer sent to responder (per the net192 codec policy)	m=audio xx RTP/AVP 97 98 99 96
from AMR-WB/PT97 media profile	a=rtpmap:97 AMR-WB/16000/1
from AMR-WB/PT98 media profile	a=rtpmap:98 AMR/8000
	a=fmtp:98 mode-set=0,1,2,3
from AMR-WB/PT99 media profile	a=rtpmap:99 AMR-WB/16000/1
	a=fmtp:99 mode-set=0,1,2,3
initiator offer	a=rtpmap:96 AMR/8000

## IWF Services

Answer received from responder	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000
Answer sent to initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000

codecs	match	AMR/AMR
payload formats	match	bandwidth-efficient (default value)
mode-sets	match	inclusive mode-set (0,1,2,3,4,5,6,7)
payload mapping	match	96/96

Here, given the identical offer and answer, neither transcoding nor payload type mapping is required.

The following SDP reflects the case where the session responder answers with the codec offered by the session initiator, but with a different payload type.

Offer received from session initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000
	INVITE SDP (partial, codec information only)
Offer sent to responder (per the net192 codec policy)	m=audio xx RTP/AVP 97 98 99 96
from AMR-WB/PT97 media profile	a=rtpmap:97 AMR-WB/16000/1
from AMR-WB/PT98 media profile	a=rtpmap:98 AMR/8000
	a=fmtp:98 mode-set=0,1,2,3
from AMR-WB/PT99 media profile	a=rtpmap:99 AMR-WB/16000/1
	a=fmtp:99 mode-set=0,1,2,3
initiator offer	a=rtpmap:96 AMR/8000
Answer received from responder	m=audio xx RTP/AVP 97
	a=rtpmap:97 AMR/8000
Answer sent to initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000

codecs	match	AMR/AMR
payload formats	match	bandwidth-efficient (default value)
mode-sets	match	inclusive mode-set (0,1,2,3,4,5,6,7)
payload mapping	no match	96/97

Here, given the identical offer/answer except for the payload type, transcoding is not required; payload type mapping is used.

The following SDP reflects the case where the session responder answers with a codec not offered by the session initiator.

Offer received from session initiator	m=audio xx RTP/AVP 96
---------------------------------------	-----------------------

	a=rtpmap:96 AMR/8000
	INVITE SDP (partial, codec information only)
Offer sent to responder (per the net192 codec policy)	m=audio xx RTP/AVP 97 98 99 96
from AMR-WB/PT97 media profile	a=rtpmap:97 AMR-WB/16000/1
from AMR-WB/PT98 media profile	a=rtpmap:98 AMR/8000
	a=fmtp:98 mode-set=0,1,2,3
from AMR-WB/PT99 media profile	a=rtpmap:99 AMR-WB/16000/1
	a=fmtp:99 mode-set=0,1,2,3
initiator offer	a=rtpmap:96 AMR/8000
Answer received from responder	m=audio xx RTP/AVP 97
	a=rtpmap:97 AMR/16000
Answer sent to initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000

codecs	no match	AMR/AMR-WB
payload formats	match	bandwidth-efficient (default value)
mode-sets	match	inclusive mode-set (0,1,2,3,4,5,6,7)
payload mapping	no match	96/97

Here, given that the offer and answer codecs (AMR and AMR-WB) are not identical, transcoding is required.

The following SDP reflects the case where the offer specifies the inclusive mode-set, and the answer specifies a non-inclusive mode-set.

Offer received from session initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000
	INVITE SDP (partial, codec information only)
Offer sent to responder (per the net192 codec policy)	m=audio xx RTP/AVP 97 98 99 96
from AMR-WB/PT97 media profile	a=rtpmap:97 AMR-WB/16000/1
from AMR-WB/PT98 media profile	a=rtpmap:98 AMR/8000
	a=fmtp:98 mode-set=0,1,2,3
from AMR-WB/PT99 media profile	a=rtpmap:99 AMR-WB/16000/1
	a=fmtp:98 mode-set=0,1,2,3
initiator offer	a=rtpmap:96 AMR/8000
Answer received from responder	m=audio xx RTP/AVP 98
	a=rtpmap:96 AMR/8000
	a=fmtp: 98 mode-set=0,1,2,3
Answer sent to initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000

## IWF Services

codecs	match	AMR/AMR-WB
payload formats	match	bandwidth-efficient (default value)
mode-sets	intersect	(0,1,2,3,4,5,6,7) / (0,1,2,3)
payload mapping	no match	96/98

Here, given that the answer mode-set (0,1,2,3) intersects with the offered mode-set (0,1,2,3,4,5,6,7), transcoding is not required; payload type mapping is used.

The following SDP reflects the case where the offer specifies a non-inclusive mode-set, and the answer specifies the inclusive mode-set.

Offer received from session initiator	m=audio xx RTP/AVP 98
	a=rtpmap:98 AMR/8000
	a=fmtp:98 mode-set=0,1,2,3
	INVITE SDP (partial, codec information only)
Offer sent to responder (per the net192 codec policy)	m=audio xx RTP/AVP 96 97 99 98
from AMR/PT96 media profile	a=rtpmap:96 AMR/8000
from AMR-WB/PT97 media profile	a=rtpmap:97 AMR-WB/16000/1
from AMR-WB/PT99 media profile	a=rtpmap:99 AMR-WB/16000/1
	a=fmtp:99 mode-set=0,1,2,3
initiator offer	a=rtpmap:98 AMR/8000
	a=fmtp:98 mode-set=0,1,2,3
Answer received from responder	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000
Answer sent to initiator	m=audio xx RTP/AVP 98
	a=rtpmap:98 AMR/8000
	a=fmtp:98 mode-set=0,1,2,3

codecs	match	AMR/AMR
payload formats	match	bandwidth-efficient (default value)
mode-sets	intersect	(0,1,2,3) / (0,1,2,3,4,5,6,7)
payload mapping	no match	98/96

Here, given that the offer mode-set (0,1,2,3) intersects with the answer mode-set (0,1,2,3,4,5,6,7), transcoding is not required; payload type mapping is used.

### Advanced Scenarios

Examples provided in this first section are supported by the following codec policy and media profile.

```

codec-policy
  name                net182
  allow-codecs        *
  add-codecs-on-egress AMR : PT97-5-6-7
  order-codecs

```

```

force-ptime                disabled
packetization-time        20
dtmf-in-audio             disabled

media-profile
  name                     AMR
  subname                  PT97-5-6-7
  media-type               audio
  payload-type             97
  transport                RTP/AVP
  req-bandwidth            0
  frames-per-packet        0
  parameters               mode-set="5, 6, 7"
  average-rate-limit       0
  peak-rate-limit          0
  max-burst-size           0
  sdp-rate-limit-headroom  0
  sdp-bandwidth            disabled
  police-rate              0
  standard-pkt-rate        0

```

The following SDP reflects the case where the offer and answer specify non-matching mode-sets.

Offer received from session initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000
	a=fmtp:96 mode-set=0,1,2,3,4
	INVITE SDP (partial, codec information only)
Offer sent to responder (per the net192 codec policy)	m=audio xx RTP/AVP 97 96
from AMR/PT97-5-6-7 media profile	a=rtpmap:97 AMR/8000
	a=fmtp:97 mode-set=5,6,7
initiator offer	a=rtpmap:96 AMR/8000
	a=fmtp:96 mode-set=1,2,3,4
Answer received from responder	m=audio xx RTP/AVP 97
	a=rtpmap:97 AMR/8000
	a=fmtp:97 mode-set=5,6,7
Answer sent to initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000
	a=fmtp:96 mode-set=0,1,2,3,4

codecs	match	AMR/AMR
payload formats	match	bandwidth-efficient (default value)
mode-sets	no match	(1,2,3,6) / (5,6,7)
payload mapping	no match	96/97

Here, given that the offer mode-set (1,2,3,4) does not intersect with the answer mode-set (5,6,7), transcoding is required.

The following SDP reflects the case where the offer and answer specify intersecting mode-sets.

## IWF Services

Offer received from session initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000
	a=fmtp:96 mode-set=0,1,2,3,4
	INVITE SDP (partial, codec information only)
Offer sent to responder (per the net192 codec policy)	m=audio xx RTP/AVP 97 96
from AMR/PT97-5-6-7 media profile	a=rtpmap:97 AMR/8000
	a=fmtp:97 mode-set=5,6,7
initiator offer	a=rtpmap:96 AMR/8000
	a=fmtp:96 mode-set=1,2,3,4
Answer received from responder	m=audio xx RTP/AVP 97
	a=rtpmap:97 AMR/8000
	a=fmtp:97 mode-set=4,5,6,7
Answer sent to initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000
	a=fmtp:96 mode-set=0,1,2,3,4

codecs	match	AMR/AMR
payload formats	match	bandwidth-efficient (default value)
mode-sets	no match	(1,2,3,4) / (4,5,6,7)
payload mapping	no match	96/97

Here, given that the offer mode-set (1,2,3,4) intersects with the answer mode-set (4,5,6,7), transcoding is not required; payload type mapping is used.

Examples provided in this next section are supported by the following codec policy and media profiles.

```

codec-policy
  name                net192
  allow-codecs        *
  add-codecs-on-egress AMR::PT96-MAXRED AMR::PT97-BE-0257
                     AMR::PT98-OA AMR::PT99-OA-0257

  order-codecs
  force-ptime         disabled
  packetization-time 20
  dtmf-in-audio      disabled

media-profile
  name                AMR
  subname             PT96-MAXRED
  media-type          audio
  payload-type        96
  transport           RTP/AVP
  req-bandwidth       0
  frames-per-packet   0
  parameters          max-red=220
  average-rate-limit  0
  peak-rate-limit     0
  max-burst-size      0

```

```

sdp-rate-limit-headroom 0
sdp-bandwidth disabled
police-rate 0
standard-pkt-rate 0

media-profile
  name AMR
  subname PT97-BE-0257
  media-type audio
  payload-type 97
  transport RTP/AVP
  req-bandwidth 0
  frames-per-packet 0
  parameters mode-set="0,2,5,7"
  average-rate-limit 0
  peak-rate-limit 0
  max-burst-size 0
  sdp-rate-limit-headroom 0
  sdp-bandwidth disabled
  police-rate 0
  standard-pkt-rate 0

media-profile
  name AMR
  subname PT98-OA
  media-type audio
  payload-type 98
  transport RTP/AVP
  req-bandwidth 0
  frames-per-packet 0
  parameters octet-align=1
  average-rate-limit 0
  peak-rate-limit 0
  max-burst-size 0
  sdp-rate-limit-headroom 0
  sdp-bandwidth disabled
  police-rate 0
  standard-pkt-rate 0

media-profile
  name AMR
  subname PT99-OA-0257
  media-type audio
  payload-type 99
  transport RTP/AVP
  req-bandwidth 0
  frames-per-packet 0
  parameters mode-set="0,2,5,7",octet-align=1
  average-rate-limit 0
  peak-rate-limit 0
  max-burst-size 0
  sdp-rate-limit-headroom 0
  sdp-bandwidth disabled
  police-rate 0
  standard-pkt-rate 0

```

The following SDP reflects the case where the offer and answer differ only in the inclusion of an AMR/AMR-WB parameter in the answer.

Offer received from session initiator	m=audio xx RTP/AVP 96
---------------------------------------	-----------------------

## IWF Services

	a=rtpmap:96 AMR/8000
	INVITE SDP (partial, codec information only)
Offer sent to responder (per the net192 codec policy)	m=audio xx RTP/AVP 97 98 99 96
from AMR/PT97-BE-0257 media profile	a=rtpmap:97 AMR/8000
	a=fmtp:97 mode-set=0,2,5,7
from AMR/PT98-OA media profile	a=rtpmap:98 AMR/8000
	a=fmtp:98 octet-align=1
from AMR/PT99-OA-0257 media profile	a=rtpmap:99 AMR/8000
	a=fmtp:99 mode-set=0,2,5,7 octet-align=1
initiator offer	a=rtpmap:96 AMR/8000
Answer received from responder	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000
	a=fmtp:96 max-red=220
Answer sent to initiator	m=audio xx RTP/AVP 96
	a=rtpmap:96 AMR/8000
	a=fmtp:96 max-red=220

codecs	match	AMR/AMR
payload formats	match	bandwidth-efficient (default value)
mode-sets	match	inclusive mode-set (0,1,2,3,4,5,6,7)
payload mapping	match	96/96

Here, given that the offer/answer are identical except for the AMR max-red parameter, neither transcoding nor payload type matching is required.

The following SDP reflects the case where the offer and answer differ only in the inclusion of an AMR/AMR-WB parameter in the offer.

Offer received from session initiator	m=audio xx RTP/AVP 97
	a=rtpmap:97 AMR/8000
	a=fmtp:97 mode-set=0,2,5,7
	a=fmtp:97 max-red=220
	INVITE SDP (partial, codec information only)
Offer sent to responder (per the net192 codec policy)	m=audio xx RTP/AVP 96 98 99 97
from AMR/PT96-MAXRED media profile	a=rtpmap:96 AMR/8000
	a=fmtp:96 max-red=220
from AMR/PT98-OA media profile	a=rtpmap:98 AMR/8000
	a=fmtp:98 octet-align=1
from AMR/PT99-OA-0257 media profile	a=rtpmap:99 AMR/8000



	a=fmtp:99 mode-set=0,2,5,7 octet-align=1
initiator offer	a=rtpmap:97 AMR/8000
	a=fmtp:97 mode-set=0,2,5,7 max-red=220
Answer received from responder	m=audio xx RTP/AVP 97
	a=rtpmap:97 AMR/8000
	a=fmtp:97 mode-set=0,2,5,7
Answer sent to initiator	m=audio xx RTP/AVP 97
	a=rtpmap:97 AMR/8000
	a=fmtp:97 mode-set=0,2,5,7

codecs	match	AMR/AMR
payload formats	match	bandwidth-efficient (default value)
mode-sets	match	(0,2,5,7) / (0,2,5,7)
payload mapping	match	97/97

Here, given that the offer/answer are identical except for the AMR max-red parameter, neither transcoding nor payload type matching is required.

## ACLI Configuration

To enable AMR Mode-set payload type SDP rewriting:

1. Navigate to the media-manager-config element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# media-manager-config
ACMEPACKET(media-manager-config)#
```

2. options—Set the options parameter by typing options, a Space, the option-name audio-payload-type-mapping=yes with a “plus” sign in front of it, and then press Enter.

```
ACMEPACKET(media-manager-config)# options +audio-payload-type-mapping=yes
```

If you type the option without the “plus” sign, you will overwrite any previously configured options. In order to append the new options to the SIP interface configuration’s options list, you must prepend the new option with a “plus” sign as shown in the previous example.

3. Save and activate your configuration.

## RFC 3326 Support

This section explains the Oracle Communications Session Border Controller’s ability to map Q.850 cause values with SIP responses for calls that require IWF.

RFC 3326 defines a header that might be included in any in-dialogue request. This reason header includes cause values that are defined as either a SIP response code or ITU-T Q.850 cause values. You can configure the Oracle Communications Session Border Controller to support sending and receiving RFC 3326 in SIP messages for:

- Mapping H.323 Q.850 cause values to SIP responses with reason header and cause value
- Mapping SIP response messages and RFC 3326 reason header and cause
- Locally generated SIP response with RFC 3326 reason header and cause

As specified in RFC 3326, the Oracle Communications Session Border Controller sends SIP responses to the softswitch that contain the received Q.850 cause code and the reason.

Though the Oracle Communications Session Border Controller can generate RFC 3326 headers, the default behavior for this feature is disabled. Furthermore, the Oracle Communications Session Border Controller can receive and pass SIP error messages (4xx, 5xx, and 6xx) that contain the SIP reason header with a Q.850 cause code and reason (as specified in RFC 3326). If the Oracle Communications Session Border Controller receives an error message without the Reason header, then the Oracle Communications Session Border Controller is not required to insert one.

In calls that require IWF, the Q.850 cause generated in the SIP response are the same as the cause received in the following H.225 messages: Disconnect, Progress, Release, Release Complete, Resume Reject, Status, and Suspend Reject. In addition, the Q.850 cause codes that the Oracle Communications Session Border Controller receives in RFC 3326 headers are passed to the H.323 part of the call unmodified; the H.323 call leg uses this cause code for releasing the call.

For interworking calls between SIP and H.323, you can configure:

- Mappings for SIP status codes to Q.850 values
- Mappings for particular Q.850 cause codes to SIP status codes

If it cannot find the appropriate mapping, then the Oracle Communications Session Border Controller uses default mappings defined in the Default Mappings table below.

The following describes how the Oracle Communications Session Border Controller handles different IWF call scenarios:

- SIP request containing a Reason header—When it receives a request containing a Reason header, the Oracle Communications Session Border Controller determines if the request is a SIP BYE or SIP CANCEL message. RFC 3326 states that the Reason header is mainly used for these types of requests. If there is a Reason header and it contains the Q.850 cause value, then the Oracle Communications Session Border Controller releases the call on the H.323 side using the specified cause value.
- SIP response—When it receives the error response to an initial SIP INVITE, the Oracle Communications Session Border Controller uses its SIP-Q.850 map to determine the Q.850 that it will use to release the call. If there is not a map entry, then the Oracle Communications Session Border Controller uses the default mappings shown in the Default Mappings table.
- Active call released from the H.323 side—If an active call is released from the H.323 side, the Oracle Communications Session Border Controller checks the outgoing realm (the SIP side) to see if the addition of the Reason header is enabled. If it is, then the Oracle Communications Session Border Controller adds the Reason header in the SIP BYE request with the Q.850 value it received from the H.323 side.
- Error during setup of the call on the H.323 side—In the event of an error during setup on the H.323 side of the call, the Oracle Communications Session Border Controller needs to send:

- An error response, if this is a SIP to H.323 call
- A SIP CANCEL, if this is a H.323 to SIP call and the H.323 side hangs up before the call is answered on the SIP side

In this case, the Oracle Communications Session Border Controller checks to see if adding the Reason header is enabled in the IWF configuration. If it is, then the Oracle Communications Session Border Controller adds the Reason header with the Q.850 cause value it received from the H.323 side.

- Call released due to a Oracle Communications Session Border Controller error—If the call is released due a Oracle Communications Session Border Controller error and adding the Reason header is enabled in the IWF configuration, the error response to the initial INVITE contains the Reason header. The Oracle Communications Session Border Controller checks the SIP to Q.850 map configurations to determine whether or not the SIP error response code it is generating is configured. If it is, then the system maps according to the configuration. If it is not, the Oracle Communications Session Border Controller derives cause mapping from the default table.

Like the configuration for SIP-only calls that enable this feature, you can set a parameter in the IWF configuration that enables adding the Reason header in the SIP requests or responses.

## Default Mappings

This table defines the default mappings the Oracle Communications Session Border Controller uses when it cannot locate an appropriate entry that you have configured.

Q.850 Cause Value		SIP Status		Comments
1	Unallocated number	404	Not found	
2	No route to specified transit network	404	Not found	
3	No route destination	404	Not found	
16	Normal calling clearing		BYE message	A call clearing BYE message containing cause value 16 normally results in the sending of a SIP BYE or CANCEL request. However, if a SIP response is to be sent to the INVITE request, the default response code should be used.
17	User busy	486	Busy here	
18	No user responding	408	Request timeout	
19	No answer from the user	480	Temporarily unavailable	
20	Subscriber absent	480	Temporarily unavailable	
21	Call rejected	603	Decline (if location filed in Cause information element indicates user; otherwise 403 Forbidden is used)	
22	Number changed	301	Moved permanently (if information in diagnostic field of Cause information element is suitable for generating SIP Contact header; otherwise 410 Gone is used)	
23	Redirection to new destination	410	Gone	
25	Exchange routing error	483	Too many hops	
27	Destination out of order	502	Bad gateway	
28	Address incomplete	484	Address incomplete	
29	Facility rejected	501	Not implemented	
31	Normal, unspecified	480	Temporarily unavailable	
34	No circuit, channel unavailable	503	Service unavailable	
38	Network out of order	503	Service unavailable	

Q.850 Cause Value		SIP Status		Comments
41	Temporary failure	503	Service unavailable	
42	Switching equipment congestion	503	Service unavailable	
47	Resource unavailable unspecified	503	Service unavailable	
55	Incoming calls barred with CUG	403	Forbidden	
57	Bearer capability not authorized	403	Forbidden	
58	Bearer capability not presently available	503	Service unavailable	
65	Bearer capability not implemented	488	Not acceptable here	
69	Requested facility not implemented	501	Not implemented	
70	Only restricted digital information available	488	Not acceptable here	
79	Service or option not implemented, unspecified	501	Not implemented	
87	User not member of CUG	403	Forbidden	
88	Incompatible destination	503	Service unavailable	
102	Recovery on timer expiry	504	Server time-out	

## RFC 3326 Support Configuration

To configure a SIP status to Q.850 Reason with cause mapping:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-q850-map** and press Enter.

```
ACMEPACKET(session-router)# sip-q850-map
```

```
ACMEPACKET(sip-q850-map)#
```

4. Type **entries** and press Enter.

```
ACMEPACKET(sip-q850-map)# entries
```

```
ACMEPACKET(sip-q850-map-entry)#
```

From here, you can view the entire menu for the SIP status to Q.850 Reason with cause mapping entries configuration by typing a **?**.

5. **sip-status**—Set the SIP response code that you want to map to a particular Q.850 cause code and reason. There is no default, and the valid range for values is:
  - Minimum—100
  - Maximum—699
6. **q850-cause**—Set the Q.850 cause code that you want to map to the SIP response code that you set in step 5. There is no default.
7. **q850-reason**—Set the Q.850 reason corresponding to the Q.850 cause code that you set in step 6. There is no default. If your value has spaces between characters, then your entry must be surrounded by quotation marks.
8. Repeat this process to create the number of local response map entries that you need.
9. Save and activate your configuration for changes to take effect.

To configure a Q.850 cause to a SIP status with reason mapping:

10. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

11. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

12. Type **sip-q850-map** and press Enter.

```
ACMEPACKET (session-router) # q850-sip-map
ACMEPACKET (q850-sip-map) #
```

13. Type **entries** and press Enter.

```
ACMEPACKET (q850-sip-map) # entries
ACMEPACKET (q850-sip-map-entry) #
```

From here, you can view the entire menu for the Q.850 cause to a SIP response code with reason mapping entries configuration by typing a **?**.

14. **q850-cause**—Set the Q.850 cause code that you want to map to a SIP status with reason. There is no default.
15. **sip-status**—Set the SIP response code to which you want to map the Q.850 cause that you set in step 5. There is no default, and the valid range for a value is
  - Minimum—100
  - Maximum—699
16. **sip-reason**—Set the reason that you want to use with the SIP response code that you specified in step 6. There is no default. If your value has spaces between characters, then your entry must be surrounded by quotation marks.
17. Repeat this process to create the number of local response map entries that you need.

To enable the Oracle Communications Session Border Controller to add the Reason header for calls that require IWF:

18. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

19. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

20. Type **iwf-config** and press Enter.

```
ACMEPACKET (session-router) # iwf-config
ACMEPACKET (iwf-config) #
```

21. **add-reason-header**—Enable this parameter to add the Reason header to IWF calls. The default is **disabled**. Valid values are:
  - enabled | disabled

## IWF Privacy Caller Privacy on Unsecure Networks

This feature enables bi-directional SIP/H.323 IWF support for CPID hiding by using the presentation indicators in the Calling Party Number information element for H.323 signaling, and RFC 3325-based privacy support for SIP signaling. It lets the Oracle Communications Session Border Controller insert the P-Asserted-Identity and the Privacy header in the INVITE when the presentation indicator is set to restricted.

The presence, or absence, of P-Asserted-Identity and Privacy headers in the SIP INVITE informs the remote SIP proxy or endpoint to either block or advertise the CPID.

### About the Presentation Indicator

When address information represents a telephone number, the relevant information can appear in the Calling Party Number information element (IE). This IE contains the caller's number, information about the number, and presentation and screening indicators found in octet 3a. In order to prevent a calling party number to be passed through, the presentation indicator parameter (octet 3a) in the Calling Party IE must be set to a value other than 00.

In a H.323 to SIP IWF call, octet 3a in the Q.931 message indicates the caller's preference for CPID restriction. If bits 7 and 6 are set to (0 1), the presentation is restricted and the outbound SIP INVITE from the IWF stack must be constructed as such.

### H.323 to SIP IWF Call

When the presentation indicator in the calling party IE is set to restricted, the INVITE's From and Contact headers sent from to sipd will be modified according to RFC 3325. When the Oracle Communications Session Border Controller receives calls initiated as H.323, it will recognize the caller's presentation bits as defined in Q.931 and use that information to construct a SIP INVITE in accordance with the user's indicated preference.

- Inclusion of a P-Asserted-Identity header in the INVITE, containing the calling party's CPID and the Oracle Communications Session Border Controller's IP address, constructed as a SIP URI (same mechanism used to construct the From-URI today).
- Addition of a Privacy header with its value set to id. This addition indicates to the upstream proxies and gateways that the caller address is to be hidden.

The sipd will either proxy or strip these headers according to RFC 3325, depending on the SIP interface and SIP session agent configurations.

#### Example 1 SETUP Sent from h323d to Remote H.323 Endpoints

```
Q.931
Protocol discriminator: Q.931
Call reference value length: 2
Call reference flag: Message sent from originating side
Call reference value: 2F62
Message type: SETUP (0x05)
Bearer capability
Information element: Bearer capability
Length: 3
...0 1000 = Information transfer capability: Unrestricted digital information
(0x08)
.00. .... = Coding standard: ITU-T standardized coding (0x00)
1... .... = Extension indicator: last octet
...1 0011 = Information transfer rate: 384 kbit/s (0x13)
.00. .... = Transfer mode: Circuit mode (0x00)
1... .... = Extension indicator: last octet
...0 0101 = User information layer 1 protocol: Recommendation H.221 and H.242
(0x05)
1... .... = Extension indicator: last octet
Display 'jdoe\000'
Information element: Display
Length: 9
```

```

Display information: jdoe\000
Calling party number
Information element: Calling party number
Length: 2
.... 0000 = Numbering plan: Unknown (0x00)
.000 .... = Number type: Unknown (0x00)
0... .... = Extension indicator: information continues through the next octet
.... ..00 = Screening indicator: User-provided, not screened (0x00)
.01. .... = Presentation indicator: Presentation restricted (0x01)
1... .... = Extension indicator: last octet

```

### Example 2 INVITE from h323d to sipd

The two new headers will be stripped by the sipd when the INVITE is sent to a untrusted SIP proxy or endpoint and will be proxied over to a trusted SIP proxy or endpoint.

```

INVITE sip:780@192.168.200.6:5060;acme_realm=internal SIP/2.0
Via: SIP/2.0/UDP
127.0.0.1:5070;branch=z9hG4bKIWF00000510d031s9kou5c0;acme_irealm=external
Contact: "Anonymous"<sip:anonymous@127.0.0.1:5070
GenericID: 7400000@000825010100
Supported: 100rel
From: "Anonymous"<sip:anonymous@anonymous.invalid>;tag=0000004a000d8cc0
To: <sip:780@192.168.200.6:5060
Call-ID: 7f00000113ce0000004a000d88d8@127.0.0.1
CSeq: 2 INVITE
P-Asserted-Identity: "jdoe"<sip:42343@192.168.200.68:5060>
Privacy: id
Content-Length: 175
Content-Type: application/sdp
v=0
o=IWF 3 3 IN IP4 192.168.1.6
s=H323 Call
c=IN IP4 192.168.1.6
t=0 0
m=audio 5666 RTP/AVP 0 101 18
a=rtpmap:0 PCMU/8000/1
a=rtpmap:101 telephone-event/8000/1
a=fmtp:101 0-15
a=rtpmap:18 G729/8000/1
a=fmtp:18 annexb=no
m=video 5668 RTP/AVP 31
a=rtpmap:31 H261/9000/1

```

## SIP to H.323

For a SIP to H.323 call, the Oracle Communications Session Border Controller must recognize the caller's Privacy request and set the presentation bits accordingly when constructing the outbound RAS/SETUP message. It must check SIP calls for the Privacy header (with value set to id). If this header is present, the SETUP's octet 3a's presentation bits must be set to restricted.

The Oracle Communications Session Border Controller does not support any other value for the Privacy header. For those calls, the SETUP will not include a presentation indicator.

### Example INVITE from SIP End Point to sipd

```

Apr 21 08:50:38.786 On [0:0]192.168.200.68:5060 received from
192.168.200.6:5062
INVITE sip:800@192.168.200.68:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.200.6:5062
From: anonymous <sip:anonymous@192.168.200.6:5062>;tag=1
To: sut <sip:800@192.168.200.68:5060
P-Asserted-Identity: sipp <sip:7789@192.168.200.6:5062
Privacy: id

```

```
Call-ID: 1.1688.192.168.200.6@sipp.call.id
Cseq: 1 INVITE
Contact: sip:anonymous@192.168.200.6:5062
Max-Forwards: 70
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 136
v=0
o=user1 53655765 2353687637 IN IP4 127.0.0.1
s=-
t=0 0
c=IN IP4 127.0.0.1
m=audio 10000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
Sample INVITE from sipp to h323d
Apr 21 08:50:38.807 On 127.0.0.1:5070 received from 127.0.0.1:5060
INVITE sip:800@127.0.0.1:5070;acme_sag=sag1;acme_irealm=internal SIP/2.0
Via: SIP/2.0/UDP 127.0.0.1:5060;branch=z9hG4bK0804o700c0f0t9gpj0g0.1
From: anonymous <sip:anonymous@192.168.200.6:5062>;tag=SDm8kvc01-1
To: sut <sip:800@192.168.200.68:5060
P-Asserted-Identity: sipp <sip:7789@192.168.200.6:5062
```

**Privacy: id**

```
Call-ID: SDm8kvc01-083221d8c0fa33f71ae85dd6ed0e4ea4-06ahc21
Cseq: 1 INVITE
```

```
Contact: <sip:anonymous@192.168.200.68:5060;transport=udp
```

```
Max-Forwards: 69
```

```
Subject: Performance Test
```

```
Content-Type: application/sdp
```

```
Content-Length: 136
```

```
GenericID: 9883100005@000825010100
```

```
v=0
```

```
o=user1 53655765 2353687637 IN IP4 127.0.0.1
```

```
s=-
```

```
t=0 0
```

```
c=IN IP4 127.0.0.1
```

```
m=audio 10000 RTP/AVP 0
```

```
a=rtpmap:0 PCMU/8000
```

```
Sample SETUP sent from h323d to remote H323 EP
```

```
Q.931
```

```
Protocol discriminator: Q.931
```

```
Call reference value length: 2
```

```
Call reference flag: Message sent from originating side
```

```
Call reference value: 664D
```

```
Message type: SETUP (0x05)
```

```
Bearer capability
```

```
Information element: Bearer capability
```

```
Length: 3
```

```
...1 0000 = Information transfer capability: 3.1 kHz audio (0x10)
```

```
.00. .... = Coding standard: ITU-T standardized coding (0x00)
```

```
1... .... = Extension indicator: last octet
```

```
...1 0000 = Information transfer rate: 64 kbit/s (0x10)
```

```
.00. .... = Transfer mode: Circuit mode (0x00)
```

```
1... .... = Extension indicator: last octet
```

```
...0 0011 = User information layer 1 protocol: Recommendation G.711 A-law (0x03)
```

```
1... .... = Extension indicator: last octet
```

```
Display 'anonymous'
```

```
Information element: Display
```

```
Length: 9
```

```
Display information: anonymous
```

```
Calling party number
```

```
Information element: Calling party number
```

```
Length: 2
```

```
.... 0000 = Numbering plan: Unknown (0x00)
```



```

.000 .... = Number type: Unknown (0x00)
0... .... = Extension indicator: information continues through the
next octet
.... ..00 = Screening indicator: User-provided, not screened (0x00)
.01. .... = Presentation indicator: Presentation restricted (0x01)
1... .... = Extension indicator: last octet

```

## IWF Privacy Caller Privacy on Secure Connections

In prior releases, when the H.323 endpoint sends a SETUP with presentation indicator set to allowed, the Oracle Communications Session Border Controller does not insert the P-Asserted-Identity in the INVITE. The SIP INVITE needs the P-Asserted-Identity header to support calling line identification presentation (CLIP) to calling line identification restriction (CLIR) in an IP multimedia subsystem (IMS) solution. This feature lets the Oracle Communications Session Border Controller insert the P-Asserted-Identity in the INVITE when the presentation indicator is set to allowed.

- CLIP is a service provided to the called party that allows the display of the calling number (caller ID). The user-provided calling number must be transported from the caller to the called party.
- CLIR is a service provided to the calling party that lets it indicate whether or not the calling number is to be displayed to the called party. It sets a calling number presentation indicator to allowed or restricted. Regulations require that network administrations remove the calling number before it is sent to the called party, if the calling party has so requested.

### H.323 to SIP IWF

When the Oracle Communications Session Border Controller translates incoming H.323 messages to SIP on a secure connection (which means the Oracle Communications Session Border Controller can rely on the data sent from the originator); it will translate the information in the H.323 messages into SIP messages as detailed in the following sections.

#### Calls with Presentation Allowed

When the Oracle Communications Session Border ControllerC receives a SETUP from the H.323 domain where presentation is allowed, it generates an INVITE to the SIP domain with the following header. (Presentation is allowed when the calling party's information element presentation indicator (octet 3a) equals 00.)

- P-Asserted-ID: the userpart should be derived from the Calling Party Number Information Element digits.

### H.323 to SIP

When h323d receives a SETUP with the calling party's information element presentation indicator set to allowed, the Oracle Communications Session Border Controller will add the P-Asserted-Identity header to the INVITE. The P-Asserted-Identity is very similar to the FROM header, except for the tag.

#### Sample SETUP sent from h323d to Remote H323 Endpoints

```

Q.931
Protocol discriminator: Q.931
Call reference value length: 2
Call reference flag: Message sent from originating side
Call reference value: 2F62
Message type: SETUP (0x05)
Bearer capability
Information element: Bearer capability
Length: 3
...0 1000 = Information transfer capability: Unrestricted digital information
(0x08)
.00. .... = Coding standard: ITU-T standardized coding (0x00)
1... .... = Extension indicator: last octet
...1 0011 = Information transfer rate: 384 kbit/s (0x13)

```

```
.00. .... = Transfer mode: Circuit mode (0x00)
1... .... = Extension indicator: last octet
...0 0101 = User information layer 1 protocol: Recommendation H.221 and H.242
(0x05)
1... .... = Extension indicator: last octet
Display 'jdoe\000'
Information element: Display
Length: 9
Display information: jdoe\000
Calling party number: '42343'
Information element: Calling party number
Length: 6
.... 1001 = Numbering plan: Private numbering (0x09)
.110 .... = Number type: Abbreviated number (0x06)
0... .... = Extension indicator: information continues through the next octet
.... ..00 = Screening indicator: User-provided, not screened (0x00)
.00. .... = Presentation indicator: Presentation allowed (0x00)
1... .... = Extension indicator: last octet
Calling party number digits: 42343
```

### SIP to H.323

When the sipd receives an INVITE with the P-Asserted-Identity header but without the Privacy header, the Oracle Communications Session Border Controller will set the presentation indicator to allowed in H.323's SETUP.

When the Privacy header is present with the value id, the presentation indicator will be set to restricted. The Oracle Communications Session Border Controller does not support any other value for the Privacy header and so for those call flows, the presentation indicator will be absent in the SETUP.

#### Example 1 INVITE from sip EP to sipd

```
Apr 20 04:43:54.220 On [0:0]192.168.200.68:5060 received from
192.168.200.6:5062
INVITE sip:800@192.168.200.68:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.200.6:5062
From: sipp <sip:7789@192.168.200.6:5062>;tag=1
To: sut <sip:800@192.168.200.68:5060>
P-Asserted-Identity: sipp <sip:7789@192.168.200.6:5062>
Call-ID: 1.1336.192.168.200.6@sipp.call.id
Cseq: 1 INVITE
Contact: sip:7789@192.168.200.6:5062
Max-Forwards: 70
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 136
^M
v=0
o=user1 53655765 2353687637 IN IP4 127.0.0.1
s=-
t=0 0
c=IN IP4 127.0.0.1
m=audio 10000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

#### Example INVITE from sipd to h323d

```
Apr 20 04:43:54.240 On 127.0.0.1:5070 received from 127.0.0.1:5060
INVITE sip:800@127.0.0.1:5070;acme_sag=sag1;acme_irealm=internal SIP/2.0
Via: SIP/2.0/UDP 127.0.0.1:5060;branch=z9hG4bK000c0210385hv9gpt001.1
From: sipp <sip:7789@192.168.200.6:5062>;tag=SDk0jpc01-1
To: sut <sip:800@192.168.200.68:5060>
Call-ID: SDk0jpc01-8e15e11e7f9a20523462972843c7e579-06ahc21
Cseq: 1 INVITE
Contact: <sip:7789@192.168.200.68:5060;transport=udp>
```

```

Max-Forwards: 69
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 136
GenericID: 160400004@000825010100
v=0
o=user1 53655765 2353687637 IN IP4 127.0.0.1
s=-
t=0 0
c=IN IP4 127.0.0.1
m=audio 10000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
Sample SETUP sent from h323d to remote H323 EP
Q.931
  Protocol discriminator: Q.931
Call reference value length: 2
  Call reference flag: Message sent from originating side
  Call reference value: 664D
  Message type: SETUP (0x05)
  Bearer capability
    Information element: Bearer capability
    Length: 3
    ...1 0000 = Information transfer capability: 3.1 kHz audio (0x10)
    .00. .... = Coding standard: ITU-T standardized coding (0x00)
    1... .... = Extension indicator: last octet
    ...1 0000 = Information transfer rate: 64 kbit/s (0x10)
    .00. .... = Transfer mode: Circuit mode (0x00)
    1... .... = Extension indicator: last octet
    ...0 0011 = User information layer 1 protocol: Recommendation G.711 A-law (0x03)
    1... .... = Extension indicator: last octet
  Display 'sipp'
    Information element: Display
    Length: 4
    Display information: sipp
  Calling party number: '7789'
    Information element: Calling party number
    Length: 6
    .... 1001 = Numbering plan: Private numbering (0x09)
    .110 .... = Number type: Abbreviated number (0x06)
    0... .... = Extension indicator: information continues through the
next octet
    .... ..00 = Screening indicator: User-provided, not screened (0x00)
    .00. .... = Presentation indicator: Presentation all 1... .... =
Extension indicator: last octet
  Calling party number digits: 7789

```

## IWF Privacy Extensions for Asserted Identity in Untrusted Networks

For IWF privacy, the Oracle Communications Session Border Controller supports:

- IWF caller privacy on unsecure networks—A variant of RFC 3325, where the P-Asserted-Id is inserted when the presentation indicator is set to allowed. This feature enables bi-directional SIP/H.323 IWF support for CPID hiding by using the presentation indicators in the Calling Party Number information element for H.323 signaling, and RFC 3325-based privacy support for SIP signaling. It allows the Oracle Communications Session Border Controller to insert the P-Asserted-Identity and the Privacy header in the INVITE when the presentation indicator is set to restricted.

The presence, or absence, of P-Asserted-Identity and Privacy headers in the SIP INVITE informs the remote SIP proxy or endpoint to either block or advertise the CPID.

- IWF caller privacy on secure connections—When the H.323 endpoint sends a SETUP with presentation indicator set to allowed, the Oracle Communications Session Border Controller does not insert the P-Asserted-Identity in the INVITE. The SIP INVITE needs the P-Asserted-Identity header to support calling line identification presentation (CLIP) to calling line identification restriction (CLIR) in an IP multimedia subsystem (IMS) solution. This feature allows the Oracle Communications Session Border Controller to insert the P-Asserted-Identity in the INVITE when the presentation indicator is set to allowed.

Now the Oracle Communications Session Border Controller supports an enhancement to IWF caller privacy where the P-Preferred-Identity is inserted instead of the P-Asserted-Identity.

In this implementation, when the incoming H.323 Setup message has a presentation indicator set to restricted and the ingress H.323 session agent has the new PPreferredId option configured, the Oracle Communications Session Border Controller sends the Privacy header with P-Preferred-Identity (instead of P-Asserted-Identity).

## IWF Call Originating in H.323

This section shows an example H.323 Setup that arrives from an H.323 endpoint, and how the Oracle Communications Session Border Controller adds the P-Preferred-Identity header (which has calling party number information) and the Privacy header to the SIP INVITE.

### Sample H.323 Setup from a Remote Endpoint

```
Q.931
  Protocol discriminator: Q.931
  Call reference value length: 2
  Call reference flag: Message sent from originating side
  Call reference value: 2FB6
  Message type: SETUP (0x05)
  Bearer capability
    Information element: Bearer capability
    Length: 3
    ...0 1000 = Information transfer capability: Unrestricted digital
information (0x08)
    .00. .... = Coding standard: ITU-T standardized coding (0x00)
    1... .... = Extension indicator: last octet
    ...1 0011 = Information transfer rate: 384 kbit/s (0x13)
    .00. .... = Transfer mode: Circuit mode (0x00)
    1... .... = Extension indicator: last octet
    ...0 0101 = User information layer 1 protocol: Recommendation H.221 and H.242
(0x05)
    1... .... = Extension indicator: last octet
  Display 'rdoe\000'
    Information element: Display
    Length: 9
    Display information: jdoe\000
  Calling party number: '42343'
    Information element: Calling party number
    Length: 6
    .... 0001 = Numbering plan: E.164 ISDN/telephony numbering (0x01)
    .000 .... = Number type: Unknown (0x00)
    0... .... = Extension indicator: information continues through the
next octet
    .... ..00 = Screening indicator: User-provided, not screened (0x00)
    .01. .... = Presentation indicator: Presentation restricted (0x01)
    1... .... = Extension indicator: last octet
    Calling party number digits: 42343
E.164 Calling party number digits: 42343
  Called party number: '780'
    Information element: Called party number
    Length: 4
    .... 0001 = Numbering plan: E.164 ISDN/telephony numbering (0x01)
    .000 .... = Number type: Unknown (0x00)
    1... .... = Extension indicator: last octet
```

```

    Called party number digits: 780
    E.164 Called party number digits: 780
User-user
    Information element: User-user
    Length: 161
    Protocol discriminator: X.208 and X.209 coded user information

```

### Sample SIP INVITE from the SBC to a SIP Endpoint

```

Aug 29 15:46:25.214 On [0:0]192.168.200.68:5060 sent to 192.168.200.6:5060
INVITE sip:780@192.168.200.6:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.200.68:5060;branch=z9hG4bK6810pr20205h2akqe381.1
Contact: "Anonymous"<sip:anonymous@192.168.200.68:5060;transport=udp>
Supported: 100rel
From:
"Anonymous"<sip:anonymous@anonymous.invalid>;tag=SDfd9sa01-000000ba00023280
To: <sip:780@192.168.200.6:5060>
Call-ID: SDfd9sa01-6f93292521b83a0980647f34451c5afd-06ahc21
CSeq: 2 INVITE
P-Preferred-Identity: "rdoe"<sip:42343@192.168.200.68:5060>
<b>Privacy: id<\b>
Content-Length: 180
Content-Type: application/sdp
Max-Forwards: 70
v=0
o=IWF 5 5 IN IP4 192.168.200.5
s=H323 Call
c=IN IP4 192.168.200.65
t=0 0
m=audio 5010 RTP/AVP 0
a=rtpmap:0 PCMU/8000/1
m=video 5014 RTP/AVP 31
a=rtpmap:31 H261/9000/1

```

## Before You Configure

Before you configure your Oracle Communications Session Border Controller to support this feature, note the following considerations:

- The ingress H.323 session agent cannot be configured with the NoPAssertedId option
- For use in Release 4.1.1 and higher, the global SIP configuration should be configured with the disable-ppi-to-pai option; the older disable-privacy option will also work

## P-Preferred-Identity Configuration

To enable the inclusion of P-Preferred-Identity:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

4. Select the session agent where you want to apply this feature.

```
ACMEPACKET(session-agent)# select
<hostname>:
1: 204.12.60.5      realm=private
```

```
2: 124.21.5.3      realm=public
selection:1
ACMEPACKET(session-agent) #
```

5. **options**—Set the options parameter by typing **options**, a Space, the option name preceded by a plus sign (+) (**PPreferredId**), and then press Enter.

```
ACMEPACKET(realm-config) # options +PPreferredId
```

If you type options PPreferredId, you will overwrite any previously configured options. In order to append the new option to the session agent's options list, you must prepend the new option with a plus sign as shown in the previous example.

6. Save and activate your configuration.

## IWF Privacy for Business Trunking

---

The Oracle Communications Session Border Controller supports IWF Privacy: Caller Privacy on Unsecure Networks and IWF Privacy: Caller Privacy on Secure Connections, but IWF Privacy for Business Trunking, supports the case where SIP and H.323 PBXs are connected to the core IMS system. Traffic originated at the IP PBXs terminates either at other PBXs or at the PSTN, and includes the possibility of accepting incoming traffic from the PSTN. CLIP and CLIR must be supported for calls in either direction for calls that require interworking between SIP and H.323. Unlike the two features described above, this new feature supports the fact that only a network-based application server has sufficient privilege to assert the identity of the calling party.

Thus, for this feature, the Oracle Communications Session Border Controller does not force privacy. Instead, the implemented feature assumes that the H.323 session agent is an IP PB X, and the Oracle Communications Session Border Controller only indicates to the SIP core that privacy is being requested. In other words, the Oracle Communications Session Border Controller is not required to interwork the H.323 presentation indicator parameter to RFC 3325 by including the P-Asserted-Identity header. The indication to the SIP core that privacy is being requested excludes identity assertion.

You configure this feature using two session agent options:

- allowCPN—Set in the egress H.323 session agent, allows the Oracle Communications Session Border Controller to send the calling party number information element (IE), even when the presentation indicator is set to restricted.
- NoPAssertedId—Set in the ingress H.323 session agent; when the incoming SETUP message has the presentation indicator is set to restricted, instructs the Oracle Communications Session Border Controller to send a Privacy header without the P-Asserted-Identity and not to make the From header anonymous.

### A Call Originating in H.323

This section describes for the IWF Privacy for Business trunking feature works for a call originating in H.323 that requires interworking to SIP.

When the Oracle Communications Session Border Controller receives an H.323 SETUP with a presentation indicator of the calling party information element (IE) is set to restricted and this SETUP was received from a session agent is configured with the NoPAssertedID option, the Oracle Communications Session Border Controller only adds the Privacy header with the value ID. In this case, there will be no P-Asserted-Identity and the From header will contain the calling Party information that was extracted from the callingPartyIE. The Oracle Communications Session Border Controller assumes that the PBX will send the callingPartyNumber in the IE, even though it would like to have the calling party number restricted.

### Sample SETUP Message from an H.323 Endpoint

```
Q.931
Protocol discriminator: Q.931
Call reference value length: 2
Call reference flag: Message sent from originating side
Call reference value: 2FB6
```

```

    Message type: SETUP (0x05)
    Bearer capability
      Information element: Bearer capability
Length: 3
  ...0 1000 = Information transfer capability: Unrestricted digital
information (0x08)
  .00. .... = Coding standard: ITU-T standardized coding (0x00)
  1... .... = Extension indicator: last octet
  ...1 0011 = Information transfer rate: 384 kbit/s (0x13)
  .00. .... = Transfer mode: Circuit mode (0x00)
  1... .... = Extension indicator: last octet
  ...0 0101 = User information layer 1 protocol: Recommendation H.221
and H.242 (0x05)
  1... .... = Extension indicator: last octet
  Display 'jdoe\000'
    Information element: Display
Length: 9
  Display information: jdoe\000
  Calling party number: '42343'
    Information element: Calling party number
    Length: 6
    .... 0001 = Numbering plan: E.164 ISDN/telephony numbering (0x01)
    .000 .... = Number type: Unknown (0x00)
    0... .... = Extension indicator: information continues through the
next octet
    .... ..00 = Screening indicator: User-provided, not screened (0x00)
    .01. .... = Presentation indicator: Presentation restricted (0x01)
    1... .... = Extension indicator: last octet
    Calling party number digits: 42343
    E.164 Calling party number digits: 42343
  Called party number: '780'
    Information element: Called party number
    Length: 4
    .... 0001 = Numbering plan: E.164 ISDN/telephony numbering (0x01)
    .000 .... = Number type: Unknown (0x00)
    1... .... = Extension indicator: last octet
    Called party number digits: 780
    E.164 Called party number digits: 780
User-user
  Information element: User-user
  Length: 161
  Protocol discriminator: X.208 and X.209 coded user information

```

### Sample INVITE from the Oracle Communications Session Border Controller to the SIP Endpoint

```

May 5 15:11:51.996 On [0:0]192.168.200.68:5060 sent to 192.168.200.6:5060
INVITE sip:780@192.168.200.6:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.200.68:5060;branch=z9hG4bK00020a20eg11s94pg700.1
Contact: "jdoe"<sip:42343@192.168.200.68:5060;transport=udp>
Supported: 100rel
From: "jdoe"<sip:42343@192.168.200.68:5060>;tag=SDetur801-00000194000e2ce8
To: <sip:780@192.168.200.6:5060>
Call-ID: SDetur801-231c7b30909ca525ce12cbfeb57754ea-06ahc21
CSeq: 2 INVITE
Privacy: id
Content-Length: 231
Content-Type: application/sdp
Max-Forwards: 70
v=0
o=IWF 2 2 IN IP4 192.168.200.65
s=H323 Call
c=IN IP4 192.168.200.65
t=0 0

```

```
m=audio 5004 RTP/AVP 8 0
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000/1
m=video 5006 RTP/AVP 31 34
a=rtpmap:31 H261/8000
a=rtpmap:34 H263/9000/1
```

### A Call Originating in SIP

This section describes for the IWF Privacy for Business trunking feature works for a call originating in SIP that requires interworking to H.323.

When the Oracle Communications Session Border Controller receives a SIP INVITE with a Privacy header that has the value ID, it sets the presentation indicator to restricted in the corresponding H.323 SETUP message. If the H.323 session agent is configured with the allowCPN option, the Oracle Communications Session Border Controller sends the display IE and the calling party number to the H.323 session agent. If that option is not set in the H.323 session agent, then the Oracle Communications Session Border Controller reverts to its default behavior, which is to not send the display IE and to hide the calling party number.

### Sample INVITE from a SIP Endpoint to the Oracle Communications Session Border Controller

```
May 5 14:41:54.513 On [0:0]192.168.200.68:5060 received from
192.168.200.6:5060
INVITE sip:800@192.168.200.68:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.200.6:5060
From: sipp <sip:sipp@192.168.200.6:5060>;tag=1
To: sut <sip:800@192.168.200.68:5060>
Call-ID: 1.3068.192.168.200.6@sipp.call.id
Cseq: 1 INVITE
Contact: sip:sipp@192.168.200.6:5060
Privacy: id
P-Asserted-Identity: sipp <sip:1234@192.168.200.6:5060>
Max-Forwards: 70
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 136
v=0
o=user1 53655765 2353687637 IN IP4 127.0.0.1
s=-
t=0 0
c=IN IP4 127.0.0.1
m=audio 10000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

### Sample SETUP from the Oracle Communications Session Border Controller to the H.323 Endpoint

```
Q.931
Protocol discriminator: Q.931
Call reference value length: 2
Call reference flag: Message sent from originating side
Call reference value: 44B0
Message type: SETUP (0x05)
Bearer capability
  Information element: Bearer capability
  Length: 3
  ...1 0000 = Information transfer capability: 3.1 kHz audio (0x10)
  .00. .... = Coding standard: ITU-T standardized coding (0x00)
  1... .... = Extension indicator: last octet
  ...1 0000 = Information transfer rate: 64 kbit/s (0x10)
  .00. .... = Transfer mode: Circuit mode (0x00)
  1... .... = Extension indicator: last octet
```



```

...0 0011 = User information layer 1 protocol: Recommendation G.711 A-
law (0x03)
1... .. = Extension indicator: last octet
  Display 'sipp'
    Information element: Display
    Length: 4
    Display information: sipp
Calling party number: '1234'
  Information element: Calling party number
Length: 6
  .... 0001 = Numbering plan: E.164 ISDN/telephony numbering (0x01)
  .010 .... = Number type: National number (0x02)
  0... .. = Extension indicator: information continues through the
next octet
  .... ..00 = Screening indicator: User-provided, not screened (0x00)
  .01. .... = Presentation indicator: Presentation restricted (0x01)
  1... .. = Extension indicator: last octet
    Calling party number digits: 1234
    E.164 Calling party number digits: 1234
Called party number: '800'
  Information element: Called party number
  Length: 4
.... 0001 = Numbering plan: E.164 ISDN/telephony numbering (0x01)
  .010 .... = Number type: National number (0x02)
  1... .. = Extension indicator: last octet
    Called party number digits: 800
    E.164 Called party number digits: 800
User-user
  Information element: User-user
  Length: 159
  Protocol discriminator: X.208 and X.209 coded user information

```

## allowCPN Configuration

You can set both of these options in the same H.323 session agent.

To set the allowCPN option for an H.323 session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# session-agent
```

4. Use the ACLI **select** command so that you can work with the session agent configuration to which you want to add this option.

```
ACMEPACKET(session-agent)# select
```

5. **options**—Set the options parameter by typing **options**, a Space, the option name **allowCPN** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(session-agent)# options +allowCPN
```

If you type **options allowCPN** (without the plus sign), you will overwrite any previously configured options. In order to append the new option to the session-agent's options list, you must prepend the new option with a plus sign as shown in the previous example.

6. Save and activate your configuration.

To set the NoPAssertedId option for an H.323 session agent:

7. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

8. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

9. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # session-agent
```

10. Use the ACLI **select** command so that you can work with the session agent configuration to which you want to add this option.

```
ACMEPACKET (session-agent) # select
```

11. **options**—Set the options parameter by typing **options**, a Space, the option name **NoPAssertedId** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (session-agent) # options +NoPAssertedId
```

**If you type options NoPAssertedId (without the plus sign), you will overwrite any previously configured options. In order to append the new option to the session-agent's options list, you must prepend the new option with a plus sign as shown in the previous example.**

12. Save and activate your configuration.

## Trunk Group URIs

---

The Oracle Communications Session Border Controller's trunk group URI feature, applicable for SIP and IWF signaling services, enables the capabilities related to trunk groups that are described in this section. This implementation follows the IPTEL draft Representing Trunk Groups in Tel/SIP Uniform Resource Identifiers (URIs) (draft-ietf-iptel-trunk-group-06.txt), and also supports more customized approaches.

- For a typical access call flow scenario, when the calling party's call arrives at the Oracle Communications Session Border Controller, the Oracle Communications Session Border Controller formulates a SIP INVITE message that it sends to a softswitch. The Oracle Communications Session Border Controller now supports a new URI contact parameter in the SIP request message so that service providers need to be able to:
  - Determine from where the Oracle Communications Session Border Controller received the call
  - Signal information about the originating gateway from a Oracle Communications Session Border Controller to a softswitch (e.g., an incoming trunk group or a SIP gateway to a Oracle Communications Session Border Controller)
- This feature supports the signaling of routing information to the Oracle Communications Session Border Controller from network routing elements like softswitches. This information tells the Oracle Communications Session Border Controller what egress route (or outgoing trunk groups) it should choose for terminating next hops/gateways. For this purpose, new SIP URI parameters in the Request-URI are defined. Additional URI parameters include the network context to identify the network in which the originating or terminating gateway resides.
- Especially important for large business applications, this feature can free Oracle Communications Session Border Controller resources by reducing the number of local policy, session agent, and session agent group configurations. By enabling the trunk group URI feature, the Oracle Communications Session Border Controller instead uses a routing scheme based on signaled SIP URI information.

## Terminology

The following IPTEL terms are used in the descriptions of and instructions for how to configure this feature:

- **Trunk**—In a network, a communication path connecting two switching systems used in the establishment of an end-to-end connection; in selected applications, it may have both its terminations in the same switching system

- Trunk group—A set of trunks, traffic engineered as a unit, for the establishment of connections within or between switching systems in which all of the paths are interchangeable except where sub-grouped
- Trunk group name—Provides a unique identifier of the trunk group; referred to as tgrp
- Trunk group context—Imposes a namespace by specifying a domain where the trunk groups are; also referred to simply as context

## Trunk Group URI Parameters

Trunk group URI parameters identify originating and terminating trunk group information in SIP requests.

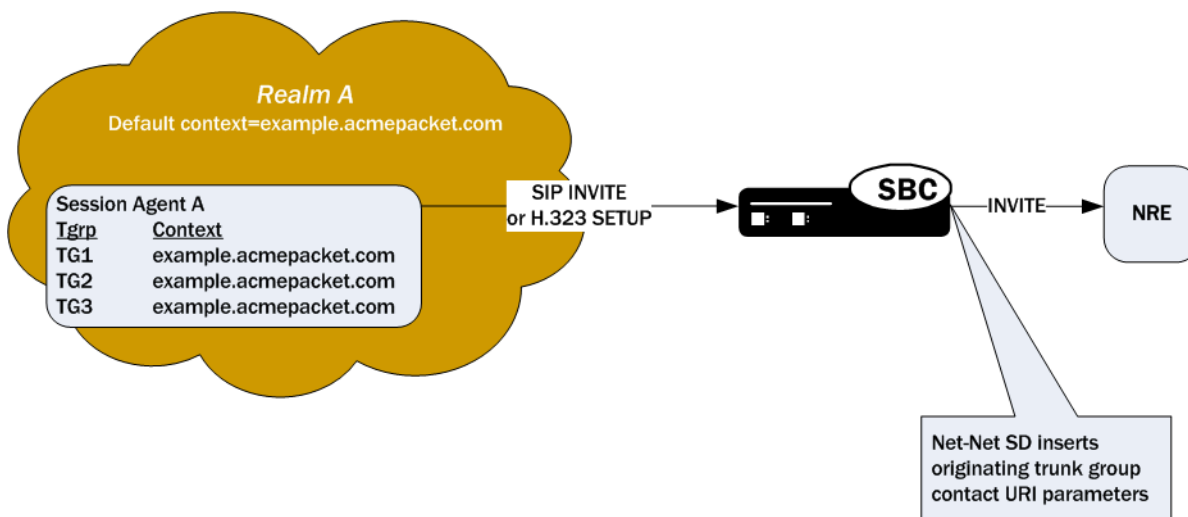
In the absence of official SIP standards for transporting trunk groups between signaling elements, the Oracle Communications Session Border Controller allows you to define URI parameters for use with originating and terminating trunk group URIs.

### Originating Trunk Group URI Parameters and Formats

You can configure session agents and session agents groups on the Oracle Communications Session Border Controller to insert trunk group URI parameters in the SIP contact header. When SIP gateways comply with the IPTEL draft, they include the originating URI parameter in the SIP contact header. For those SIP and H.323 gateways that are not compliant, the Oracle Communications Session Border Controller inserts SIP trunk group URI parameters on the gateway's behalf.

When there are no applicable session agent or session agent group configurations, the Oracle Communications Session Border Controller uses the source IP address of the endpoint or gateway as the trunk group name (tgrp) parameter in the originating trunk group URI.

The following diagram shows a scenario where the Oracle Communications Session Border Controller inserts originating trunk group URI parameters.



There are two available formats for the originating trunk group URIs:

1. In compliance with the IPTEL draft, the first format has two parameters: tgrp (identifier of the specific trunk group) and trunk-context (defines the network domain of the trunk group). These appear in the following formats:
  - tgrp="trunk group name"
  - trunk-context="network domain"

The URI BNF for would appear as it does in the example directly below, where the tgrp is tg55 and the trunk-context is trunk-context is telco.example.com:

```
tel:+15555551212;tgrp=tg55;trunk-context=telco.example.com
```

2. The second format is customized specifically for access URIs and contains two provisioned parameters: tgrp (or tgrname) and context (or provstring). This appears as tgrp.context, where these definitions apply:

- tgrp (tgrname)—Provisioned trunk group name for the originating session agent; this value must have at least one alphabetical character, cannot contain a period (.), and can contain a hyphen (-) but not as the first or the last character
- context (provstring)—Name of the originating trunk group context; this value must have at least one alphabetical character in the top label

This format conforms to format for a hostname in the SIP URI as specified in RFC 3261, such that a trunk group identifier would appear as:

```
custsite2NY-00020.type2.voip.carrier.net
```

where the tgrp is custsite2NY-00020, and the context is type2.voip.carrier.net.

The BNF for an access URI conforms to the following:

```
SIP-URI = "sip:" [userinfo ] hostport uri-parameters [headers ]
```

```
uri-parameters = *( ";" uri-parameter )
```

```
uri-parameter = transport-param / user-param / method-param
```

```
/ ttl-param / maddr-param / lr-param / other-param
```

```
other-param = accessid / pname [ '=' pvalue ]
```

```
accessid = "access=" accessURI
```

```
accessURI = scheme tgrname [ "." provstring ]
```

```
scheme = "sip:" / token
```

```
tgrname = ALPHA / *(alphanumeric) ALPHA *(alphanumeric / "-") alphanumeric /  
alphanumeric *(alphanumeric / "-") ALPHA *(alphanumeric) # up to 23 characters
```

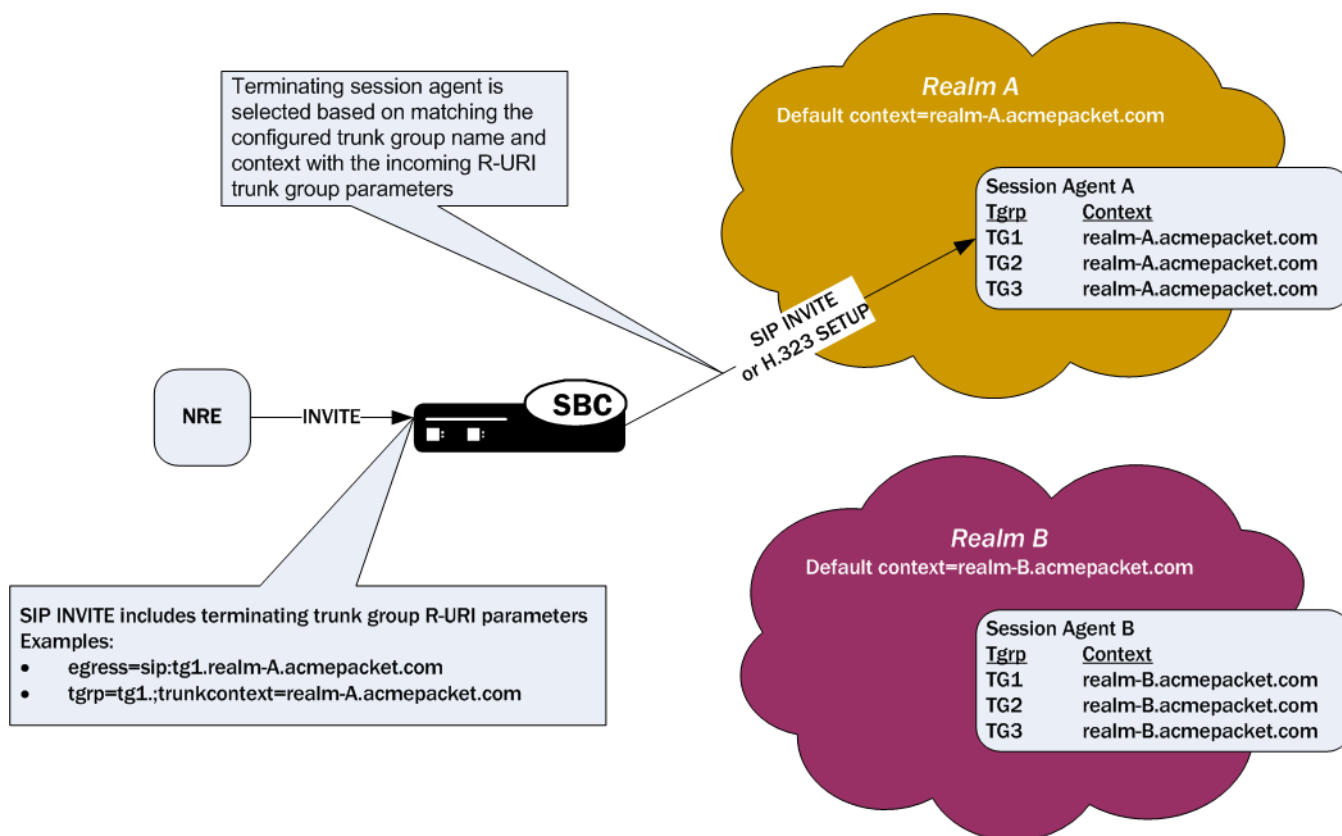
```
provstring = *(domain ".") toplabel # up to 24 characters
```

```
toplabel = ALPHA / ALPHA *( alphanumeric / "-") alphanumeric
```

```
domain = alphanumeric/ alphanumeric *( alphanumeric / "-") alphanumeric
```

### Terminating Trunk Group URI Parameters and Formats

Terminating trunk group URI parameters appear in the R-URI, and they can be included in by a network routing element to instruct the Oracle Communications Session Border Controller which egress trunk groups to use. By matching the trunk group URI parameter with configured session agents or session agent groups, the Oracle Communications Session Border Controller can locate the terminating gateway. The trunk group name can also be expressed as the IP address of the terminating gateway.



In the absence of official SIP standards for transporting trunk groups between signaling elements, the Oracle Communications Session Border Controller allows you to define the URI parameters used in terminating trunk groups.

There are two available formats for the terminating trunk group URIs:

1. In compliance with the IPTEL draft, the first format has two parameters: tgrp (which can be either a trunk group name or an IP address) and trunk-context (defines the network domain of the trunk group). These appear in the following formats:

- tgrp="trunk group name"
- trunk-context="network domain"

An example R-URI with terminating trunk group parameters appears as follows, where the tgrp is TG2-1 and the context is isp.example.net@egwy.isp.example.net:

```
INVITE sip:+15555551212;tgrp=TG2-1;trunk-context=isp.example.net@egwy.isp.example.net SIP/2.0
```

2. The second format is customized specifically for egress URIs and contains two provisioned parameters: tgrp (or tgrname) and context (or tgdomain). This appears as tgrp.context (or tgrname.tgdomain), where definitions apply:

- tgrp (tgrname)—Provisioned trunk group name for the originating session agent; this value must have at least one alphabetical character, cannot contain a period (.), and can contain a hyphen (-) but not as the first or the last character
- context (tgdomain)—Name of the terminating trunk group context; this value can be up to twenty-four characters

The use of multiple terminating trunk groups is not supported.

The BNF for a single, egress URI with trunk group information conforms to:

```
SIP-URI = "sip:" [userinfo ] hostport uri-parameters [headers ]
uri-parameters = *( ";" uri-parameter )
uri-parameter = transport-param / user-param / method-param
```

```
          / ttl-param / maddr-param / lr-param / other-param
other-param = egressid / pname [ '=' pvalue ]
egressid = "egress=" egressURI
egressURI = scheme tname [ "." tgdomain ]
scheme = "sip:" / token
tname = ALPHA / *(alphanum) ALPHA *(alphanum / "-") alphanum /
        alphanum *(alphanum / "-") ALPHA *(alphanum) # up to 23 characters
tgdomain = *(domain ".") toplabel # up to 24 characters
toplabel = ALPHA / ALPHA *( alphanum / "-") alphanum
domain = alphanum/ alphanum *( alphanum / "-") alphanum
```

For all trunk group URI support, you must set the appropriate parameters in the SIP manipulations configuration and in the session agent or session agent group configurations.

In the originating trunk group URI scenario, a call arrives at the Oracle Communications Session Border Controller from a configured session agent or session agent group. If this session agent or session agent group has the appropriate trunk group URI parameters and inbound manipulation rules configured, the Oracle Communications Session Border Controller then looks to the SIP manipulations configuration and add the trunk group URI information according to those rules. Those rules tell the Oracle Communications Session Border Controller where and how to insert the trunk group URI information, and the Oracle Communications Session Border Controller forwards the call.

In the terminating trunk group scenario, a call arrives at the Oracle Communications Session Border Controller from, for instance, a call agent. This call contains information about what trunk group to use. If the information matches a session agent or session agent group that has outbound manipulation rules configured, the Oracle Communications Session Border Controller will then look up the SIP manipulations configuration and strip information according to those rules. Those rules tell the Oracle Communications Session Border Controller where and how to remove the information, and the Oracle Communications Session Border Controller forwards the call.

### SIP Header and Parameter Manipulation

SIP header and parameter manipulation is its own configuration where you can set up rules for the addition, removal, and modification of a SIP header or the elements of a SIP header. For example, you can set up the configuration to add a URI parameter to the URI in a SIP header or replace an FQDN with in IP address. For trunk group URI support, this configuration tells the Oracle Communications Session Border Controller where and how to manipulate the SIP message to use originating (access) and terminating (egress) trunk group URI parameters.

These manipulations can be applied at the realm or at the session agent level.

### Trunk Group Routing

You can configure SIP interfaces (using the ACLI **term-tgrp-mode** parameter) to perform routing based on the trunk group information received in SIP requests. There are three options: none, IPTEL, and egress URI.

- If you leave this parameter set to none (its default), the Oracle Communications Session Border Controller will not look for or route based on terminating trunk group URI parameters
- When you set this parameter to either **iptel** or **egress-uri** and the incoming request has the trunk group parameter of this type (IPTEL or egress URI), the Oracle Communications Session Border Controller will select the egress next hop by matching the “tgrp” and trunk context with a configured session agent or session agent group.

If the received terminating trunk group URI parameters include an IP address, the egress next hop is the IP address specified. The Oracle Communications Session Border Controller determines the egress realm by matching the trunk context it receives with the trunk context you configure for the realm.

- If the incoming request does not have trunk group parameters or it does not have trunk group parameters of the type that you configure, the Oracle Communications Session Border Controller uses provisioned procedures and/or local policy for egress call routing.

The Oracle Communications Session Border Controller returns errors in these cases:

- If the terminating trunk group URI parameters do not identify a local Oracle Communications Session Border Controller session agent or session agent group, then the Oracle Communications Session Border Controller returns a SIP final response of 488 Not Acceptable Here.
- If the Oracle Communications Session Border Controller receives a SIP INVITE with terminating trunk group URI parameters that do not match the specified syntax, the Oracle Communications Session Border Controller returns a 400 final response with the reason phrase Bad Egress=Parameters.

### Trunk Group URIs and SIP Registration Caching

For calls where SIP registration caching is used, you will need to set certain parameters that enable the Oracle Communications Session Border Controller to preserve trunk group URI parameters on the outgoing side.

- For SIP-H.323 calls requiring IWF, you set the `preserve-user-info-sa` **option** in the session agent configuration.

## Trunk Group URI Configuration

Before you configure your Oracle Communications Session Border Controller to support trunk group URIs, you need to determine:

- How you want to manipulate SIP headers (entered in the SIP header manipulations configuration)
- For terminating trunk group routing, the trunk group mode you want to use (none, IPTEL, or egress URI); this decides routing based on trunk group information
- The trunk group name and context to use entered in a session agent or session agent group configuration
- Whether you are using originating or terminating trunk group URIs (entered in the session agent configuration)
- The trunk group context for use in a realm configuration, in case the trunk group name in the session agent or session agent group does not have a context

### Configuring SIP Manipulations

When you configure the SIP header manipulations to support trunk group URIs, take note of:

- The name of the configuration, so that you can use it when you apply the manipulations in a session agent for the inbound or outbound manipulations
- The **new-value** parameter, which specifies the trunk group and trunk group context that you want to manipulate; the possible values that apply to trunk group URI configurations are `$TRUNK_GROUP` and `$TRUNK_GROUP_CONTEXT`

### Setting the Trunk Group URI Mode for Routing

To set the mode for routing for terminating trunk group URIs:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure)# session-router
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router)# sip-interface
ACMEPACKET (sip-interface)#
```

4. **term-tgrp-mode**—Set the mode that you want to use for routing for terminating trunk group URIs. The default value is **none**. Valid values are:

- **none**—Disables routing based on trunk groups
- **iptel**—Uses trunk group URI routing based on the IPTEL formats
- **egress-uri**—Uses trunk group URI routing based on the egress URI format

### Configuring a Session Agent for Trunk Group URIs

In a session agent, you can configure the outbound or inbound SIP header manipulation rules to use, as well as a list of trunk group names and contexts. For the trunk group names and contexts, you can use either the IPTEL or the custom format.

To configure a session agent for trunk group URIs:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET (session-router) # session-agent  
ACMEPACKET (session-agent) #
```

4. **out-manipulationid**—Enter the name of the SIP header manipulations configuration that you want to apply to the traffic exiting the Oracle Communications Session Border Controller via this session agent. There is no default.
5. **in-manipulationid**—Enter the name of the SIP header manipulations configuration that you want to apply to the traffic entering the Oracle Communications Session Border Controller via this session agent. There is no default.
6. **trunk-group**—In either IPTEL or custom format, enter the trunk group names and trunk group contexts to match. If you do not set the trunk group context, then the Oracle Communications Session Border Controller will use the one you set in the realm for this session agent.

Your ACLI entries for this list must one of these formats: `tgrp:context` or `tgrp.context`.

To make multiple entries, surround your entries in parentheses and separate them from each other with spaces. For example:

```
ACMEPACKET (session-agent) # trunk-group (tgrp1:context1  
tgrp2:context2)
```

7. **options**—If you want to configure trunk group URIs for SIP-H.323 calls that use the IWF and you are using SIP registration caching, you might need to add the `preserve-user-info-sa` to your list of session agent options.

If you are adding this option to a new session agent, you can just type **options**, a Space, and **preserve-user-info-sa**.

**If are adding this to an existing session agent, you must type a plus (+) sign before the option or you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign: options +preserve-user-info-sa.**

### Configuring a Session Agent Group for Trunk Group URIs

In a session agent group, you can configure the outbound or inbound SIP header manipulation rules to use, as well as a list of trunk group names and contexts. For the trunk group names and contexts, you can use either the IPTEL or the custom format.

To configure a session agent group for trunk group URIs:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-group** and press Enter.



```
ACMEPACKET(session-router) # session-group
ACMEPACKET(session-agent-group) #
```

4. **trunk-group**—In either IPTEL or custom format, enter the trunk group names and trunk group contexts to match. If you do not set the trunk group context, then the Oracle Communications Session Border Controller will use the one you set in the realm for this session agent group.

Your ACLI entries for this list must take one of these formats: `tgrp:context` or `tgrp.context`.

To make multiple entries, surround your entries in parentheses and separate them from each other with spaces. For example:

```
ACMEPACKET(session-agent-group) # trunk-group (tgrp1:context1 tgrp2:context2)
```

## Setting a Trunk Group Context in a Realm

You can set trunk group contexts at the realm level, which will be used by all session agents and session agent groups if there is no context specified in their configurations.

The realm trunk group URI context accommodates the IPTEL and the custom format.

To configure a trunk group context for a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the session-related configurations.

```
ACMEPACKET(configure) # media-manager
```

3. Type **realm-config** and press Enter.

```
ACMEPACKET(media-manager) # realm-config
ACMEPACKET(realm-config) #
```

4. **trunk-context**—Enter the trunk group context to use for this realm. There is no default.

## Using this Feature with SIP Interface Registration Caching

If you are using the trunk group URIs feature with SIP interface that has registration caching enabled, then you need to configure the `preserve-user-info` option for that SIP interface.

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET(configure) # session-router
```

3. Type **session-group** and press Enter.

```
ACMEPACKET(session-router) # sip-interface
ACMEPACKET(sip-interface) #
```

4. **options**—Add support for trunk group URIs with SIP interface that uses registration caching.

If you are adding this option to a new SIP interface, you can just type **options**, a Space, and **preserve-user-info**.

**If are adding this to an existing SIP interface, you must type a plus (+) sign before the option or you will remove any previously configured options. In order to append the new option to the options list, you must prepend the new option with a plus sign: `options +preserve-user-info`.**

## Example 1 Adding Originating Trunk Group Parameters in IPTEL Format

This ACLI sample shows you how the ACLI SIP manipulations might appear in a case where you want to add originating trunk parameters in IPTEL format.

```

sip-manipulation
  name                               add_ipstel
  header-rule
    name                               contact
    action                             manipulate
    match-value
    msg-type                           any
    element-rule
      name                               tgrp
      type                               uri-user-param
      action                             add
      match-val-type                    any
      match-value
      new-value                          $TRUNK_GROUP
element-rule
      name                               trunk-context
      type                               uri-user-param
      action                             add
      match-val-type                    any
      match-value
      new-value                          $TRUNK_GROUP_CONTEXT

```

**Example 1 Adding Originating Trunk Group Parameters in Custom Format**

This ACLI sample shows you how the ACLI SIP manipulations might appear in a case where you want to add originating trunk parameters in custom format.

```

sip-manipulation
  name                               add_att
  header-rule
    name                               contact
    action                             manipulate
    match-value
    msg-type                           any
    element-rule
      name                               egressURI
      type                               uri-param
      action                             add
      match-val-type                    any
      match-value
      new-value                          "sip:"+$TRUNK_GROUP
+"."+$TRUNK_GROUP_CONTEXT

```

**Example 2 Removing IPTEL Trunk Group Names**

This ACLI sample shows you how the ACLI SIP manipulations might appear in a case where you want to remove IPTEL trunk groups names.

```

sip-manipulation
  name                               strip_ipstel
  header-rule
    name                               request-uri
    action                             manipulate
    match-value
    msg-type                           any
    element-rule
      name                               tgrp
      type                               uri-user-param
      action                             delete-element
      match-val-type                    any
      match-value
      new-value
element-rule
      name                               trunk-context
      type                               uri-user-param

```

```

        action                delete-element
        match-val-type       any
        match-value
        new-value

```

### Example 3 Removing Custom Trunk Group Names

This ACLI sample shows you how the ACLI SIP manipulations might appear in a case where you want to remove custom trunk groups names.

```

sip-manipulation
  name                strip_egress
  header-rule
    name              request-uri
    action            manipulate
    match-value
    msg-type          any
  element-rule
    name              egressURI
    type              uri-param
    action            delete-element
    match-val-type   any
    match-value
    new-value

```

### Configuring SIP Manipulations

When you configure the SIP header manipulations to support trunk group URIs, take note of:

- The name of the configuration, so that you can use it when you apply the manipulations in a session agent for the inbound or outbound manipulations
- The **new-value** parameter, which specifies the trunk group and trunk group context that you want to manipulate; the possible values that apply to trunk group URI configurations are \$TRUNK\_GROUP and \$TRUNK\_GROUP\_CONTEXT

## IWF COLP COLR Support

When you enable the connected line identity presentation (COLP) and connected line identity restriction (COLR) feature for calls being translated between SIP and H.323, the Oracle Communications Session Border Controller converts the H.323 Connected Number Information element (IE) to the SIP P-Asserted-Identity (PAI) header and vice versa.

When there is no Q.931 Connected Number IE, the Oracle Communications Session Border Controller converts the H.225 Connected Address alias (either E.164 or Public Party Number).

This section describes show the IWF COLP/COLR feature works for IWF calls that originate in SIP and are translated to H.323, and for calls that originate in H.323 and are translated to SIP.

### SIP to H.323 Calls

For this type of call, the Oracle Communications Session Border Controller checks the Connect that it receives for a Q.931 Connected Number IE. If it does not find one, then it continues by checking for H.225 Connected Address alias (either E.164 or Public Party Number). Then, it takes one of the following courses of action depending on circumstances:

- If it finds the Q.931 Connected Number IE, the Oracle Communications Session Border Controller extracts the screening indicator and the presentation indicator.
- If there is no Q.931 Connected Number IE, the Oracle Communications Session Border Controller extracts the screening indicator and the presentation indicator from the H.225 Connect-UUIE of the Connect message.

With these pieces of information in place, the Oracle Communications Session Border Controller performs the conversion from H.323 Connected Number IE to SIP P-Asserted-Identity (PAI) header if and only if the screening indicator is either one of the following:

- Network provided
- User-provided, verified and passed

Then the Oracle Communications Session Border Controller adds a SIP PAI header (with URI value) to the 200 OK message that it sends in the SIP call leg. The user part of the URI is set to the value of the Q.931 Connected Number IE's numberDigits field, or to dialDigits value from the Connected Address alias. When the number type is a national number, the Oracle Communications Session Border Controller adds a plus sign (+) and the IWF country code (that you configure) to the beginning of the user part. If the number type is an international number, the Oracle Communications Session Border Controller only adds a plus sign (+). And when the Connected Number is empty, the Oracle Communications Session Border Controller sets the user part of the PAI header URI to anonymous. When the value in the presentation indicator is Presentation restricted, the Oracle Communications Session Border Controller adds the SIP Privacy header (with the value id) to the 200 OK.

In cases when it does not find a screening indicator, the Oracle Communications Session Border Controller will not perform the conversion from the H.323 Connected Number IE to the SIP P-Asserted-Identity (PAI) header.

### H.323 to SIP Calls

For this type of call, the Oracle Communications Session Border Controller checks the 200 OK message for a SIP PAI header and a SIP Privacy header. Before it sends a Connect message on the H.323 call leg, the Oracle Communications Session Border Controller generates a Connected Number. It uses the Connected Number to insert a Q.931 Connected Number IE and an H.225 Connected Address alias (type E.164) into the Connect message. The Connected Number is generated in this way:

- If the
  - SIP PAI header is not found, or
  - User part of its URI value is unknown or anonymous, or
  - User part of its URI does not follow the H.225 NumberDigits syntax,

then the Connect Number that the Oracle Communications Session Border Controller generates is a Q.931 Connected Number IE that has no digits and a number type of unknown. In this case, the Oracle Communications Session Border Controller will not insert an H.225 Connected Address alias into the Connect message.

The presentation indicator is set to Number not available due to interworking, and the screening indicator to Network provided. The H.225 NumberDigits's syntax requires that it be between 1 and 128 characters, and only contain these characters: 0 through 9, the pound sign (#), the asterisk (\*), and the comma (,).

- In all other cases, the Oracle Communications Session Border Controller uses the user part of the URI as the digits for the Connected Number after it performs the following:
  - Strips the plus sign in front of the number, if there is one
  - Strips the IWF country code at the beginning of the number, if there is one

Then the Oracle Communications Session Border Controller inserts the Connected Number into the Connect message as the Q.931 Connected Number IE and an H.225 Connected Address alias (type E.164).

If the IWF country code is found in the PAI, the Oracle Communications Session Border Controller sets the type of Q.931 Connected Number IE to National Number. Otherwise, the Oracle Communications Session Border Controller sets it to international. The screening indicator is set to Network provided, and the presentation indicator is set to Presentation Restricted if the Oracle Communications Session Border Controller finds a SIP Privacy header with a value of id, or Presentation Allowed is there is not SIP Privacy header.

### IWF COLP COLR Configuration

You configure IWF COLP/COLR support in the IWF configuration by setting two options:

- **colp-colr-iwf**—Setting this option enables support for IWF COLP/COLR
- **colp-colr-country-code**—Must be set if you configure the **colp-colr-iwf** option to recognize or build a national number; the value you enter here:
  - Must be a string of digits from 0 to 9
  - Cannot exceed 32 digits
  - Cannot contain any non-numeric characters; while it allows you to enter them, the system ignores any non-digits characters and so the feature might not work as needed

To enable IWF COLP/COLR support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the signaling-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **iwf-config** and press Enter. The system prompt will change to let you know that you can configure individual

```
ACMEPACKET(session-router)# iwf-config
```

4. **options**—Set the options parameter by typing **options**, a Space, the option names with a plus sign in front, and then press Enter.

Your entry for the **colp-colr-country-code** option require that you type in the entire option name, an equal sign (=), and then the country code value.

To enter both options at once, separate the two with one command and enclose your entire entry in quotation marks ( ); see the following example for command-line syntax.

```
ACMEPACKET(iwf-config)# options +colp-colr-iwf,colp-colr-country-code=1
```

**If you type this enter without the plus sign, you will overwrite any previously configured options. In order to append options to the IWF configuration's options list, you must prepend the new options with a plus sign as shown in the previous example.**

5. Save and activate your configuration.

## Options for Calls that Require the IWF

You can configure several specific behaviors by configuring options for calls that require the IWF, and set them for the H.323 side of the call. These options are listed and defined in the table below. Options can be configured either globally for the H.323 configuration, individually for an H.323 interface, or for H.323 session agents.

### Global Configuration for H.323

To configure options globally for H.323:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
```

From this point, you can configure H.323 parameters. To view see all H.323 parameters, enter a ? at the system prompt.

4. Type **options**, a space, and the name of the option you want to use. In this example, the MapG729 will map H.245 G.729 to SDP G.729 with Annex B and vice versa.

```
ACMEPACKET (h323) # options MapG729
```

### Individual Configuration for H.323

To configure options per individual H.323 interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET (session-router) # h323
```

4. Type **h323-stacks** and press Enter. The system prompt changes again to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (h323) # h323-stacks
ACMEPACKET (h323-stack) #
```

From this point, you can configure H.323 interface parameters. To view see all H.323 interface parameters, enter a **?** at the system prompt.

5. Type **options**, a space, and the name of the option you want to use. In this example, the MapG729 will map H.245 G.729 to SDP G.729 with Annex B and vice versa.

```
ACMEPACKET (h323-stack) # options
```

### Configuring H.323 SA Options

To configure options for H.323 session agents:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET (session-router) # session-agent
```

From this point, you can configure session agent parameters. To view see all session agent parameters, enter a **?** at the system prompt.

4. Type **options**, a space, and the name of the option you want to use. In this example, the MapG729 will map H.245 G.729 to SDP G.729 with Annex B and vice versa.

```
ACMEPACKET (h323-stack) # options MapG729
```

### H.323 SA Options

Options	Description
MapG729	Oracle Communications Session Border Controller maps H.245 G.729 to SDP G.729 with Annex B and vice versa. Applicable only to calls that require the IWF.
ColonG729	Oracle Communications Session Border Controller uses the : (colon) instead of the = (equal sign) in the media attribute line a=fmtp:18 annex=yes/no when mapping H.245 G.729 or SDP G.729 with Annex B. Applicable only to calls that require the IWF.

Options	Description
IwfLRQ	Oracle Communications Session Border Controller sends an INVITE (with no SDP) to a redirect server in response to an incoming LRQ received on an H.323 interface. If a 3xx message with a redirected contact header is returned, the Oracle Communications Session Border Controller will send an LCF in response to the LRQ. Otherwise, it will send an LRJ.
NoG729AnnexB	SDP received by the IWF with H.729 and no FMTP will be mapped to G.729 on the H.323 side of the call. Can also be set in the session agent options parameter.
sameT38Port	Oracle Communications Session Border Controller's H.323 process does not allocate separate ports for audio and T.38. Oracle Communications Session Border Controller will send the same audio port in the OLCAck that it sees in a request mode for T.38 and a new OLC for T.38.
pvtStats	Oracle Communications Session Border Controller includes program value tree (PVT) statistics in the show h323d display that are a sum of the PVT statistics for all H.323 interfaces. Used for debugging purposes.
acceptAl	Oracle Communications Session Border Controller accepts all the codecs received in the SIP 200OK and builds the TCS accordingly.

## Suppress SIP Reliable Response Support for IWF

For IWF-originated calls, the Oracle Communications Session Border Controller now allows you to configure the suppression of the SIP 100rel option tag on a per-H.323 interface (stack) basis.

When a call originates on the H.323 side for a call that requires interworking between H.323 and SIP, the Oracle Communications Session Border Controller inserts the 100rel option tag in the Supported header of the outgoing SIP INVITE. Although this behavior is required for RFC 3262 conformance, and is ignored by endpoints that do not support this RFC, suppressing the reliable response can alleviate processing burdens and avoid the possibility that an endpoint could mishandle the response.

In addition, enabling this feature suppresses the same 100rel options tag in the Required header for outgoing IWF responses for which an incoming SIP INVITE had that same tag in its Supported header. If an incoming INVITE requires reliable provisional responses and the SIP feature configuration is set to accept the 100rel, the Oracle Communications Session Border Controller then includes the 100rel option tag in the outgoing response's Required header. When the SIP feature is not so configured, the Oracle Communications Session Border Controller rejects the INVITE with a 420 Bad Extension response.

Without this option, you can suppress the reliable response on a global basis or per SIP next-hop by using the SIP feature configuration. However, using this feature allows a finer degree of granularity by making the functionality only applicable to IWF calls that originate in H.323.

### suppress100rel Configuration

To suppress the SIP 100rel option tag:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **h323** and press Enter.

```
ACMEPACKET(session-router)# h323
ACMEPACKET(h323)#
```

4. Type **h323-stacks** and press Enter.

```
ACMEPACKET(h323) # h323-stacks
ACMEPACKET(h323-stack) #
```

If you are adding support for this feature to a pre-existing H.323 interface (stack), then you must select (using the ACLI **select** command) the configuration that you want to edit.

5. **options**—Set the options parameter by typing **options**, a Space, the option name **suppress100rel** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(h323-stack) # options +suppress100rel
```

**If you type options** and then the option value for either of these entries **without the plus sign, you will overwrite any previously configured options. In order to append the new option to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.**

## IWF Codec Negotiation H.323 Slow Start to SIP

---

For instances when the Oracle Communications Session Border Controller is translating a call initiated in H.323 slow start to SIP, you can enable a setting in the IWF configuration that prevents the sending an SDP offer in the SIP INVITE. Instead, the Oracle Communications Session Border Controller expects to see an SDP offer from the SIP endpoint in a provisional or reliable/provisional 200 OK, and then sends an answer in an ACK or PRACK.

With this parameter disabled (default), the Oracle Communications Session Border Controller populates the SIP INVITE with SDP based on the media profiles applied to the ingress H.323 session agent or the IWF configuration.

## IWF Codec Negotiation Configuration

To prevent the Oracle Communications Session Border Controller from sending an SDP offer in the SIP INVITE for a call being translated between H.323 slow start and SIP:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

3. Type **iwf-config** and press Enter.

```
ACMEPACKET(session-router) # iwf-config
ACMEPACKET(iwf-config) #
```

4. **slow-start-no-sdp-in-invite**—Enable this parameter if you want to prevent the Oracle Communications Session Border Controller from sending an SDP offer in the SIP INVITE for an IWF call initiated in H.323 slow start (being translated to SIP). The default is disabled. Valid values are:

- enabled | disabled

5. Save and activate your configuration.

## IWF H.245 Signaling Support for G.726

---

In addition to providing G.726 support for pure SIP and pure H.323 calls, the Oracle Communications Session Border Controller supports the G.726 payload type for H.245 and calls that require interworking (IWF) between SIP and H.323.

For IWF calls using ITU-T G.726 as the audio codec, the SIP call leg requires G.726 in the SDP. The H.323 side of the call signals G.726 (in the H.245 openLogicalChannel and TerminalCapabilitySet messages) by including a GenericCapability defining G.726 as the codec. In the GenericCapability, the capabilityIdentifier and maxBitRate



parameters identify G.726. While a capabilityIdentifier with 0.0.7.726.1.0 designates G.726, the maxBitRate designate the data transmission rate.

Codec	Max Bit Rate	Data Rate
G726-16	160	16 kbit/s
G726-24	240	24 kbit/s
G726-32	320	32 kbit/s
G726-40	400	40 kbit/s

To support G.726 for IWF calls, the Oracle Communications Session Border Controller converts the G726-X value in the SDP of SIP messages to a GenericCapability structure in H.323/H.245 messages, and the conversion works the same way in reverse.

## H.245 and G.726 Configuration

To enable this feature, you do need to set up media profile configurations appropriately. Media profiles now allow you to set the configuration to any of the four G.726 encodings (as defined by ITU G726 Annex B and RFC 3551). You must create one media profile for each of the four different supported data rates. In addition, you are also required to set a genericAudioCapability media profile.

### Media Profile for H.245 and G.726 Configuration

To set a media profile for H.245 and IWF G.726 support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **media-profile** and press Enter.

```
ACMEPACKET (session-router) # media-profile
ACMEPACKET (media-profile) #
```

4. **name**—Set the name of the media profile to G726-16. Values to support this feature are: G726-16, G726-24, G726-32, and G726-40.
5. **media-type**—Set the media type to use for this media profile; for generic video, set this parameter to **audio**. Valid values are:
  - audio | video | application | data
6. **payload-type**—Set the payload type to use for the generic video media profile.
7. **transport**—Set the transport type to use for the generic video media profile. The default value is **RTP/AVP**. Valid values are:
  - UDP | RTP/AVP
8. Complete the rest of the media profile configuration as needed.
9. Save and activate your configuration.

The following is a sample of a media profile configuration for H.245/IWF G.726 support:

```
media-profile
  name                g726-40
  media-type          audio
  payload-type        105
  transport            RTP/AVP
  req-bandwidth       0
  frames-per-packet   0
```

```

parameters
average-rate-limit          0
sdp-rate-limit-headroom    0
sdp-bandwidth               disabled

```

## Media Profile Configuration for Generic Audio Support

To set a media profile for generic audio support:

1. In Superuser mode, type **configure terminal** and press Enter.

```

ACMEPACKET# configure terminal
ACMEPACKET (configure) #

```

2. Type **session-router** and press Enter.

```

ACMEPACKET (configure) # session-router

```

3. Type **media-profile** and press Enter.

```

ACMEPACKET (session-router) # media-profile
ACMEPACKET (media-profile) #

```

4. **name**—Set the name of the generic audio media profile to genericAudioCapability. There is no default for this parameter.
5. **media-type**—Set the media type to use for this media profile; for generic video, set this parameter to **audio**. Valid values are:
  - audio | video | application | data
6. **payload-type**—Set the payload type to use for the generic audio media profile.
7. **transport**—Set the transport type to use for the generic audio media profile. The default value is **RTP/AVP**. Valid values are:
  - UDP | RTP/AVP
8. Complete the rest of the media profile configuration as needed.
9. Save and activate your configuration.

The following is a sample of a generic audio media profile configuration:

```

media-profile
  name          genericAudioCapability
  media-type    audio
  payload-type  104
  transport     RTP/AVP
  req-bandwidth 0
  frames-per-packet 0
  parameters
  average-rate-limit 0
  sdp-rate-limit-headroom 0
  sdp-bandwidth disabled

```

## Flow Control Mapping for Interworking Function (IWF) Video

H.245 is a protocol for the transmission of call management and control signals in networks using H.323 equipment. The H.245 specification is used in audio, video, and data transmissions, as well as in voice over IP (VoIP). H.245 messages are sent over special channels called H.245 control channels.

H.245 signaling is used to manage and control call setup and connection. Functions of H.245 include determining which endpoint is to be the master and which is to be the slave during the call, opening and closing of multiplexed data-transfer paths between the endpoints, establishing an upper limit to the data transfer speed on each logical channel, information exchanges between endpoints concerning the types of data each endpoint can send and receive, requests by the receiving endpoint for changes in the mode of the data sent by the transmitting endpoint, and requests by either endpoint to end the call.

In the H.245 standard, the FlowControlCommand message is used to specify the upper limit of bit rate of either a single logical channel or the whole multiplex. The following is an excerpt from the H.245 standard.

Command Message: Flow Control (from H.245 standard)

```

=====
FlowControlCommand ::= SEQUENCE
{
    scope CHOICE
    {
        logicalChannelNumber LogicalChannelNumber,
        resourceID INTEGER (0..65535),
        wholeMultiplex NULL
    },
    restriction CHOICE
    {
        maximumBitRate INTEGER (0..16777215), -- units 100 bit/s
        noRestriction NULL
    },
    ...
}
=====

```

A terminal may send this command to restrict the bit rate that the far-end terminal sends. A receiving terminal must comply with this command.

In an H.323 environment, the Oracle Communications Session Border Controller previously used the FlowControlCommand to map to SIP using either the Real-Time Control Protocol (RTCP) feedback function, or the SIP signaling path (for example, the INFO method).

The Oracle Communications Session Border Controller now supports the SIP counter part of the H.245 FlowControlCommand using the SIP signaling path with the INFO method. The Oracle Communications Session Border Controller sends the SIP INFO message with "change\_bitrate" rate parameter that has the value 100\* maxBitRate from the corresponding H.245 FlowControlCommand message. For example, in the following messages, the incoming H.323 message with the H.245 FlowControlCommand, is converted into the outgoing SIP INFO message with the message body.

Incoming H.323 Message with H.245 FlowControlCommand:

```

H.245
PDU Type: command (2)
  command: flowControlCommand (4)
    flowControlCommand
      scope: logicalChannelNumber (0)
        logicalChannelNumber: 102
      restriction: maximumBitRate (0)
        maximumBitRate: 4480

```

Outgoing SIP INFO Message:

```

Message Body
eXtensible Markup Language
  <?xml
    version="1.0"
    encoding="utf-8"
  >
  <media_control>
    <vc_primitive>
      <to_encoder>
        <change_bitrate>
          4480000
        </change_bitrate>
      </to_encoder>
    </vc_primitive>
  </media_control>

```

## Customized G.729 Support

The Oracle Communications Session Border Controller supports the use of custom G.729 encoding for calls that require interworking between SIP and H.323. If you use a proprietary G.729 encoding format in your network, then you might need to use this feature.

When you set the **acceptG729abFormat** option in the global H.323 configuration, the Oracle Communications Session Border Controller performs conversions like those in the following examples:

- For calls initiated in SIP, the Oracle Communications Session Border Controller can parse RTP map strings such as G.729a and G.729ab in the SDP, and then map them to H.245 data types.
  - G.729a becomes g729AnnexA.
  - G.729ab becomes g729AnnexAwAnnexB.
- For calls initiated in H.323, the Oracle Communications Session Border Controller can create non-standard RTP map strings such as G.729a and G.729ab from mapped H.245 data types.
  - g729 becomes G729.
  - g729AnnexA becomes G.729a.
  - g729AnnexAwAnnexB becomes G.729ab.

When you enable the **acceptG729abFormat** option, the Oracle Communications Session Border Controller performs customized G.729 mapping in the following instances.

- For calls initiated in SIP and translated to H.323, the Oracle Communications Session Border Controller:
  - Converts the SDP in an incoming SIP INVITE to a list of fastStart OpenLogicalChannel requests that are in turn included in the outgoing Setup message.
  - Converts the list of fastStart OpenLogicalChannelAck responses (which can be received in any message up to and including the Connect message) to SDP sent with a SIP response.
- For calls initiated in H.323 and translated to SIP, the Oracle Communications Session Border Controller:
  - Converts the list of fastStart OpenLogicalChannel requests to SDP in the outgoing SIP INVITE.
  - Converts SDP in a SIP response (such as a 200 OK) to the list of fastStart OpenLogicalChannelAck responses included with the callProceeding, Progress, Alerting, or Connect message. This depends on when the SDP is received on the SIP side.
- For all IWF calls regardless of initiating protocol, the Oracle Communications Session Border Controller:
  - Converts SDP on the SIP side to the terminalCapabilitySet message to be sent on the H.323 side.

Also note that when the format is G729, the Oracle Communications Session Border Controller maps it to g729wAnnexB if the a=fmtp:18 annexb=yes attribute is present. When the a=fmtp:18 annexb=no attribute is present, the Oracle Communications Session Border Controller maps G729 to g729. And with no a=fmtp:18 annexb=no attribute, the Oracle Communications Session Border Controller also maps G729 to g729 when this option is enabled.

The Oracle Communications Session Border Controller also maps G729 to g729 because pure G729 with static payload type 18 does not include an fmtp attribute where annexb=no.

## About Dynamic Payload Mapping

G.729a and G.729ab use dynamic payload types, but the Oracle Communications Session Border Controller does not propagate these dynamic payload types to corresponding dynamicRTPPayloadType (an optional field in OpenLogicalChannel requests) on the H.323 side.

For an IWF call initiated in H.323, the dynamic payload types for G.729a and G.729ab are retrieved from media profile configurations when the Oracle Communications Session Border Controller converts the list of fastStart OpenLogicalChannel requests to SDP sent on the SIP side. As a result, you must set up media profile configurations for G.729a and G.729ab for the feature to work properly. In these media profiles, the following parameters must be set as follows:

- **name**—For the G.729a profile, set the **name** to **G.729a**. For the **G.729ab** profile, set the **name** to **G.729ab**.

- **payload-type**—For each media profile (**G. 729a** and **G. 729ab**), DO NOT use payload type 18, which is the static payload type used for G729.

## Customized G.729 Configuration

This section shows you how to configure the **acceptG729abFormat** option in the global H.323 configuration.

To enable customized G.729 support for IWF calls:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **h323-config** and press Enter.

```
ACMEPACKET(session-router)# h323-config
ACMEPACKET(h323-config)#
```

If you are adding this feature to a pre-existing configuration, select the configuration to edit it.

4. **options**—Set the options parameter by typing options, a Space, the option name **acceptG729abFormat** with a plus sign in front of it. Then press Enter.

```
ACMEPACKET(h323-stack)# options +acceptG729abFormat
```

If you type **options** and then the option value for either of these entries without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

## SIP-H.323 IWF Support for H.264 and H.263+

Signaling protocol interworking between SIP and H.323 supports the H.264 and H.263+ video codecs.

### H.264 in H.323 (H.241)

This section describes the H.264 capabilities and media packetization in H.323. Capability exchange signaling looks like this:

```
openLogicalChannel . SEQUENCE [EMPTY -1] ...
forwardLogicalChannelNumber = 3 . INTEGER [EMPTY -1] (1..65535)
forwardLogicalChannelParameters . SEQUENCE [EMPTY -1] ...
.. dataType . CHOICE [EMPTY -1] ...
.. . . . videoData . CHOICE [EMPTY -1] ...
.. . . . genericVideoCapability . SEQUENCE [EMPTY -1] ...
.. . . . capabilityIdentifier . CHOICE [EMPTY -1] ...
.. . . . . standard = 7 {itu-t recommendation h 241 0 0 1}.OBJECT
IDENTIFIER [EMPTY-1]
.. . . . . maxBitRate = 4480 . INTEGER [EMPTY -1] (0..-1)
.. . . . . collapsing . SEQUENCE OF [EMPTY -1] SEQUENCE [EMPTY -1] ...
.. . . . . * . SEQUENCE [EMPTY -1] ...
.. . . . . . parameterIdentifier . CHOICE [EMPTY -1] ...
.. . . . . . . standard = 41 . INTEGER [EMPTY -1] (0..127)
.. . . . . . . parameterValue . CHOICE [EMPTY -1] ...
.. . . . . . . . booleanArray = 64 . INTEGER [EMPTY -1] (0..255)
.. . . . . . . * . SEQUENCE [EMPTY -1] ...
.. . . . . . . . parameterIdentifier . CHOICE [EMPTY -1] ...
.. . . . . . . . standard = 42 . INTEGER [EMPTY -1] (0..127)
.. . . . . . . . parameterValue . CHOICE [EMPTY -1] ...
```

```

. . . . . unsignedMin = 29 . INTEGER [EMPTY -1] (0..65535)
. . . . . * . SEQUENCE [EMPTY -1] ...
. . . . . parameterIdentifier . CHOICE [EMPTY -1] ...
. . . . . . standard = 3 . INTEGER [EMPTY -1] (0..127)
. . . . . parameterValue . CHOICE [EMPTY -1] ...
. . . . . . unsignedMin = 81 . INTEGER [EMPTY -1] (0..65535)
. . . . . * . SEQUENCE [EMPTY -1] ...
. . . . . parameterIdentifier . CHOICE [EMPTY -1] ...
. . . . . . standard = 6 . INTEGER [EMPTY -1] (0..127)
. . . . . parameterValue . CHOICE [EMPTY -1] ...
. . . . . . unsignedMin = 15 . INTEGER [EMPTY -1] (0..65535)
. . . . . * . SEQUENCE [EMPTY -1] ...
. . . . . parameterIdentifier . CHOICE [EMPTY -1] ...
. . . . . . standard = 4 . INTEGER [EMPTY -1] (0..127)
. . . . . parameterValue . CHOICE [EMPTY -1] ...
. . . . . . unsignedMin = 7 . INTEGER [EMPTY -1] (0..65535)
. . . . multiplexParameters . CHOICE [EMPTY -1] ...
. . . . h2250LogicalChannelParameters . SEQUENCE [EMPTY -1] ...
. . . . . sessionID = 2 . INTEGER [EMPTY -1] (0..255)
. . . . . mediaControlChannel . CHOICE [EMPTY -1] ...
. . . . . . unicastAddress . CHOICE [EMPTY -1] ...
. . . . . . ipAddress . SEQUENCE [EMPTY -1] ...
. . . . . . . network = 4 ' .e.' =0xac10650b <172.16.101.11> .OCTET STRING
[EMPTY -1]
. . . . . . tsapIdentifier = 50137 . INTEGER [EMPTY -1] (0..65535)
. . . . . . dynamicRTPPayloadType = 109 . INTEGER [EMPTY -1] (96..127)
. . . . . mediaPacketization . CHOICE [EMPTY -1] ...
. . . . . rtpPayloadType . SEQUENCE [EMPTY -1] ...
. . . . . . payloadDescriptor . CHOICE [EMPTY -1] ...
. . . . . . . oid = 8 {itu-t recommendation h 241 0 0 0 0}.OBJECT
IDENTIFIER [EMPTY -1]
. . . . . . . payloadType = 109 . INTEGER [EMPTY -1] (0..127)

```

This table outlines H.241 to H.264 mappings.

Identifier	Description
Capability name	ITU-T Rec H.241 H.264 Video Capabilities
Capability identifier type	Standard
Capability identifier value	{itu-t(0) recommendation(0) h(8) 241 specificVideoCodecCapabilities(0) h264(0) generic-capabilities(1)}
maxBitRate	This field shall be included, in units of 100 bit/s. This field represents the maximum bitrate of the H.264 Type II bitstream as defined in Annex C/H. 264.
collapsing	This field shall contain the H.264 Capability Parameters as given below.

### Capabilities

The H.264 capability set is structured as a list of one or more H.264 capabilities, each of which has:

- Profile (mandatory)
- Level (mandatory)
- Zero or more additional parameters

These capabilities communicate the ability to decode using one or more H.264 profiles contained in a GenericCapability structure. For each H.264 capability, optional parameters can appear. These parameters permits a terminal to communicate that it has capabilities in addition to meeting the support requirements for the signaled profile and level.

Optional parameters include: CustomMaxMBPS, CustomMaxDPB, CustomMaxBRandCPB, MaxStaticMBPS, max-rcmd-unit-size, max-nal-unit-size, SampleAspectRatiosSupported, AdditionalModesSupported, and AdditionalDisplayCapabilities.

## H.264 Media Packetization

For H.323, systems signal their H.264 mediaPacketization by including:

MediaPacketizationCapability.rtpPayloadType.payloadDescriptor.oid, with the OID having the value {itu-t(0) recommendation(0) h(8) 241 specificVideoCodecCapabilities(0) h264(0) iPpacketization(0) h241AnnexA(0)}.

In compliance with RFC 3984's non-interleaved mode, the following is supported:

MediaPacketizationCapability.rtpPayloadType.payloadDescriptor.oid, with the OID having the value {itu-t(0) recommendation(0) h(8) 241 specificVideoCodecCapabilities(0) h264(0) iPpacketization(0) RFC3984NonInterleaved(1)}.

In compliance with RFC 3984's interleaved mode, the following is supported:

MediaPacketizationCapability.rtpPayloadType.payloadDescriptor.oid, with the OID having the value {itu-t(0) recommendation(0) h(8) 241 specificVideoCodecCapabilities(0) h264(0) iPpacketization(0) RFC3984Interleaved(2)}.

## H.264 in SIP

H.264 in SIP can contain these optional parameters, which be included in the "a=fmtp" line of SDP if they appear: profile-level-id, max-mbps, max-fs, max-cpb, max-dpb, maxbr, redundant-pic-cap, sprop-parameter-sets, parameter-add, packetization-mode, spropinterleaving-depth, deint-buf-cap, sprop-deint-buf-req, sprop-init-buf-time, sprop-max-dondiff, and max-rcmd-nalu-size.

The profile-level-id parameter is a base 16[6] hexadecimal representation of the following three bytes in sequence:

1. profile\_idc
2. profile\_oip—Composed of the values from constraint\_set0\_flag, constraint\_set1\_flag, constraint\_set2\_flag, and reserved\_zero\_5bits—in order of bit significance, starting from the most significant bit.
3. level\_idc—Note that reserved\_zero\_5bits is required to be equal to 0 in [1], but other values for it may be specified in the future by ITU-T or ISO/IEC.

### H.264 Packetization Mode

In SIP, the packetization-mode parameter signals the properties of the RTP payload type or the capabilities of a receiver's implementation. Only a single configuration point can be indicated. So when capabilities support more than one packetization-mode are declared, multiple configuration points (RTP payload types) must be used.

- When the value of packetization-mode equals 0 or packetization-mode is not present, the single NAL mode is used.
- When the value of packetization-mode equals 1, the non-interleaved mode is used.
- When the value of packetization-mode equals 2, the interleaved mode is used.

This example shows a SIP offer-answer exchange. Here is the offer SDP:

```
m=video 49170 RTP/AVP 100 99 98
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42A01E; packetization-mode=0;
a=rtpmap:99 H264/90000
a=fmtp:99 profile-level-id=42A01E; packetization-mode=1;
a=rtpmap:100 H264/90000
a=fmtp:100 profile-level-id=42A01E; packetization-mode=2;
```

And here is the answer SDP for the example:

```
m=video 49170 RTP/AVP 100 99 97
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42A01E; packetization-mode=0;
a=rtpmap:99 H264/90000
a=fmtp:99 profile-level-id=42A01E; packetization-mode=1;
```

## IWF Services

```
a=rtpmap:100 H264/90000
a=fmtp:100 profile-level-id=42A01E; packetization-mode=2;
```

### H.264 IWF Conversions

This section contains two table that show profile, level, and media packetization conversions for H.264 undergoing interworking.

Profile	H.264 in SIP	H.264 (H.241 in H.323)
H264_PROFILE_STR_BASELINE	66	64
H264_PROFILE_STR_MAIN	77	32
H264_PROFILE_STR_EXTENDED	88	32

H.264 Level	H.2264 in SIP	H.264 (H.241 in H.323)	Constraints
1	10	15	0x00
1b	11	19	0x10
1.1	11	22	0x00
1.2	12	29	0x00
1.3	13	36	0x00
2	20	43	0x00
2.1	21	50	0x00
2.2	22	57	0x00
3	30	64	0x00
3.1	31	71	0x00
3.2	32	78	0x00
4	40	85	0x00
4.1	41	92	0x00
4.2	42	99	0x00
5	50	106	0x00
5.1	51	113	0x00

H.264 SIP Packetization	H.264 (H.241 in H.323) OID in mediaPacketization
packetization-mode=0	{itu-t(0) recommendation(0) h(8) 241 specificVideoCodecCapabilities(0) h264(0) iPpacketization(0) h241AnnexA(0)}
packetization-mode=1	{itu-t(0) recommendation(0) h(8) 241 specificVideoCodecCapabilities(0) h264(0) iPpacketization(0) RFC3984NonInterleaved(1)}
packetization-mode=2	{itu-t(0) recommendation(0) h(8) 241 specificVideoCodecCapabilities(0) h264(0)}



H.264 SIP Packetization	H.264 (H.241 in H.323) OID in mediaPacketization
	iPpacketization(0) RFC3984Interleaved(2)}

### IWF Unsupported Parameters

The following H.241 parameters are not supported for interworking: CustomMaxMBPS, CustomMaxFS, CustomMaxDPB, CustomMaxBRandCPB, MaxStaticMBPS, max-rcmd-nal-unit-size, max-nal-unit-size, SampleAspectRatioSupported, AdditionalModesSupported, and AdditionalDisplayCapabilities.

The following SDP parameters are not supported for interworking: max-mbps, max-fs, max-cpb, max-dpb, maxbr, redundant-pic-cap, sprop-parameter-sets, parameter-add, spropinterleaving-depth, deint-buf-cap, sprop-deint-buf-req, sprop-init-buf-time, sprop-max-dondiff, and max-rcmd-nalu-size.

## H.263+ in H.323

This section describes the H.264 capabilities and media packetization in H.323. Capability exchange signaling looks like this:

```

. . . . . capability . CHOICE [EMPTY -1] ...
. . . . . . receiveVideoCapability . CHOICE [EMPTY -1] ...
. . . . . . . h263VideoCapability . SEQUENCE [EMPTY -1] ...
. . . . . . . . sqcifMPI = 1 . INTEGER [EMPTY -1] (1..32)
. . . . . . . . qcifMPI = 1 . INTEGER [EMPTY -1] (1..32)
. . . . . . . . cifMPI = 1 . INTEGER [EMPTY -1] (1..32)
. . . . . . . . maxBitRate = 1000 . INTEGER [EMPTY -1] (1..192400)
. . . . . . . . unrestrictedVector = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . arithmeticCoding = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . advancedPrediction = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . pbFrames = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . temporalSpatialTradeOffCapability = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . errorCompensation = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . h263Options . SEQUENCE [EMPTY -1] ...
. . . . . . . . . advancedIntraCodingMode = 1 . BOOLEAN [EMPTY -1]
. . . . . . . . . deblockingFilterMode = 1 . BOOLEAN [EMPTY -1]
. . . . . . . . . improvedPBFramesMode = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . unlimitedMotionVectors = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . fullPictureFreeze = 1 . BOOLEAN [EMPTY -1]
. . . . . . . . . partialPictureFreezeAndRelease = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . resizingPartPicFreezeAndRelease = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . fullPictureSnapshot = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . partialPictureSnapshot = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . videoSegmentTagging = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . progressiveRefinement = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . dynamicPictureResizingByFour = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . dynamicPictureResizingSixteenthPel = 1 . BOOLEAN [EMPTY -1]
. . . . . . . . . dynamicWarpingHalfPel = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . dynamicWarpingSixteenthPel = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . independentSegmentDecoding = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . slicesInOrder-NonRect = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . slicesInOrder-Rect = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . slicesNoOrder-NonRect = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . slicesNoOrder-Rect = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . alternateInterVLCMode = 1 . BOOLEAN [EMPTY -1]
. . . . . . . . . modifiedQuantizationMode = 1 . BOOLEAN [EMPTY -1]
. . . . . . . . . reducedResolutionUpdate = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . separateVideoBackChannel = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . videoBadMBsCap = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . h263Version3Options . SEQUENCE [EMPTY -1] ...
. . . . . . . . . . dataPartitionedSlices = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . . fixedPointIDCT0 = 0 . BOOLEAN [EMPTY -1]
. . . . . . . . . . interlacedFields = 0 . BOOLEAN [EMPTY -1]

```

```

. . . . . currentPictureHeaderRepetition = 0 . BOOLEAN [EMPTY -1]
. . . . . previousPictureHeaderRepetition = 0 . BOOLEAN [EMPTY -1]
. . . . . nextPictureHeaderRepetition = 0 . BOOLEAN [EMPTY -1]
. . . . . pictureNumber = 0 . BOOLEAN [EMPTY -1]
. . . . . spareReferencePictures = 0 . BOOLEAN [EMPTY -1]

```

**H.263+ in SIP**

H.263+ in SIP appears looks like this:

```

a=rtpmap:100 H263-1998/90000
a=fmtp:100 CIF=1; QCIF=1; SQCIF=1; D=1; F=1; I=1; J=1; L=1; S=1; T=1
a=rtpmap:34 H263/90000
a=fmtp:34 CIF=1; QCIF=1; SQCIF=1

```

**H.263+ IWF Conversions**

This section contains a table showing H.263+ conversions for SIP-h.323 interworking.

<b>H.263+ in H.323 Parameters (Annex) in ftmp line</b>	<b>H.263+ in SIP</b>
sqcifMPI	SQCIF
qcifMPI	QCIF
cifMPI	CIF
	CIF4
	CIF16
maxBitRate	
unrestrictedVector	D
arithmeticCoding	E
advancedPrediction	F
pbFrames	G
temporalSpatialTradeOffCapability	
errorCompensation	H
h263Options	
advancedIntraCodingMode	I
deblockingFilterMode	J
improvedPBFramesMode	
unlimitedMotionVectors	
fullPictureFreeze	L
partialPictureFreezeAndRelease	
resizingPartPicFreezeAndRelease	
fullPictureSnapshot	
partialPictureSnapshot	
videoSegmentTagging	

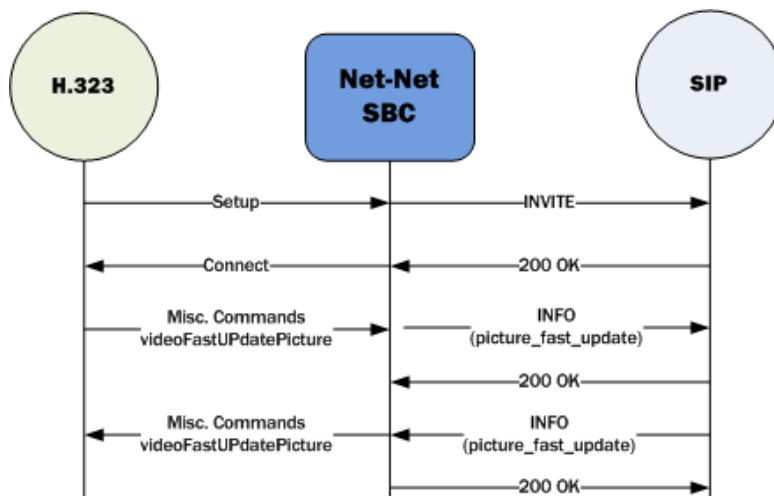
H.263+ in H.323 Parameters (Annex) in ftmp line	H.263+ in SIP
progressiveRefinement	
dynamicPictureResizingByFour	P = 1
dynamicPictureResizingSixteenthPel	P = 2
dynamicWarpingHalfPel	P = 3
DynamicWarpingSixteenthPel	P = 4
independentSegmentDecoding	R
slicesInOrder-NonRect	K = 1
slicesInOrder-Rect	K = 2
slicesNoOrder-NonRect	K = 3
slicesNoOrder-Rect	K = 4
alternateInterVLCMode	S
modifiedQuantizationMode	T
reducedResolutionUpdate	Q
separateVideoBackChannel	
videoBadMBsCap	
	PAR
	CPCF
	CUSTOM
h263Version3Options	

### IWF Unsupported Parameters

The following optional SDP parameters are not supported for H.263+ interworking: SQCIF, QCIF, CIF, CIF4, CIF16, CUSTOM, PAR, CPCF.

## SIP-H.323 IWF in Video Conferencing Applications

For video conferencing and other video applications, the Oracle Communications Session Border Controller supports interworking between the H.323 Miscellaneous Commands videoFastUpdatePicture and the SIP INFO containing XML schema for Full Update. The noted H.323 message commands the video encoder to enter fast-update mode.



There is no configuration required for the interworking between these two messages to work.

## International Peering with IWF and H.323 Calls

When you do not enable this feature, SIP to H.323 IWF calls default to a National Q.931 Number Type and it is not possible to change it to an International number. This feature allows you to override that behavior by configuring the option **cpnType=X**, where X is an integer that maps to various Q.931 Number Types. When this option is set, Q.931 Number Type for both calling party and called party are updated to the configured value for all outgoing calls on the h323-stack.

The following is a list of possible **cpnType=X** option values for X:

- 0—Unknown public number
- 1—International public number
- 2—National public number
- 3—Specific public network number
- 4—Public subscriber number
- 5—Public abbreviated number
- 6—Private abbreviated number

## International Peering Configuration

You configure this feature as an option in the **h323-stack** configuration.

To configure the **cpnType=X** option for H323-H323 calls:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **h323-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# h323-config
ACMEPACKET(h323)#
```

4. Type **h323-stacks** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (h323) # h323-stack
ACMEPACKET (h323-stack) #
```

5. Set the options parameter by typing **options**, a Space, the option name **cpnType=x** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (h323-stack) # options +cpnType=x
```

If you type **options** without the plus sign, you will overwrite any previously configured options. In order to append the new options to the h323-stack's options list, you must prepend the new option with a plus sign as shown in the previous example.

6. Save and activate your configuration.

## IWF Codec Renegotiation for Audio Sessions

For calls requiring interworking between SIP and H.323, there can be several instances for audio sessions when a mid-call codec change is necessary. These are some examples of when the codec used for voice transportation is necessary:

- Sessions between analog FAX machines that start as regular voice calls but then must use a codec that is fax-signalling tolerant (like transparent G.711) when FAX tones are detected; detection takes place after the call has been answered. The case of modem calls is similar.
- An established call is redirected in one carrier's network either to a different enduser or to a media server. In this case, the party to which the call is redirected might not support the codec used in the redirection. If request for a codec change is carried out at the signalling level, the call can proceed with the party to which the call was redirected.
- Endusers might want to change codecs when they suffer low voice quality.

Both SIP and H.323 provide mechanisms for changing codecs during a call: SIP uses the ReINVITE, and H.323 uses the H.245 Request Mode. Using the option called **processRequestModeForIWF=all** either in an H.323 interface (stack) or an H.323 session agent configuration, you can enable the Oracle Communications Session Border Controller to interwork SIP ReINVITE and H.245 Request Mode requests.

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### Codec Request Change from the SIP Side

When a SIP party requests a code change, the Oracle Communications Session Border Controller communicates with the H.323 endpoint to renegotiate support for an updated codec. In this renegotiation, the Oracle Communications Session Border Controller presents codec for use ordered according to the SIP side's preference and one is selected. Then the Oracle Communications Session Border Controller handles opening of a new logical channel that uses the updated codec, and closes the old logical channel (that uses the now-outdated codec). On the SIP side, the Oracle Communications Session Border Controller sends a 200 OK with the necessary RTP port and codec information for the new logical channel.

### Codec Request Change from the H.323 Side

When the Oracle Communications Session Border Controller receives a codec request change on the H.323 side of an IWF call, it sends a Re-INVITE to the SIP endpoint containing new codec and information. The Oracle Communications Session Border Controller uses IP address and port information it has cached for the H.323 side of the call for the Re-INVITE since H.245 Request Mode requests do not have this data. If the IP address and port combination should subsequently change (in an OLC from the H.323 side), the Oracle Communications Session Border Controller handles additional INTVITEs on the SIP side to support the change.

### Exceptional Cases

When the relevant option is enabled, the Oracle Communications Session Border Controller can handle properly the following cases of codec change:

## IWF Services

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- When the H.323 side rejects the request mode change, the Oracle Communications Session Border Controller response to the SIP side with a 488 Not Acceptable. Session description and state remain unchanged, and the call continues using the original session description.
- When the H.323 side does not respond to the request mode change within the timeout limitation, the Oracle Communications Session Border Controller releases the call on both sides.
- When the SIP side does not respond to the ReINVITE within in the timeout limitation, the Oracle Communications Session Border Controller releases the call on both sides.
- When the intersection of codec is empty, the Oracle Communications Session Border Controller rejects the codec change on the SIP side with a 488 Not Acceptable and on the H.323 side with an H.245 RequestModeReject. Session description and state remain unchanged, and the call continues using the original session description.
- If the Oracle Communications Session Border Controller does not receive any of the LogicalChannel request or acknowledgement messages, the Oracle Communications Session Border Controller releases the call on both sides.

Note that for protocol timeout errors, the preferred behavior is to release the call on both sides. Timeout errors usually indicate network problems, such as an endpoint being unreachable.

## IWF Codec Renegotiation Configuration

You can apply the `processRequestModeForIWF=all` to H.323 interfaces (stacks) and to H.323 session agents (sessions agents for which H.323 has been identified in the `protocol` parameter). The example below shows you how to enable this option for an H.323 session agent.

To enable IWF codec renegotiation for an H.323 session agent:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type `session-router` and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type `session-agent` and press Enter. If you are adding this feature to a pre-existing configuration, you will need to select and edit it.

```
ACMEPACKET (session-router) # session-agent
ACMEPACKET (session-agent) #
```

4. **options**—Set the options parameter by typing `options`, a Space, the option name `processRequestModeForIWF=all` with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (session-agent) # options +processRequestModeForIWF=all
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the realm configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save your work.

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# Application Layer Gateway Services

## DNS ALG

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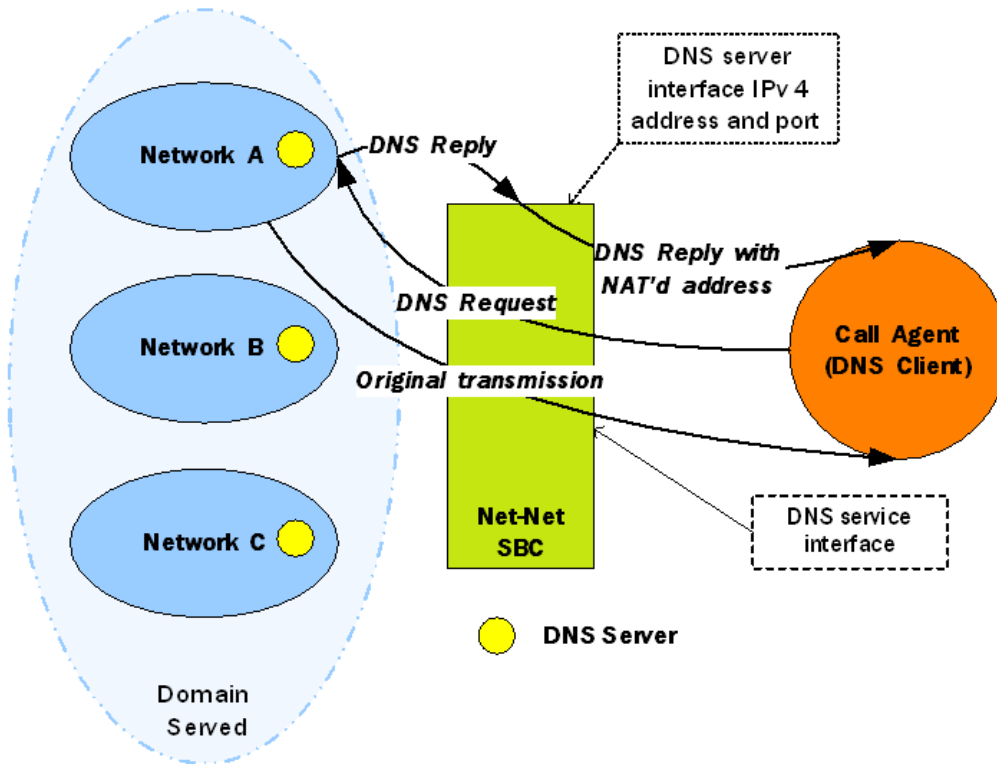
The Oracle Communications Session Border Controller's DNS Application Layer Gateway (ALG) feature provides an application layer gateway for DNS transactions on the Oracle Communications Session Border Controller. With DNS ALG service configured, the Oracle Communications Session Border Controller can support the appearance of multiple DNS servers on one side and a single DNS client on the other.

### Overview

DNS ALG service provides an application layer gateway for use with DNS clients. DNS ALG service allows a client to access multiple DNS servers in different networks and provides routing to/from those servers. It also supports flexible address translation of the DNS query/response packets. These functions allow the DNS client to query many different domains from a single DNS server instance on the client side of the network.

The Oracle Communications Session Border Controller's DNS ALG service is commonly used when a DNS client (such as a call agent) needs to authenticate users. In this case, the DNS client that received a message from a certain network would need to authenticate the endpoint in a remote network. Since the DNS client and the sender of the message are on different networks, the Oracle Communications Session Border Controller acts as an intermediary by interoperating with both.

In the following diagram, the DNS client has received a message from an endpoint in Network A. Since the DNS client is in a different realm, however, the DNS client receives the message after the Oracle Communications Session Border Controller has performed address translation. Then the DNS client initiates a DNS query on the translated address. The Oracle Communications Session Border Controller forwards the DNS request to the DNS server in Network A, using the domain suffix to find the appropriate server. Network A's DNS server returns a response containing its IPv4 address, and then the Oracle Communications Session Border Controller takes that reply and performs a NAT on the private address. The private address is turned into a public one that the DNS client can use to authenticate the endpoint.



### Configuring DNS ALG Service

This section tells you how to access and set the values you need depending on the configuration mechanism you choose. It also provides sample configurations for your reference.

Configuring DNS ALG service requires that you carry out two main procedures:

- Setting the name, realm, and DNS service IP interfaces
- Setting the appropriate parameters for DNS servers to use in other realms

### Before You Configure

Before you begin to configure DNS ALG service on the Oracle Communications Session Border Controller, complete the following steps.

1. Configure the client realm that you are going to use in the main DNS ALG profile and note its name to use in this chapter's configuration process.
2. Configure the server realm that contains the DNS servers and note its name to use in this chapter's configuration process.
3. Determine the domain suffixes for the network where the DNS servers are located so that you can enter them in the domain suffix parameter.
4. Devise the NAT scheme that you want to use when the DNS reply transits the Oracle Communications Session Border Controller.

### DNS ALG Service Name Configuration

This section explains how to configure the name of the DNS ALG service you are configuring and set its realm.

To add DNS ALG service:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `media-manager` and press Enter.



```
ACMEPACKET (configure) # media-manager
```

3. Type **dns-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # dns-config
ACMEPACKET (dns-config) #
```

From this point, you can configure DNS ALG parameters and access this configuration's DNS server subelement. To view all DNS ALG service parameters and the DNS server subelement, enter a ? at the system prompt.

```
dns-config
  client-realm
  description                dns-alg1
  client-address-list
  last-modified-date         2005-02-15 10:50:07
  server-dns-attributes
    server-realm
    domain-suffix
    server-address-list
    source-address
    source-port               53
    transaction-timeout      10
    address-translation
      server-prefix           10.3.0.0/16
      client-prefix
192.168.0.0/16
```

### Identity Realm and Interface Addresses

To configure the identity, realm, and IPv4 interface addresses for your DNS ALG profile:

1. **description**—Set a name for the DNS ALG profile using any combination of characters entered without spaces. You can also enter any combination with spaces if you enclose the whole value in quotation marks. For example: DNS ALG service.
2. **client-realm**—Enter the name of the realm from which DNS queries are received. If you do not set this parameter, the DNS ALG service will not work.
3. **client-address-list**—Configure a list of one or more addresses for the DNS server interface. These are the addresses on the Oracle Communications Session Border Controller to which DNS clients send queries.

To enter one address in this list, type **client-address-list** at the system prompt, a Space, the IPv4 address, and then press Enter

```
ACMEPACKET (dns-config) # client-address-list 192.168.0.2
```

To enter more than one address in this list, type **client-address-list** at the system prompt, and a Space. Then type an open parenthesis ( ( ), each IPv4 address you want to use separated by a Space, and closed parenthesis ( ) ), and then press Enter.

```
ACMEPACKET (dns-config) # client-address-list (192.168.0.2 196.168.1.1 192.168.1.2)
```

### DNS Server Attributes

To configure attributes for the DNS servers that you want to use in the DNS ALG profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET # configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
```

3. Type **dns-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.


```
ACMEPACKET (media-manager) # dns-config
```

4. Type **server-dns-attributes** and then press Enter.

```
ACMEPACKET (dns-config) # server-dns-attributes
```

From this point, you can configure DNS server parameters. To see all parameters for the DNS server, enter a ? at the system prompt.

5. **server-realm**—Enter the name of the realm in which the DNS server is located. This value is the name of a configured realm.
6. **domain-suffix**—Enter a list of one or more domain suffixes to indicate the domains you want to serve. These values are matched when a request is sent to a specific DNS server. If you leave this list empty (default), then your configuration will not work.

 **Note:** If you want to use a wildcard value, you can start your entry to an asterisk ( \* ) (e.g. \*.com). You can also start this value with a dot (e.g., .com).

To enter one address in this list, type **client-address-list** at the system prompt, a Space, the domain suffix, and then press Enter

```
ACMEPACKET (server-dns-attributes) # domain-suffix acmepacket.com
```

To enter more than one address in this list, type **domain-suffix** at the system prompt, and a Space. Then type an open parenthesis ( ( ), each IPv4 address you want to use separated by a Space, and closed parenthesis ( ) ), and then press Enter.

```
ACMEPACKET (server-dns-attributes) # domain-suffix (acmepacket.com  
acmepacket1.com acmepacket2.com)
```

7. **server-address-list**—Enter a list of one or more DNS IPv4 addresses for DNS servers. These DNS servers can be used for the domains you specified in the domain suffix parameter. Each domain can have several DNS servers associated with it, and so you can populate this list with multiple IPv4 addresses. If you leave this list empty (default), your configuration will not work.
8. **source-address**—Enter the IPv4 address for the DNS client interface on the Oracle Communications Session Border Controller. If you leave this parameter empty (default), your configuration will not work.
9. **source-port**—Enter the number of the port for the DNS client interface on the Oracle Communications Session Border Controller. The default value is **53**. The valid range is:
  - Minimum—1025
  - Maximum—65535
10. **transaction-timeout**—Enter the time in seconds that the ALG should keep information to map a DNS server response back to the appropriate client request. After the transaction times out, further response to the original request will be discarded. The default value is **10**. The valid range is:
  - Minimum—0
  - Maximum—999999999
11. **address-translation**—Enter a list of address translations that define the NAT function for the DNS servers.

You can access the NAT parameters for the DNS servers by typing **address-translation** and pressing enter within the DNS server attributes configuration.

```
ACMEPACKET (dns-config) # server-dns-attributes  
ACMEPACKET (server-dns-attributes) # address-translation
```

To configure the NAT, enter two values:

- **server-prefix**: address/prefix that will be returned by the DNS server
- **client-prefix**: address/prefix that to which a response is returned

Each of these is a two-part value:

- IPv4 address
- Number of bits indicating how much of the IPv4 address to match

If you do not specify the number of bits, then all 32 bits of the IPv4 address will be used for matching. If you set the number of bits to 0, then the address will simply be copied.

For example, if you set the server prefix to 10.3.17.2/16 and the client prefix to 192.168.0.0/16, then the Oracle Communications Session Border Controller will return an address of 192.168.17.2 to the DNS client.

```
ACMEPACKET (server-dns-attributes) # address-translation
ACMEPACKET (address-translation) # server-prefix 10.3.17.2/16
ACMEPACKET (address-translation) # client-prefix 192.168.0.0/16
```

## DNS Transaction Timeout

To provide resiliency during DNS server failover, you can now enable a transaction timeout for DNS servers. If you have endpoints that are only capable of being configured with a single DNS server, this can allow DNS queries to be sent to the next configured server—even when contacting the Oracle Communications Session Border Controller’s DNS ALG on a single IP address. So when the first server in the list times out, the request is sent to the next server in the list.

The Oracle Communications Session Border Controller uses the transaction timeout value set in the **dns-server-attributes** configuration (part of the **dns-config**).

### DNS Transaction Timeout Configuration

To enable the DNS transaction timeout:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **media-manager** and press Enter

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **media-manager** and press Enter.

```
ACMEPACKET (media-manager) # media-manager
ACMEPACKET (media-manager-config) #
```

4. **dnsalg-server-failover**—Change this parameter from **disabled** (default) to **enabled** to allow DNS queries to be sent to the next configured server—even when contacting the Oracle Communications Session Border Controller’s DNS ALG on a single IP address. So when the first server in the list times out, the request is sent to the next server in the list. The Oracle Communications Session Border Controller uses the transaction timeout value set in the **dns-server-attributes** configuration (part of the **dns-config**).

5. Save your work.

### DNS Server Operation States

After the first failed attempt to reach a DNS server, The Oracle Communications Session Border Controller places it in a Time Out state. The server stays in Time Out state for 30 seconds. The Oracle Communications Session Border Controller does not send DNS queries to a server in Time Out state. Instead, it directs queries to the next DNS server in the server-address-list. After 30 seconds, the DNS server goes back to an In Service state and the Oracle Communications Session Border Controller sends queries to it.

If a DNS server fails to respond to 5 consecutive queries, it goes into Out of Service (OOS) state. The Oracle Communications Session Border Controller directs all subsequent queries to the first In Service server. The Oracle Communications Session Border Controller returns OOS DNS servers to In Service state after 600 seconds and repeats the cycle above. If, for any given call, all configured DNS servers are OOS, the Oracle Communications Session Border Controller fails the call.

### DNS ALG Message Throttling

---

You can limit DNS ALG-bound throughput by using the `dns-alg-constraints` configuration element. Message throttling is performed on request messages, and since DNS-ALG is transaction stateful, the responses are automatically throttled. Once a `dns-alg-constraints` element is configured, it can be referenced by a DNS configuration object. This allows users to create constraint profiles and apply them to multiple DNS configuration objects.

When any of the constraints are exceeded, the DNS server's status changes to Constraints Exceeded and requests are rejected with a 503 error message. The server, via the ALG, remains in that state until the time-to-resume parameter period ends. When this period ends, the Oracle Communications Session Border Controller re-examines the server's traffic with respect to the current burst or sustain window and the state is determined again, along with the decision to readmit traffic.

### Bursty Traffic Throttling

You can create separate inbound and outbound maximum burst rates. First configure a `burst rate window` in seconds. Then configure the `max inbound burst rate` and `max outbound burst rate` parameters for the traffic you wish to constrain. In addition you can set an overall maximum burst rate with the `max burst rate` parameter.

Regardless of the inbound or outbound burst rates' headroom before exceeding the message constraints, if the total in/out traffic exceeds the `max burst rate` (when configured), the target DNS server is set to Constraints Exceeded status and taken out of service.

### Sustained Traffic Throttling

You can create separate inbound and outbound maximum sustained traffic rates. First configure a `sustain rate window` in seconds. Then configure the `max inbound sustain rate` and `max outbound sustain rate` parameters. In addition you can set an overall sustain rate with the `max sustain rate` parameter.

Regardless of the inbound or outbound sustained traffic rates' headroom before exceeding the message constraints, if the total in/out traffic exceeds the `max sustained rate` (when configured), the target DNS server is set to Constraints Exceeded status and taken out of service.

### Maximum Latency

The `max-latency` parameter sets a threshold under which requests must be responded to. If this value is exceeded, the DNS server is put into the Constraints Exceeded State.

### DNS ALG Constraints Configuration

To configure the DNS ALG Constraints object:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `media-manager` and press Enter to access the media-manager path.

```
ACMEPACKET(configure)# media-manager
```

3. Type `dnsalg-constraints` and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# dnsalg-constraints
ACMEPACKET(dnsalg-constraints)#
```

4. **name**—Enter a name for this `dnsalg-constraints` configuration element. You will reference this value from the `constraint-name` parameter in the `dns-config` configuration element.
5. **state**—Set this parameter to `enabled` to enable this configuration element.

6. **max-burst-rate**—Enter the maximum number of messages that can pass through the system in the burst rate window before setting the element to Constraints Exceeded.
7. **max-inbound-burst-rate**—Enter the maximum number of inbound messages received by the referencing element within the burst rate window before setting the element to Constraints Exceeded.
8. **max-outbound-burst-rate**—Enter the maximum number of outbound messages forwarded from the referencing element within the burst rate window before setting the element to Constraints Exceeded.
9. **burst-rate-window**—Enter the number of seconds during which to count messages toward a maximum burst rate.
10. **max-sustain-rate**—Enter the maximum number of messages that can pass through the system in the sustained rate window before setting the element to Constraints Exceeded.
11. **max-inbound-sustain-rate**—Enter the maximum number of inbound messages received by the referencing element within the sustained rate window before setting the element to Constraints Exceeded.
12. **max-outbound-sustain-rate**—Enter the maximum number of outbound messages forwarded from the referencing element within the sustained rate window before setting the element to Constraints Exceeded.
13. **sustain-rate-window**—Enter the number of seconds during which to count messages toward a maximum sustained rate.
14. **max-latency**—Enter the time the Oracle Communications Session Border Controller waits to receive a response to a request. After this value is exceeded, the DNS server is placed in Constraints Exceeded state.
15. **time-to-resume**—Enter the number of seconds that the referencing element stays in Constraints Exceeded state and rejects messages before it returns to service.
16. Type **done** when finished.

### Applying Traffic Constraints

To apply DNS ALG constraints to an existing DNS config object:

1. Type **exit** to return to the media-manager path.

```
ACMEPACKET(dnsalg-constraints)# exit
ACMEPACKET(media-manager)#
```

2. Type **dns-config** and **select** an existing configuration element.

```
ACMEPACKET(media-manager)# dns-config
ACMEPACKET(dns-config)# select
<realm-id>:
1: realm01
selection: 1
```

3. **constraint-name**—Enter the name of an existing diameter-director-constraints configuration element.
4. Type **done** when finished.

### ACLI DNS ALG Statistics

The Oracle Communications Session Border Controller provides comprehensive DNS ALG monitoring with the `show dns-alg` command. This command is entered as:

```
ACMEPACKET# show dnsalg [lookup | rate | stats | status]
```

Entered without any arguments, this command shows a summary of all dnsalg agents.

You must set the **extra-dnsalg-stats** parameter in the `dns-config` to enabled to collect these statistics.

```
ACMEPACKET#show dnsalg stats
      ---Queries---  --Successful--  ---NotFound---  ---TimedOut--
DNS ALG Realm  Current  Total  Current  Total  Current  Total  Current  Total
net172         0        1        0        1        0        0        0        0
net192         0        0        0        0        0        0        0        0
Avg Latency=0.000 for 0
```

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```
Max Latency=0.000  
Rate =0.0
```

### Other Show commands

```
ACMEPACKET# show dnsalg stats <realm-id>  
ACMEPACKET# show dnsalg lookup <client realm>  
ACMEPACKET# show dnsalg status <dnsalg-client-realm>
```

### SNMP DNS ALG Statistics and Traps

The AP-DNSALG-MIB is defined to provide standard-compliant object definitions and descriptions for the SNMP information provided by Oracle Communications Session Border Controller for dns-alg query related information. This MIB corresponds to the related ACLI show commands.

By setting the **trap-on-status-change** parameter in the dns-config to enabled, the following traps become active on the system:

Traps	Description
apDnsAlgStatusChangeTrap apDnsAlgStatusChangeClearTrap	This trap is set and clears when any DNS-config object changes its operating status.
apDnsAlgConstraintStateChangeTrap apDnsAlgConstraintStateChangeClearTrap	This trap is set and clears when any dns-config object enters constraints exceeded state.
apDnsAlgSvrConstraintStateChangeTrap apDnsAlgSvrConstraintStateChangeClearTr ap	This trap is set and clears when any single DNS server enters constraints exceeded state.

Refer to the MIB ZIP archive to view the AP-DNSALG-MIB MIB file.

---

## Session Routing and Load Balancing

This chapter explains how to configure session routing and load balancing for SIP and H.323 services. It contains information about configuring session agents and session agent groups, as well as local policies that can be used for routing SIP or H.323 signals.

---

### Routing Overview

This section provides an overview of routing SIP and H.323 sessions when using the Oracle Communications Session Border Controller. The Oracle Communications Session Border Controller chooses the next hop through the network for each SIP and H.323 session based on information received from routing policies and constraints. Routing policies can be as simple as routing all traffic to a proxy or routing all traffic from one network to another. Routing policies can also be more detailed, using constraints to manage the volume and rate of traffic that can be routed to a specific network. For example, you can manage volume and rate of traffic to enable the Oracle Communications Session Border Controller to load balance and route around softswitch failures.

When a call request arrives at the Oracle Communications Session Border Controller, a decision making process then occurs to determine whether the message is coming from a session agent. If so, the Oracle Communications Session Border Controller checks whether that session agent is authorized to make the call. Local policy is then checked to determine where to send the message on to.

---

### Session Agents Session Groups and Local Policy

When you configure session routing for SIP and H.323, you can use session agents, session agent groups and local policies to define routing. (Using session agents and session agent groups is not required.)

- session agent: defines a signaling endpoint. It is a next hop signaling entity that can be configured to apply traffic shaping attributes.
- session agent group (SAG): contains individual session agents. Members of a SAG are logically equivalent (although they might vary in their individual constraints) and can be used interchangeably.

You apply an allocation strategy to the SAG to allocate traffic across the group members. Session agent groups also assist in load balancing among session agents.

- local policy: indicates where session request messages, such as SIP INVITES, are routed and/or forwarded. You use a local policy to set a preference for selecting one route over another.

Another element of routing is the realm. Realms are used when a any log level you set here overrides the log level you set in the system configuration's process log level parameter, communicates with multiple network elements over

## Session Routing and Load Balancing

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a shared intermediate connection. Defining realms allows sessions to go through a connection point between the two networks.

When you configure a realm, you give it an identifier, which stores the name of the realm associated with the any log level you set here overrides the log level you set in the system configuration's process log level parameter. The realm identifier value is also needed when you configure session agents and local policies. You can associate a realm with a session agent to identify the realm for sessions coming from or going to the session agent. You also need the realm identifier when you configure local policy to identify the egress realm (realm of the next hop).

## About Session Agents

---

This section describes session agents. A session agent defines a signaling endpoint. It is a next hop signaling entity that can be configured to apply traffic shaping attributes. Service elements such as gateways, softswitches, and gatekeepers are defined automatically within the Oracle Communications Session Border Controller as session agents. For each session agent, concurrent session capacity and rate attributes can be defined. You can group session agents together into session agent groups and apply allocation strategies to achieve traffic load balancing.

You can assign a media profile to a session agent and indicate whether the transport protocol is SIP or H.323. If the protocol is H.323, you need to indicate whether the session agent is a gateway or a gatekeeper.

You can configure a set of attributes and constraints for each session agent to support the following:

- session access control: Oracle Communications Session Border Controller only accepts requests from configured session agents
- session admission control (concurrent sessions): Oracle Communications Session Border Controller limits the number of concurrent inbound and outbound sessions for any known service element.
- session agent load balancing: session agents are loaded based on their capacity and the allocation strategy specified in the session agent group.
- session (call) gapping: Oracle Communications Session Border Controller polices the rate of session attempts to send to and receive from a specific session agent.

## SIP Session Agents

---

SIP session agents can include the following:

- softswitches
- SIP proxies
- application servers
- SIP gateways
- SIP endpoints

In addition to functioning as a single logical next hop for a signaling message (for example, where a SIP INVITE is forwarded), session agents can provide information about next or previous hops for packets in a SIP agent, including providing a list of equivalent next hops.

You can use the session agent to describe one or more SIP next or previous hops. Through the configured carriers list, you can identify the preferred carriers to use for traffic coming from the session agent. This set of carriers will be matched against the local policy for requests coming from the session agent. You can also set constraints for specific hops.

## Session Agent Status Based on SIP Response

The Oracle Communications Session Border Controller can take session agents out of service based on SIP response codes that you configure, and you can also configure SIP response codes that will keep the session agent in service.

With this feature disabled, the Oracle Communications Session Border Controller determines session agents' health by sending them ping messages using a SIP method that you configure. Commonly, the method is an OPTIONS



request. If it receives any response from the session agent, then the Oracle Communications Session Border Controller deems that session agent available for use.

However, issues can arise when session agents are administratively out of service, but able to respond to OPTIONS requests. A session agent like this might only respond with a 200 OK when in service, and send a 4xx or 5xx message otherwise.

The session agent status feature lets you set the SIP response message that either takes a session agent out of service or allows it to remain in service when it responds to the Oracle Communications Session Border Controller's ping request.

Details of this feature are as follows:

- The Oracle Communications Session Border Controller only considers a session agent in service when it responds to a request method you set with the final response code that you also set. If a final response code is set, then provisional responses are not used for determining whether or not to take a session agent out of service. If the Oracle Communications Session Border Controller receives a final response code that does not match the session agent configuration, it treats the session agent as though it had not responded.
- The Oracle Communications Session Border Controller takes a session agent out of service when it receives an error response for dialog creating request with a response code listed in the new **out-service-response-codes** parameter.

In the case where a the session agent's response has a Retry-After header, the Oracle Communications Session Border Controller tries to bring the session agent back into service after the period of time specified in the header. To do so, it sends another ping request.

There are two lists you can configure in the session agent configuration to determine status:

- In-service list—Set in the ACLI **ping-in-service-response-codes** parameter, this list defines the response codes that keep a session agent in service when they appear in its response to the Oracle Communications Session Border Controller's ping request. Furthermore, the Oracle Communications Session Border Controller takes the session agent out of service should a response code be used that does not appear on this list.
- Out-of-service list—Set in the ACLI **out-service-response-codes** parameter, this list defines the response codes that take a session agent out of service when they appear in its response to the Oracle Communications Session Border Controller's ping request or any dialog-creating request.

When the Oracle Communications Session Border Controller receives a session agent's response to its ping request, it first checks to see if there is an in-service list of responses configured for that session agent. If the list is configured and the Oracle Communications Session Border Controller determines that there is a match, the session agent is deemed in service. Otherwise it takes the session agent out of service. In this way, the in-service list takes precedence over the out-of-service list. If you configure the in-service list, then the Oracle Communications Session Border Controller ignores the out-of-service list.

If there is no list of in-service responses for the session agent, then the Oracle Communications Session Border Controller checks the out of service list. If it is configured and the Oracle Communications Session Border Controller determines that there is a match, the Oracle Communications Session Border Controller removes that session agent from service. If there is no match, then the session agent is deemed in service.

## SIP Session Agent Continuous Ping

You can configure the Oracle Communications Session Border Controller to use either a keep-alive or continuous method for pinging SIP session agents to determine their health—i.e., whether or not the Oracle Communications Session Border Controller should route requests to them. To summarize the two methods:

- keep-alive—Oracle Communications Session Border Controller sends a ping message of a type you configure to the session agent in the absence of regular traffic. Available in Release C5.1.0 and in earlier releases.
- continuous—The Oracle Communications Session Border Controller sends a ping message regardless of traffic state (regular or irregular); the Oracle Communications Session Border Controller regularly sends a ping sent based on the configured ping interval timer. Available in Release C5.1.1p6 and in later releases.

## Session Routing and Load Balancing

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By sending ping messages, the Oracle Communications Session Border Controller monitors session agents' health and can determine whether or not to take a session out of service (OOS), leave it in service, or bring it back into service after being OOS.

When you set it to use the keep-alive mode of pinging (available in Release C5.1.0 and before), the Oracle Communications Session Border Controller starts sending a configured ping message to a session agent when traffic for that session agent has become irregular. The Oracle Communications Session Border Controller only sends the ping if there are no SIP transactions with a session agent over a configurable period of time, to which the session agent's response can have one of the following results:

- **Successful response**—A successful response is either any SIP response code or any response code not found in the **out-service-response-codes** parameter; these leave the session agent in service. In addition, any successful response or any response in the **ping-in-service-response-codes** parameter can bring a session agent from OOS to in-service status.
- **Unsuccessful response**—An unsuccessful response is any SIP response code configured in the **out-service-response-codes** parameter and takes the session agent sending it OOS. Because this parameter is blank by default, the Oracle Communications Session Border Controller considers any SIP response code successful.
- **Transaction timeout**—A transaction timeout happens when the session agent fails to send a response to the Oracle Communications Session Border Controller's request, resulting in the session agent's being taken OOS.

Despite the fact that the keep-alive ping mode is a powerful tool for monitoring session agents' health, you might want to use the continuous ping method if you are concerned about the Oracle Communications Session Border Controller not distinguishing between unsuccessful responses from next-hop session agents and ones from devices downstream from the next-hop session agent. For example, if a SIP hop beyond the session agent responds with a 503 Service Unavailable, the Oracle Communications Session Border Controller does not detect whether a session agent or the device beyond it generated the response.

When you use the continuous ping method, only the next-hop session agent responds—preventing the request from being sent to downstream devices. The Oracle Communications Session Border Controller also sends the ping in regular traffic conditions when in continuous ping mode, so it is certain the response comes from the next hop associated with the session agent. And in continuous ping mode, only entries for the **ping-out-service-response-codes** parameter and transaction timeouts bring session agents OOS.

### SIP SA Continuous Ping Configuration

You can set the ping mode in the session agent or session constraints configuration. For backward compatibility, the default for the **ping-send-mode** parameter is **keep-alive**, or the functionality available in Release C5.1.0 and in earlier releases.

To configure the ping mode for a session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

If you are adding rate constraints to an existing configuration, then you will need to select the configuration you want to edit.

4. **ping-send-mode**—If to want to use continuous ping mode to send ping messages to session agents in regular traffic conditions, set this parameter to **continuous**. If you want to use the keep-alive mode, leave this parameter set to **keep-alive** (default).
5. Save and activate your configuration.

To configure the ping mode for the session constraints:

- In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

- Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

- Type **session-constraints** and press Enter.

```
ACMEPACKET (session-router) # session-constraints
ACMEPACKET (session-constraints) #
```

If you are adding rate constraints to an existing configuration, then you will need to select the configuration you want to edit.

- ping-send-mode**—If to want to use continuous ping mode to send ping messages to session agents in regular traffic conditions, set this parameter to **continuous**. If you want to use the keep-alive mode, leave this parameter set to **keep-alive** (default).
- Save and activate your configuration.

## H.323 Session Agents

H.323 session agents can include the following:

- Gatekeepers
- Gateways
- MCUs

### Overlapping H.323 Session Agent IP Address and Port

You can now configure H.323 session agents to use overlapping IP addresses.

H.323 session agents continue are identified by their hostname when used in referencing configuration parameters—such as local policy next hops and session agent group destinations. This is why the hostname must be unique. However, when the Oracle Communications Session Border Controller selects a session agent to use, it chooses the appropriate realm and H.323 stack based on the hostname. This is the case even if there are other session agents with the same IP address and port. Likewise, incoming calls are matched to the session agent based on the incoming realm.

There are no specific parameters to configure in order to enable this feature. For it to work properly, however, each H.323 session agent must be configured with a unique hostname (still the primary index). Otherwise, session agents with non-unique hostnames will overwrite one another.

To create overlapping H.323 session agents, you give each of them a unique hostname, which only serves to identify each individually. The Oracle Communications Session Border Controller subsequently uses this label as the next hop destination in relevant local policy route entries.

### Managing Session Agent Traffic

The Oracle Communications Session Border Controller monitors availability, session load, and session rate for each session agent in real time. The session agent's state is determined by its performance relative to the constraints applied to it and its availability.

The following table lists the conditions that cause the Oracle Communications Session Border Controller to suspend the routing of traffic to a session agent, along with the criteria for restoring the route.

## Session Routing and Load Balancing

Constraint Condition	SIP Criteria	H.323 Criteria	Action	Criteria for Resuming
Maximum sessions exceeded	Maximum concurrent SIP sessions exceeded.	Maximum concurrent H.323 sessions exceeded.  If the session agent is a gatekeeper and gatekeeper routed mode is not used, this constraint is an aggregate of all the destination gateways. Only maximum outbound sessions are measured.	Session agent is declared in constraint violation state.	Concurrent sessions drop below the maximum sessions value.
Maximum outbound sessions exceeded	Maximum concurrent outbound SIP sessions exceeded.	Maximum concurrent outbound H.323 sessions exceeded.  If the session agent is a gatekeeper and gatekeeper routed mode is not used, this constraint is an aggregate of all the destination gateways. Only maximum outbound sessions are measured.	Session agent is declared in constraint violation state.	Concurrent sessions drop below the maximum outbound sessions value.
Maximum burst rate exceeded	Maximum burst rate exceeded in current window.	Maximum burst rate exceeded in current window.  If the session agent is a gatekeeper and gatekeeper routed mode is not used, this constraint is an aggregate of all the destination gateways. Only maximum outbound sessions are measured.	Session agent is declared in constraint violation state.	Burst rate in subsequent window drops below maximum burst rate.
Maximum sustained rate exceeded	Maximum sustained rate exceeded in current window.	Maximum burst rate exceeded in current window.  If the session agent is a gatekeeper and gatekeeper routed mode is not used, this constraint is an aggregate of all the	Session agent is declared in constraint violation state.	Sustained rate in subsequent window drops below the maximum sustained rate.

Constraint Condition	SIP Criteria	H.323 Criteria	Action	Criteria for Resuming
		destination gateways. Only maximum outbound sessions are measured.		
Session agent unavailable or unresponsive	SIP transaction expire timer expires for any out-of-dialogue request. For example, INVITE, REGISTER, or ping.	Response timer expires. The default is T301=4 seconds. Connect timer expires. The default is T303=32 seconds. If the session agent is a peer gatekeeper, the LRQ response time is used to determine availability. The RAS response timer is 4 seconds.	Session agent is declared in constraint violation state or out-of-service. The time to resume timer starts.	Time to resume timer expires and the Oracle Communications Session Border Controller declares the session agent in-service. or Session agent responds to subsequent pings (SIP only).

## Session Agent Groups

Session agent groups contain individual session agents. Members of a session agent group are logically equivalent (although they might vary in their individual constraints) and can be used interchangeably. You can apply allocation strategies to session agent groups.

Examples of session agent groups include the following:

- application server cluster
- media gateway cluster
- softswitch redundant pair
- SIP proxy redundant pair
- gatekeeper redundant pair

Session agent group members do not need to reside in the same domain, network, or realm. The Oracle Communications Session Border Controller can allocate traffic among member session agents regardless of their location. It uses the allocation strategies configured for a SAG to allocate traffic across the group members.

Allocation strategies include the following:

Allocation Strategy	Description
Hunt	Oracle Communications Session Border Controller selects the session agents in the order in which they are configured in the SAG. If the first agent is available, and has not exceeded any defined constraints, all traffic is sent to the first agent.  If the first agent is unavailable, or is in violation of constraints, all traffic is sent to the second agent. And so on for all session agents in the SAG. When the first agent returns to service, the traffic is routed back to it.
Round robin	Oracle Communications Session Border Controller selects each session agent in the order in which it is configured, routing a session to each session agent in turn.

## Session Routing and Load Balancing

Allocation Strategy	Description
Least busy	Oracle Communications Session Border Controller selects the session agent with the least number of active sessions, relative to the maximum outbound sessions or maximum sessions constraints (lowest percent busy) of the session agent.
Proportional distribution	Session agents are loaded proportionately based upon the respective maximum session constraint value configured for each session agent.
Lowest sustained rate	Oracle Communications Session Border Controller routes traffic to the session agent with the lowest sustained session rate, based on observed sustained session rate.

You apply allocation strategies to select which of the session agents that belong to the group should be used. For example, if you apply the Hunt strategy session agents are selected in the order in which they are listed.

### Request URI Construction as Sent to SAG-member Session Agent

The Oracle Communications Session Border Controller constructs the request URI for a session agent selected from a session agent group by using the **session-agent > hostname** value of the selected **session-agent** target. This default behavior enables features such as trunk groups and ENUM to work properly. However, care must be given when the **hostname** parameter is not a resolvable FQDN. The **sag-target-uri=<value>** option can be used to overcome the default behavior.

The value is either

- **ip** – request URI constructed from **session-agent > ip-address**
- **host** – request URI constructed from **session-agent > hostname**

This option is global and is configured in the **sip-config** configuration element.

### Request URI Construction as Forwarded to SAG-member Session Agent

The Oracle Communications Session Border Controller constructs the request URI for a session agent selected from a session agent group by using the **session-agent > hostname** value of the selected **session-agent** target. This default behavior enables features such as trunk groups and ENUM to work properly. However, care must be given when the **hostname** parameter is not a resolvable FQDN. Use the **sag-target-uri=<value>** option to override the default behavior.

The value can be set to either:

- **ip** - request URI constructed from **session-agent > ip-address**
- **host** - request URI constructed from **session-agent > hostname**

This option is global and is configured in the **sip-config** configuration element.

### SIP Session Agent Group Recursion

You can configure a SIP session agent group (SAG) to try all of its session agents rather than to the next-best local policy match if the first session agent in the SAG fails.

With this feature disabled, the Oracle Communications Session Border Controller performs routing by using local policies, trunk group URIs, cached services routes, and local route tables. Local policies and trunk group URIs can use SAGs to find the most appropriate next-hop session agent based on the load balancing scheme you choose for that SAG: round robin, hunt, proportional distribution, least busy, and lowest sustained rate. When it locates a SAG and selects a specific session agent, the Oracle Communications Session Border Controller tries only that single session agent. Instead of trying other members of the SAG, the Oracle Communications Session Border Controller recurses to the local policy that is the next best match. This happens because the Oracle Communications Session Border Controller typically chooses a SAG based on the fact that it has not breached its constraints, but the Oracle Communications Session Border Controller only detects failed call attempts (due to unreachable next hops,

unresolved ENUM queries, or SIP 4xx/5xx/6xx failure responses) after it has checked constraints. So the Oracle Communications Session Border Controller only re-routes if there are additional matching local policies.

When you enable SIP SAG recursion, the Oracle Communications Session Border Controller will try the additional session agents in the selected SAG if the previous session agent fails. You can also set specific response codes in the SAG configuration that terminate the recursion. This method of terminating recursion is similar to the Oracle Communications Session Border Controller's ability to stop recursion for SIP interfaces and session agents.

Session agents are selected according to the strategy you set for the SAG, and these affect the way that the Oracle Communications Session Border Controller selects session agents when this feature enabled:

- Round robin and hunt—The Oracle Communications Session Border Controller selects the first session agent according to the strategy, and it selects subsequent session agents based on the order they are entered into the configuration.
- Proportional distribution, least busy, and lowest sustained rate—The Oracle Communications Session Border Controller selects session agents based on the list of session agents sorted by the criteria specified.

You can terminate recursion based on SIP response codes that you enter into the SAG configuration. You can configure a SAG with any SIP response code in the 3xx, 4xx, and 5xx groups. Since you can also set such a list in the session agent configuration, this list is additive to that one so that you can define additional codes for a session agent group without having to repeat the ones set for a session agent.

## About Local Policy

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This section explains the role of local policy. Local policy lets you indicate where session requests, such as SIP INVITES, should be routed and/or forwarded. You use a local policy to set a preference for selecting one route over another. The local policy contains the following information that affects the routing of the SIP and H.323 signaling messages:

- information in the From header

Information in the message's From header is matched against the entries in the local policy's from address parameter to determine if the local policy applies.

- list of configured realms

This list identifies from what realm traffic is coming and is used for routing by ingress realm. The source realms identified in the list must correspond to the valid realm IDs you have already configured

- local policy attributes

The attributes serve as an expression of preference, a means of selecting one route over another. They contain information such as the next signaling address to use (next hop) or whether you want to select the next hop by codec, the realm of the next hop, and the application protocol to use when sending a message to the next hop. You can also use the attributes to filter specific types of traffic.

## Routing Calls by Matching Digits

Local policy routing of a call can be based on matching a sequence of digits against what is defined in the local policy. This sequence refers to the first digits in the (phone) number, matching left to right.

The following examples show how the Oracle Communications Session Border Controller matches an area code or number code against configured local policies.

- If the number or area code being matched is 1234567 (where 123 is an area code), and the from address value in one local policy is 123, and the from address value in another local policy is 12, the Oracle Communications Session Border Controller forwards the call to the server that is defined as the next hop in the local policy with 123 as the from address value.
- If the number or area code being matched is 21234, and the from address value in one local policy is 123, and the from address value in another local policy is 12, the Oracle Communications Session Border Controller will not find a match to either local policy because the first character of the number or area code must match the first character in a from address or to address field.

## Session Routing and Load Balancing

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The following examples show how the Oracle Communications Session Border Controller matches an area or number code against different local policies: the first one has a From address value of 12 and the second has a From address value of 123. The Oracle Communications Session Border Controller chooses the route of the local policy that is configured with the most digits matching the area or number code in its From address and To address fields.

- When the two different local policies route to two different servers, and the area or number code being matched is 123, the Oracle Communications Session Border Controller selects the second local policy based on the From address value of 123.
- When the two different local policies route to two different servers, and the area or number code being matched is 124, the Oracle Communications Session Border Controller selects the first local policy based on the From address value of 12.

## SIP and H.323 Interworking


You need to configure local policies, including the requisite local policy attributes, to use the H.323<—>SIP interworking (IWF). Flow progression in H.323<—>SIP traffic depends heavily on the local policies configured for the Oracle Communications Session Border Controller, which determine what protocol is used on the egress side of a session.

You set the application protocol (an local policy attribute option) to instruct the Oracle Communications Session Border Controller to interwork the protocol of an ingress message into a different protocol (H.323<—>SIP or SIP—>H.323) upon its egress to the next hop.

For example, if the application protocol is set to SIP, an inbound H.323 message will be interworked to SIP as it is sent to the next hop. An inbound SIP message would pass to the next hop unaffected. If the application protocol is set to H323, an inbound SIP message will be interworked to H.323 before being sent to the next hop.

## Route Preference

The Oracle Communications Session Border Controller builds a list of possible routes based on the source realm and the From-address (From-URI) and To-address (Request-URI), which forms a subset from which preference then decides. Any local policy routes currently outside of the configured time/day are not used, if time/day are set. Also, any local policy routes not on the list of carriers (if carriers is set and the requests has a Carrier header) are not used.

 **Note:** Source realm is used in the local policy lookup process, but it is not used in route preference calculations.

The Oracle Communications Session Border Controller applies preference to configured local policies in the following order:

1. Cost (cost in local policy attributes) is always given preference.
2. Matching media codec (media profiles option in local policy attributes).
3. Longest matching To address (to address list in local policy).
4. Shortest matching To address (to address list in local policy).
5. Longest matching From address (from address list in local policy).
6. Shortest matching From address (from address list in local policy).
7. Narrowest/strictest day of week specification (days of week option in local policy attributes).
8. Narrowest/strictest time of day specification (start time and end time options in local policy attributes).
9. Wildcard matches (use of an asterisk as a wildcard value for the from address and to address lists in local policy).
10. Wild card matches are given the least preference. A prefix value of 6 is given a higher preference than a prefix value of \* even though both prefix values are, in theory, the same length.

## DTMF-Style URI Routing

The Oracle Communications Session Border Controller supports the alphanumeric characters a-d, A-D, the asterisk (\*), and the ampersand (#) for local policy matching purposes. The Oracle Communications Session Border Controller handles these characters as standards DN (POTS) or FQDN when found in the to-addr (req-uri username) or from-addr (from0uri username for SIP, SIPS, and TEL URIs).



In addition, before performing the lookup match, the Oracle Communications Session Border Controller strips characters that provide ease-of-reading separation. For example, if the Oracle Communications Session Border Controller were to receive a req-uri containing tel:a-#1-781-328-5555, it would treat it as tel:a#17813285555.

## SIP Routing

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This section describes SIP session routing. When routing SIP call requests, the Oracle Communications Session Border Controller communicates with other SIP entities, such as SIP user devices, other SIP proxies, and so on, to decide what SIP-based network resource each session should visit next. The Oracle Communications Session Border Controller processes SIP call requests and forwards the requests to the destination endpoints to establish, maintain, and terminate real-time multimedia sessions.

Certain items in the messages are matched with the content of the local policy, within constraints set by the previous hop session agent, and the SIP configuration information (for example, carrier preferences) to determine a set of applicable next hop destinations.

The sending session agent is validated as either a configured session agent or a valid entry in a user cache. If the session INVITATION does not match any registering user, the SIP proxy determines the destination for routing the session INVITATION.

### Limiting Route Selection Options for SIP

You can configure the local policy to use the single most-preferred route. And you can configure the SIP configuration max routes option to restrict the number of routes which can be selected from a local policy lookup:

- A **max-routes=1** value limits the Oracle Communications Session Border Controller to only trying the first route from the list of available preferred routes.
- A **max-routes=0** value or no **max-routes** value configured in the options field allows the Oracle Communications Session Border Controller to use all of the routes available to it.

A Oracle Communications Session Border Controller configured for H.323 architectures will have access to all of the routes it looks up by default.

### About Loose Routing

According to RFC 3261, a proxy is loose routing if it follows the procedures defined in the specification for processing of the Route header field. These procedures separate the destination of the request (present in the Request-URI) from the set of proxies that need to be visited along the way (present in the Route header field).

When the SIP NAT's route home proxy field is set to enabled, the Oracle Communications Session Border Controller looks for a session agent that matches the home proxy address and checks the loose routing field value. If the loose routing is:

- **enabled**—A Route header is included in the outgoing request in accordance with RFC 3261.
- **disabled**—A Route header is not included in the outgoing request; in accordance with the route processing rules described in RFC 2543 (referred to as strict routing). That rule caused proxies to destroy the contents of the Request-URI when a Route header field was present.

Whether loose routing field is enabled is also checked when a local policy 's next hop value matches a session agent. Matching occurs if the hostname or the session agent's IP address field value corresponds to the next hop value. If loose routing is enabled for the matching session agent, the outgoing request retains the original Request-URI and Route header with the next hop address.

### About the Ingress Realm

You can create a list of realms in your local policy that is used by the Oracle Communications Session Border Controller to determine how to route traffic. This list determines from which realm traffic is coming and is used for routing by ingress realm.

The source realm values must correspond to valid identifier entered when the realm was configured.

### About the Egress Realm

An egress realm allows SIP signaling to travel out of the Oracle Communications Session Border Controller through a network other than the home realm. The Oracle Communications Session Border Controller uses egress realms for signaling purposes (when matching flows). When a packet arrives at the Oracle Communications Session Border Controller with a destination address that does not match any defined session agents, the Oracle Communications Session Border Controller uses the address associated with the realm that is, in turn, associated with the SIP configuration's egress realm ID, as the outgoing network. With the use of the egress realm ID, it is possible to define a default route for SIP requests addressed to destinations outside the home realm. If no egress realm is defined, the home realm (default ingress realm) is used as the default egress realm.

With session agent egress realm configured, the Oracle Communications Session Border Controller adds a default egress realm to the session agent to identify the signaling interface used for ping requests. The Oracle Communications Session Border Controller also uses the default egress realm when the normal routing request does not yield an egress realm—for example, when a local policy does not specify the next hop's realm.

When you configure session agents, you can define them without realms or you can wildcard the realm value. These are global session agents, and multiple signaling interfaces can reach them. Then, when you use session agent pinging, the Oracle Communications Session Border Controller sends out ping requests using the signaling interface of the default egress realm defined in the global SIP configuration. The global session agents in certain environments can cause problems when multiple global session agents residing in multiple networks, some of which might not be reachable using the default SIP interface egress realm.

The Oracle Communications Session Border Controller uses the session agent egress realm for ping messages even when the session agent has a realm defined. For normal request routing, the Oracle Communications Session Border Controller uses the egress realm for global session agents when local policies or SIP-NAT bridge configurations do not point to an egress realm.

#### Ping Message Egress Realm Precedence

For ping messages, the egress realm precedence occurs in the following way (in order of precedence):

- Egress realm identified for the session agent.
- Session agent realm (set in the realm-id parameter) or the wildcarded value
- Global SIP configuration egress realm, when configured in the egress-realm parameter
- Global SIP configuration home realm

#### Normal Request Egress Realm Precedence

For normal request routing, the egress realm precedence occurs in the following way (in order of precedence):

- Egress SIP-NAT realm, when the **route-home-proxy** parameter is set to **forced** and no local policy match is found
- Matching local policy realm, when configured in the local policy attributes
- Session agent realm (set in the realm-id parameter) or the wildcarded value
- Session agent egress realm, when configured in the egress-realm-id parameter
- Global SIP configuration egress realm, when configured in the egress-realm parameter
- Global SIP configuration home realm

#### Session Agent Egress Realm Configuration

Configuring a session agent egress realm is optional.

To configure a session agent egress realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

If you are adding this feature to an existing configuration, you need to select the configuration (using the ACLI **select** command) before making your changes.

4. **egress-realm-id**—Enter the name of the realm you want defined as the default egress realm used for ping messages. The Oracle Communications Session Border Controller will also use this realm when it cannot determine the egress realm from normal routing. There is no default value for this parameter.
5. Save and activate your configuration.

## About SIP Redirect

SIP redirect involves proxy redirect and tunnel redirect.

### Proxy Redirect

You can configure the SIP proxy mode to define how the SIP proxy will forward requests coming from the session agent. This value is used if the session agent's proxy mode has no value (is empty).

### Tunnel Redirect

You can use tunnel redirect when requests are routed to a server behind a SIP NAT that sends redirect responses with addresses that should not be modified by the SIP NAT function. For example, a provider might wish to redirect certain calls (like 911) to a gateway that is local to a the UA that sent the request. Since the gateway address is local to the realm of the UA, it should not be modified by the SIP NAT of the server's realm. Note that the server must have a session agent configured with the redirect-action field set to the proxy option in order to cause the redirect response to be sent back to the UA.

## SIP Method Matching and To Header Use for Local Policies

For SIP, this feature grants you greater flexibility when using local policies and has two aspects: basing local policy routing decisions on one or more SIP methods you configure and enabling the Oracle Communications Session Border Controller to use the TO header in REGISTER messages for routing REGISTER requests.

### SIP Methods for Local Policies

This feature allows the Oracle Communications Session Border Controller to include SIP methods in routing decisions. If you want to use this feature, you set a list of one or more SIP methods in the local policy attributes. These are the SIP methods you can enter in the list: INVITE, REGISTER, PRACK, OPTIONS, INFO, SUBSCRIBE, NOTIFY, REFER, UPDATE, MESSAGE, and PUBLISH.

After the Oracle Communications Session Border Controller performs a local policy look-up for SIP, it then searches for local policy attributes that have this methods list configured. If it finds a a set of policy attributes that matches a method that matches the traffic it is routing, the Oracle Communications Session Border Controller uses that set of policy attributes. This means that the Oracle Communications Session Border Controller considers first any policy attributes with methods configured before it considers those that do not have methods. In the absence of any policy attributes with methods, the Oracle Communications Session Border Controller uses the remaining ones for matching.

In cases where it finds neither matching policy attributes with methods or matching policy attributes without them, the Oracle Communications Session Border Controller either rejects the calls with a 404 No Routes Found (if the request calls for a response) or drops the call.

You configure local policy matching with SIP methods in the local policy attributes parameter calls **methods**. This parameter is a list that takes either one or multiple values. If you want to enter multiple values, you put them in the same command line entry, enclosed in quotation marks and separated by spaces.

To configure SIP methods for local policy matching:

## Session Routing and Load Balancing

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1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **local-policy** and press Enter. If you are adding this feature to a pre-existing local policy configuration, you will need to select and edit your local policy.

```
ACMEPACKET (session-router) # local-policy
ACMEPACKET (local-policy) #
```

4. Type **policy-attributes** and press Enter. If you are adding this feature to a pre-existing local policy configuration, you will need to select and edit your local policy.

```
ACMEPACKET (local-policy) # policy-attributes
ACMEPACKET (policy-attributes) #
```

5. **methods**—Enter the SIP methods you want to use for matching this set of policy attributes. Your list can include: INVITE, REGISTER, PRACK, OPTIONS, INFO, SUBSCRIBE, NOTIFY, REFER, UPDATE, MESSAGE, and PUBLISH.

By default, this parameter is empty—meaning that SIP methods will not be taken into consideration for routing based on this set of policy attributes.

If you want to enter more than one method, your entry will resemble the following example.

```
ACMEPACKET (local-policy-attributes) # methods "PRACK INFO REFER"
```

6. Save and activate your configuration.

### Routing Using the TO Header

For the Oracle Communications Session Border Controller's global SIP configuration, you can enable the use of an ENUM query to return the SIP URI of the Registrar for a SIP REGISTER message. Without this feature enabled, the Oracle Communications Session Border Controller uses the REQUEST URI. This ability can be helpful because REGISTER messages only have the domain in the REQUEST URI, whereas the SIP URI in the To header contains the user's identity.

There are two parts to enabling this feature. First, you must enable the **register-use-to-for-lp** parameter in the global SIP configuration. Then you can set the next-hop in the applicable local policy attributes set to **ENUM**.

To enable your global SIP configuration for routing using the TO header:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-config** and press Enter. If you are adding this feature to a pre-existing SIP configuration, you will need to select and edit it.

```
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) #
```

4. **register-use-to-for-lp**—Set this parameter to enabled if you want the Oracle Communications Session Border Controller to use, for routing purposes, an ENUM query to return the SIP URI of the Registrar for a SIP REGISTER message. This parameter defaults to **disabled**.

To set up your local policy attributes for routing using the TO header:

5. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

6. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

7. Type **local-policy** and press Enter. If you are adding this feature to a pre-existing local policy configuration, you will need to select and edit your local policy.

```
ACMEPACKET (session-router) # local-policy
ACMEPACKET (local-policy) #
```

8. Type **policy-attributes** and press Enter. If you are adding this feature to a pre-existing local policy configuration, you will need to select and edit your local policy.

```
ACMEPACKET (local-policy) # policy-attributes
ACMEPACKET (policy-attributes) #
```

9. **next-hop**—This is the next signaling host. Set this parameter to ENUM if you want to use SIP methods in local policy attribute information for routing purposes.

10. Save and activate your configuration.

## H.323 Routing

This section describes H.323 routing.

### Egress Stack Selection

Egress stack selection includes static stack selection and policy-based stack selection

#### Static Stack Selection

In static stack selection, the outgoing stack is determined through the establishment of associated stacks in the h323 stack.

The incoming stack (configured in the h323 stack) uses its associated stack value to determine the associated outgoing stack. The associated stack value corresponds to the name of an h323 stack. This type of selection is referred to as static because the incoming stack always uses the stack specified in the associated stack as the outgoing stack; no other stacks are considered.

#### Policy-Based Stack Selection

The Oracle Communications Session Border Controller performs dynamic, policy-based stack selection when an H.323 call arrives at the Oracle Communications Session Border Controller and a configured associated outgoing stack cannot be found.

For policy-based stack selection, the Oracle Communications Session Border Controller refers to local policies that contain address information that corresponds to incoming traffic. This information is contained in the local policy's To address and From address fields. For the source, this information is matched with the Q.931 calling party number; if there is no calling party number, the H.323 source address is used. For the destination, this information is matched with the called party number; if there is no called party number, then the H.323 destination address is used.

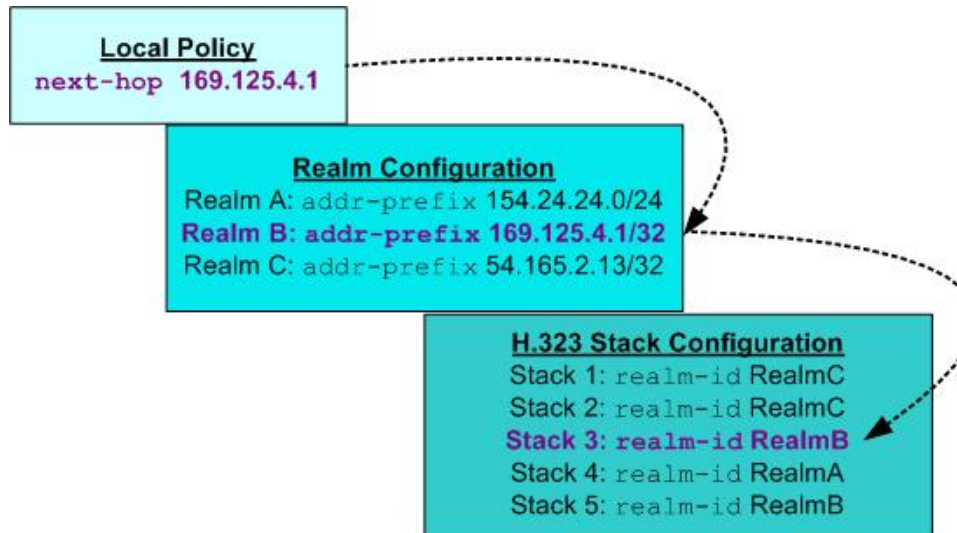
After a local policy corresponding to the incoming traffic has been found, the Oracle Communications Session Border Controller looks at the next hop value (a local policy attribute) and selects a local policy for the basis of stack selection. If the local policy look-up yields multiple local policies with the same next hop values, but with different cost values, the local policy with the lowest cost value is selected.

If a realm is not defined in the local policy, the next hop address is then matched against the address prefix values for the realms that are configured for the system. Thus, the Oracle Communications Session Border Controller discovers the realm for this traffic. Using this realm information, the Oracle Communications Session Border Controller performs stack selection. It uses the first configured H.323 stack in the Oracle Communications Session Border

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Controller's configuration that has a realm ID value matching the identifier field of the realm with the appropriate address prefix.

In the following example, the local policy matching yields a local policy with a next hop value of 169.125.4.1, which corresponds to RealmB. The outgoing stack selected is Stack 3 because it is the first stack to have been configured with RealmB as the realm ID.



Policy-Based Stack Selection

## Registration Caching

The Oracle Communications Session Border Controller can cache and proxy an H.225 RegistrationRequest (RRQ) between an H.323 endpoint and a gatekeeper. Registration caching serves two functions:

- It allows aggregation of RRQs sent to a gatekeeper stack and proxies those requests through the gateway stack. If the external gatekeeper associated with the gatekeeper stack supports additive registration, the requests will be consolidated. Furthermore, if the gatekeeper supports additive registration, the Oracle Communications Session Border Controller will register in an additive manner, meaning that will send additive RRQs.
- It allows the gatekeeper stack to use the registration information to route calls from other realms to endpoints in its realms.

To perform registration caching, the Oracle Communications Session Border Controller must be configured with at least two stacks. One of these stacks will receive registrations (gatekeeper stack), and one stack will proxy registrations (gateway stack). The Oracle Communications Session Border Controller caches all successful registrations and uses the cache to route calls to the endpoints.

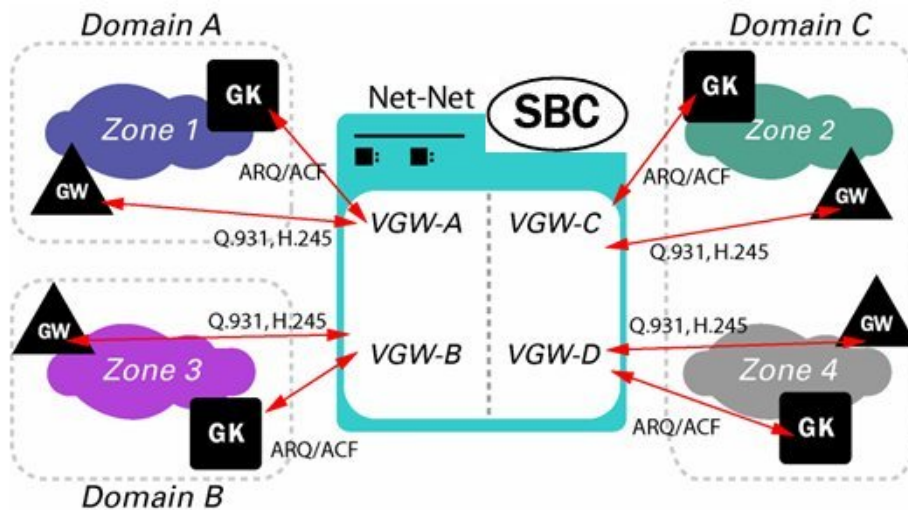
## Gatekeeper Provided Routes

Gatekeeper provided routes includes back-to-back gateways, back-to-back gatekeeper and gateway, and interworking gatekeeper/gateway.

### Back-to-Back Gateway

When the Oracle Communications Session Border Controller is functioning as a back-to-back gateway (B2BGW), it appears as multiple H.323 gateways to multiple networks. Each Oracle Communications Session Border Controller virtual gateway discovers and registers with a gatekeeper in its respective domain. Each gateway relies on its gatekeeper for admission and location services through the ARQ/ACF exchange. H.225 call control and H.245 messages are exchanged directly with the terminating gateway or gatekeeper. Routing policies are used to associate one virtual gateway with another.

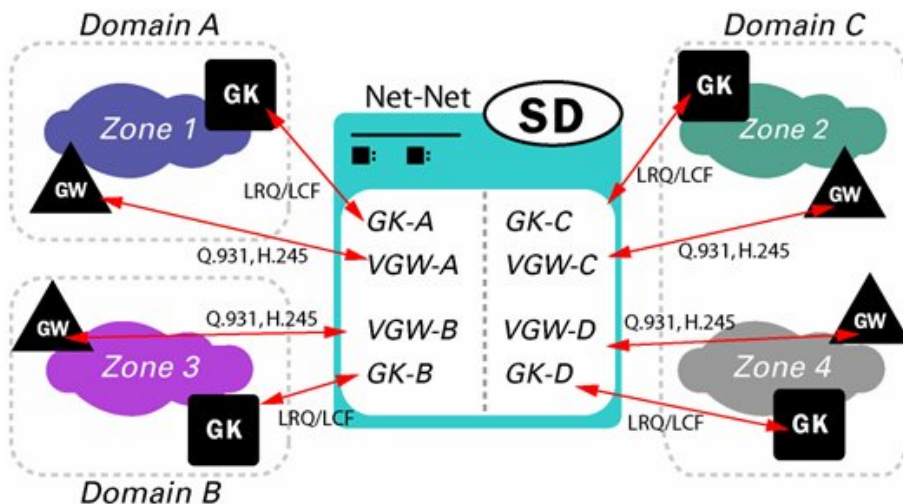
The following diagram illustrates the back-to-back gateway.



### Back-to-Back Gatekeeper and Gateway

For peering connections where both networks use inter-domain gatekeeper signaling, the Oracle Communications Session Border Controller is configured as a back-to-back gatekeeper proxy and gateway mode of operation. The Oracle Communications Session Border Controller responds and issues LRQs and LCFs/LRJs acting as a routed gatekeeper. Peered gatekeepers send LRQ to the RAS address of one of the Oracle Communications Session Border Controller’s virtual gatekeepers and it responds by providing its call signaling address that performs the gateway functions. Routing policies are used to determine the egress virtual gatekeeper that then exchanges LRG/LCF to determine the call signaling address of the terminating gateway.

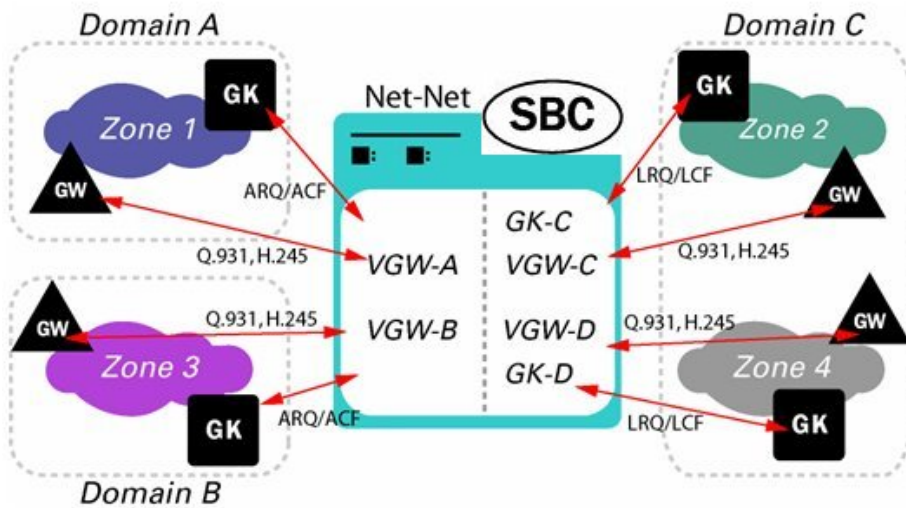
The following diagram illustrates the back-to-back gatekeeper and gateway.



### Interworking Gatekeeper Gateway

In the interworking gatekeeper/gateway signaling mode of operation, the Oracle Communications Session Border Controller interworks between the other two modes; presenting a routed gatekeeper interface to one zone and a gateway to the other.

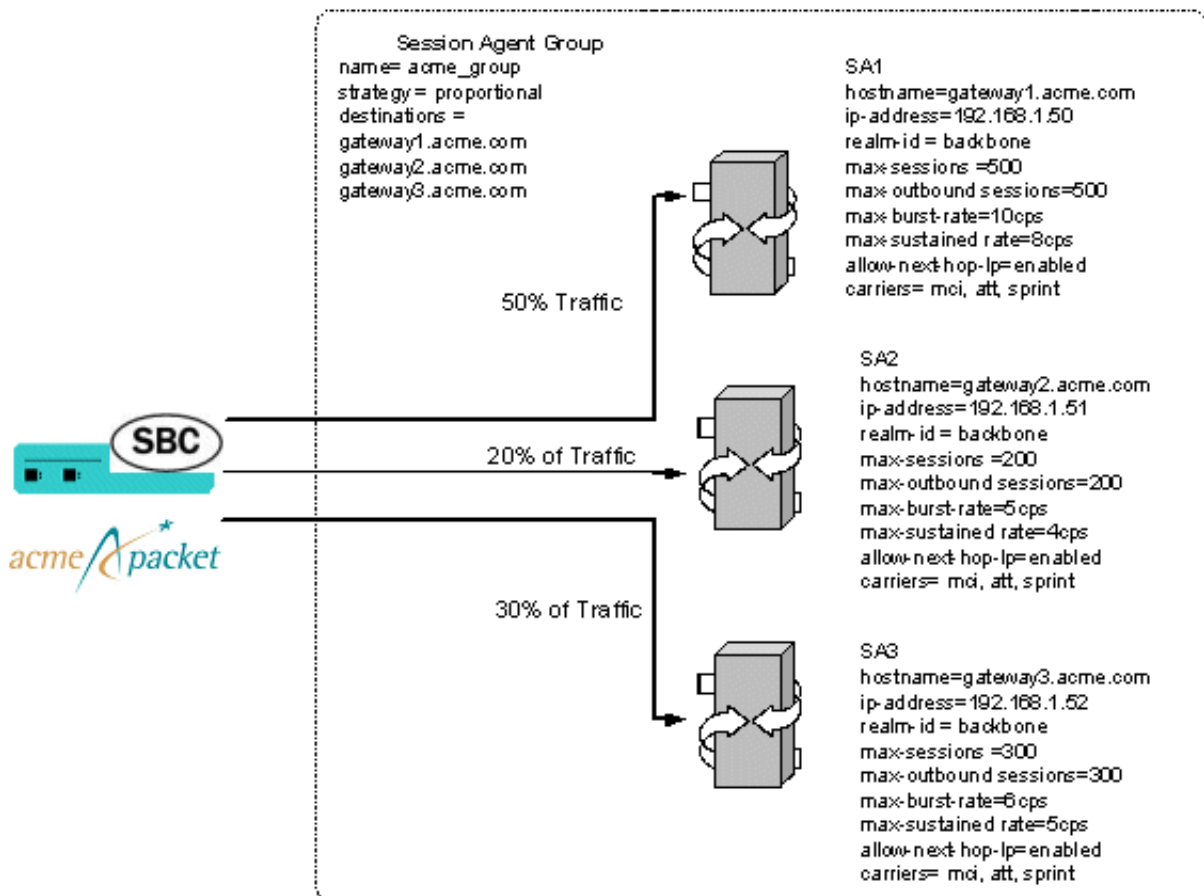
The following diagram illustrates the interworking gatekeeper/gateway.



## Load Balancing

This section describes Oracle Communications Session Border Controller load balancing. You can use session agent groups to assist in load balancing among session agents. You define concurrent session capacity and rate attributes for each session agent and then define the session agent group. Next, you select the allocation strategy you want applied to achieve the load balancing you want.

The following example shows a configuration for load balancing gateways based on a proportional allocation strategy.





Routing and load balancing capabilities include the following:

- least cost, which includes cost-based and time-based routing
- customer preference
- traffic aggregation
- routing by media (codec) type
- capacity-based, by destination
- service element load balancing
- service element failure detection and re-route
- session agent failure
- routing by codec

## Configuring Routing

This section explains how to configure routing on the Oracle Communications Session Border Controller.

### Configuration Prerequisite

You should have already configured the realms for your environment before you configure the routing elements. You need to know the realm identifier when configuring session agents and local policy.

You can use an asterisk (\*) when the session agent exists in multiple realms.

### Configuration Order

Recommended order of configuration:

- realm
- session agent
- session agent group
- local policy

## Routing Configuration

You can enable, then configure, individual constraints that are applied to the sessions sent to the session agent. These constraints can be used to regulate session activity with the session agent. In general, session control constraints are used for session agent groups or SIP proxies outside or at the edge of a network. Some individual constraints, such as maximum sessions and maximum outbound sessions are not applicable to core proxies because they are transaction stateful, instead of session stateful. Other constraints, such as maximum burst rate, burst rate window, maximum sustained rate, and sustained rate are applicable to core routing proxies.

### Configuring Session Agents

To configure session agents:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # session-agent  
ACMEPACKET (session-agent) #
```


4. **host-name**—Enter the name of the host associated with the session agent in either hostname or FQDN format, or as an IP address.

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If you enter the host name as an IP address, you do not have to enter an IP address in the optional IP address parameter. If you enter the host name in FQDN format, and you want to specify an IP address, enter it in the optional IP address parameter. Otherwise you can leave the IP address parameter blank to allow a DNS query to resolve the host name.


If the initial DNS query for the session agent fails to get back any addresses, the session agent is put out-of-service. When session agent is pinged, the DNS query is repeated. The ping message is not sent until the DNS query gets back one or more IP addresses. After the query receives some addresses, the ping message is sent. The session agent remains out of service until one of the addresses responds.

 **Note:** The value you enter here must be unique to this session agent. No two session agents can have the same hostname.

The hostnames established in the session agent populate the corresponding fields in other elements.

5. **ip-address**—Optional. Enter the IP address for the hostname you entered in FQDN format if you want to specify the IP address. Otherwise, you can leave this parameter blank to allow a DNS query to resolve the host name.
6. **port**—Enter the number of the port associated with this session agent. Available values include:
  - zero (0)—If you enter zero (0), the Oracle Communications Session Border Controller will not initiate communication with this session agent (although it will accept calls).
  - 1025 through 65535

The default value is **5060**.

 **Note:** If the transport method value is TCP, the Oracle Communications Session Border Controller will initiate communication on that port of the session agent.

7. **state**—Enable or disable the session agent by configuring the state. By default, the session agent is **enabled**.
  - enabled | disabled
8. **app-protocol**—Enter the protocol on which you want to send the message. The default value is **SIP**. Available values are:
  - SIP | H.323
9. **app-type**—If configuring H.323, indicate whether the application type is a gateway or a gatekeeper. Available values include:
  - **H.323-GW**—gateway
  - **H.323-GK**—gatekeeper
10. **transport-method**—Indicate the IP protocol to use (transport method) to communicate with the session agent. **UDP** is the default value. The following protocols are supported:
  - **UDP**—Each UDP header carries both a source port identifier and destination port identifier, allowing high-level protocols to target specific applications and services among hosts.
  - **UDP+TCP**—Allows an initial transport method of UDP, followed by a subsequent transport method of TCP if and when a failure or timeout occurs in response to a UDP INVITE. If this transport method is selected, INVITEs are always sent through UDP as long as a response is received.
  - **DynamicTCP**—dTCP indicates that dynamic TCP connections are the transport method for this session agent. A new connection must be established for each session originating from the session agent. This connection is torn down at the end of a session.
  - **StaticTCP**—sTCP indicates that static TCP connections are the transport method for this session agent. Once a connection is established, it remains and is not torn down.
  - **DynamicTLS**—dTLS indicates that Dynamic TLS connections are the transport method for this session agent. A new connection must be established for each session originating from the session agent. This connection is torn down at the end of a session.
  - **StaticTLS**—sTLS indicates that Static TLS connections are the transport method for this session agent. Once a connection is established, it will remain and not be torn down.

11. **realm-id**—Optional. Indicate the ID of the realm in which the session agent resides.

The realm ID identifies the realm for sessions coming from or going to this session agent. For requests coming from this session agent, the realm ID identifies the ingress realm. For requests being sent to this session agent, the realm ID identifies the egress realm. In a Oracle Communications Session Border Controller, when the ingress and egress realms are different, the media flows must be steered between the realms.

- no value: the egress realm is used unless the local policy dictates otherwise
- asterisk (\*): keep the egress realm based on the Request URI



**Note:** The realm ID you enter here must match the valid identifier value entered when you configured the realm.

12. **description**—Optional. Enter a descriptive name for this session agent.

13. **carriers**—Optional. Add the carriers list to restrict the set of carriers used for sessions originating from this session agent.

Carrier names are arbitrary names that can represent specific service providers or traditional PSTN telephone service providers (for sessions delivered to gateways). They are global in scope, especially if they are exchanged in TRIP. Therefore, the definition of these carriers is beyond the scope of this documentation.

You could create a list using carrier codes already defined in the North American Numbering Plan (NANP); or those defined by the local telephone number or carrier naming authority in another country.



**Note:** If this list is empty, any carrier is allowed. If it is not empty, only local policies that reference one or more of the carriers in this list will be applied to requests coming from this session agent.

14. **allow-next-hop-lp**—Indicate whether this session agent can be used as a next hop in the local policy.

If you retain the default value of **enabled**, the session agent can be used as the next hop for the local policy. Valid values are:

- enabled | disabled

15. **constraints**—Enable this parameter to indicate that the individual constraints you configure in the next step are applied to the sessions sent to the session agent. Retain the default value of **disabled** if you do not want to apply the individual constraints. Valid values are:

- enabled | disabled



**Note:** In general, session control constraints are used for SAGs or SIP proxies outside or at the edge of a network.

16. Enter values for the individual constraints you want applied to the sessions sent to this session agent. The following table lists the available constraints along with a brief description and available values.

Constraint	Description
maximum sessions	Maximum number of sessions (inbound and outbound) allowed by the session agent. The range of values is: <ul style="list-style-type: none"> <li>• minimum: zero (0) is the default value and means there is no limit</li> <li>• maximum: 4294967295</li> </ul>
maximum outbound sessions	Maximum number of simultaneous outbound sessions (outbound from the Oracle Communications Session Border Controller) that are allowed from the session agent. The range of values is: <ul style="list-style-type: none"> <li>• minimum: zero (0) is the default value and means there is no limit</li> <li>• maximum: 4294967295</li> </ul> The value you enter here cannot be larger than the maximum sessions value.

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Constraint	Description
maximum burst rate	<p>Number of session invitations allowed to be sent to or received from the session agent within the configured burst rate window value. SIP session invitations arrive at and leave from the session agent in intermittent bursts. By entering a value in this field, you can limit the amount of session invitations that are allowed to arrive at and leave from the session-agent.</p> <p>For example, if you enter a value of 50 here and a value of 60 (seconds) for the burst rate window constraint, no more than 50 session invitations can arrive at or leave from the session agent in that 60 second time frame (window). Within that 60-second window, any sessions over the limit of 50 are rejected.</p> <p>The range of values is:</p> <ul style="list-style-type: none"> <li>• minimum: zero (0) session invitations per second</li> <li>• maximum: 4294967295 session invitations per second</li> </ul> <p>Zero is the is the default value.</p>
maximum sustain rate	<p>Maximum rate of session invitations (per second) allowed to be sent to or received from the session agent within the current window. The current rate is determined by counting the number of session invitations processed within a configured time period and dividing that number by the time period. By entering a value in this field, you can limit the amount of session invitations that are allowed to arrive at and leave from the session agent over a sustained period of time.</p> <p>For the sustained rate, the Oracle Communications Session Border Controller maintains a current and previous window size. The period of time over which the rate is calculated is always between one and two window sizes.</p> <p>For example, if you enter a value of 5000 here and a value of 3600 (seconds) for the sustain rate window constraint, no more than 5000 session invitations can arrive at or leave from the session agent in any given 3600 second time frame (window). Within that 3600-second window, sessions over the 5000 limit are rejected.</p> <p>The range of values is:</p> <ul style="list-style-type: none"> <li>• minimum: zero (0) invitations per second</li> <li>• maximum: 4294967295 invitations per second</li> </ul> <p>Zero is the is the default value.</p> <p>The value you set here must be larger than the value you set for the maximum burst rate constraint.</p>
time to resume	<p>Time in seconds after which the SIP proxy resumes sending session invitations to this session agent. This value only takes effect when the SIP proxy stops sending invitations because a constraint is exceeded.</p> <p>The range of values is:</p> <ul style="list-style-type: none"> <li>• minimum: zero (0) seconds</li> <li>• maximum: 4294967295 seconds</li> </ul> <p>Default is zero.</p>

Constraint	Description
time to resume (ttr) no response	<p>Delay in seconds that the SIP proxy must wait from the time that it sends an invitation to the session agent and gets no response before it tries again.</p> <p>The range of values is:</p> <ul style="list-style-type: none"> <li>• minimum: zero (0) seconds</li> <li>• maximum: 4294967295 seconds</li> </ul> <p>Default is zero.</p> <p>The value you enter here must be larger than the value you enter for the time to resume constraint.</p>
in service period	<p>Amount of time in seconds the session agent must be operational (once communication is re-established) before the session agent is declared as being in-service (ready to accept session invitations). This value gives the session agent adequate time to initialize.</p> <p>The range of values is:</p> <ul style="list-style-type: none"> <li>• minimum: zero (0) seconds</li> <li>• maximum: 4294967295 seconds</li> </ul> <p>Default is zero.</p>
burst rate window	<p>Burst window period (in seconds) that is used to measure the burst rate. The term window refers to the period of time over which the burst rate is computed. Refer to the maximum burst rate information.</p> <p>The range of values is:</p> <ul style="list-style-type: none"> <li>• minimum: zero (0) seconds</li> <li>• maximum: 4294967295seconds</li> </ul> <p>Zero is the is the default value.</p> <p>The value you set here must be smaller than the value you set for the maximum burst rate constraint.</p>
sustain rate window	<p>Sustained window period (in seconds) that is used to measure the sustained rate. Refer to the maximum sustain rate information.</p> <p>The range of values is:</p> <ul style="list-style-type: none"> <li>• minimum: zero (0) seconds</li> <li>• maximum: 4294967295seconds</li> </ul> <p>Zero is the is the default value.</p> <p>The value you set here must be larger than the value you set for the maximum sustain rate constraint.</p>

**17. req-uri-carrier-mode**—SIP only. Set whether you want the selected carrier (determined by a value in the local policy) added to the outgoing message by configuring the request uri carrier mode parameter.

You can set this parameter to let the system perform simple digit translation on calls sent to gateways. A 3-digit prefix is inserted in front of the telephone number (the Request-URI) that the gateway will use to select a trunk group. Most often, the Oracle Communications Session Border Controller needs to insert the carrier code into the signaling message that it sends on.

The default value is **none**. The following lists the available modes.

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- **none**—Carrier information will not be added to the outgoing message.
- **uri-param**—Adds a parameter to the Request-URI. For example, cic-XXX.
- **prefix**—Adds the carrier code as a prefix to the telephone number in the Request-URI (in the same manner as PSTN).

**18. proxy-mode**—SIP only. Indicate the proxy mode to use when a SIP request arrives from this session agent.

If this field is empty (upon initial runtime or upgrade), it's value is set to the value of the SIP configuration's proxy mode by default. If no proxy mode value was entered for the SIP configuration, the default for this field is **proxy**.

The following are valid proxy modes:

- **proxy**—If the Oracle Communications Session Border Controller is a Session Router, the system will proxy the request coming from the session agent and maintain the session and dialog state. If the Oracle Communications Session Border Controller is a Session Director, the system behaves as a B2BUA when forwarding the request.
- **redirect**—The system sends a SIP 3xx reDIRECT response with contacts (found in the local policy) to the previous hop.

**19. redirect-action**—SIP only. Indicate the action you want the SIP proxy to take when it receives a Redirect (3XX) response from the session agent.

If the response comes from a session agent and this field is empty (upon initial runtime or upgrade), the redirect action will be recurse. If no session agent is found (for example, if a message comes from an anonymous user agent), the redirect action is set to proxy. If the Redirect (3xx) response does not have any Contact header, the response will be sent back to the previous hop.

The following table lists the available proxy actions along with a brief description

- **proxy**—The SIP proxy passes the response back to the previous hop; based on the proxy mode of the original request.
- **recurse**—The SIP proxy serially sends the original request to the list of contacts in the Contact header of the response (in the order in which the contacts are listed in the response). For example, if the first one fails, the request will be sent to the second, and so on until the request succeeds or the last contact in the Contact header has been tried.

**20. loose-routing**—SIP only. Enable this parameter if you want to use loose routing (as opposed to strict routing). The default is **enabled**. Valid values are:

- enabled | disabled

When the SIP NAT route home proxy parameter is enabled, the Oracle Communications Session Border Controller looks for a session agent that matches the home proxy address and checks the loose routing value. If loose routing is enabled, a Route header is included in the outgoing request in accordance with RFC 3261. If loose routing is disabled, the Route header is not included in the outgoing request (in accordance with strict routing procedures defined in RFC 2543).

The loose routing value is also checked when the local policy's next hop value matches a session agent. If loose routing is set to enabled, the outgoing request retains the original Request-URI and Route header with the next hop address.

**21. send-media-session**—SIP only. Enable this parameter if you want to include a media session description (for example, SDP) in the INVITE or REINVITE message sent by the Oracle Communications Session Border Controller. Setting this field to **disabled** prevents the Oracle Communications Session Border Controller from establishing flows for that INVITE message.

The default is **enabled**. Valid values are:

- enabled | disabled



**Note:** Only set send media session to disabled for a session agent that always redirects requests. It returns an error or 3xx response instead of forwarding an INVITE message. In addition, do not disable

send media session on session agents that support SIP-to-H.323 IWF call flows. This can cause call failure.

**22. response-map**—Optional and for SIP only. Enter the name of the response map to use for this session agent. The mappings in each SIP response map is associated with a corresponding session agent. You can also configure this value for individual SIP interfaces.

**23. ping-method**—SIP only. Indicate the SIP message/method to use to ping a session agent. The ping confirms whether the session agent is in service. If this field is left empty, no session agent will be pinged.

Setting this field value to the OPTIONS method might produce a lengthy response from certain session agents and could potentially cause performance degradation on your Oracle Communications Session Border Controller.

**24. ping-interval**—SIP only. Indicate how often you want to ping a session agent by configuring the ping interval parameter. Enter the number of seconds you want the Oracle Communications Session Border Controller to wait between pings to this session agent. The default value is 0. The valid range is:

- Minimum: 0
- Maximum: 999999999

The Oracle Communications Session Border Controller only sends the ping if no SIP transactions (have occurred to/from the session agent within the time period you enter here.

**25. trunk-group**—Enter up to 500 trunk groups to use with this single session agent. Because of the high number of trunk groups you can enter, the ACLI provides enhanced editing mechanisms for this parameter:

- You use a plus sign (+) to add single or multiple trunk groups to the session agent's list.

When you add a single trunk group, simply use the plus sign (+) in front of the trunk group name and context. Do not use a Space between the plus sign and the trunk group name and context.

For example, you might have already configured a list of trunk groups with the following entries:

**tgrpA:contextA**, **tgrpB:contextB**, and **tgrpC:contextC**. To add **tgrp1:context1**, you would make the following entry:

```
ACMEPACKET(session-agent) # trunk-group +tgrp1:context1
```

Your list would then contain all four trunk groups.

When you add multiple trunk groups, simply enclose your entry in quotation marks (") or in parentheses (()). While you put spaces between the trunk group name and context entries, you do not use spaces with the plus sign, parentheses or quotation marks.

```
ACMEPACKET(session-agent) # trunk-group +tgrp1:context1 tgrp2:context2
tgrp3:context3
```

- You use a minus sign (-) to delete single or multiple trunk groups from the session agent's list.

When you remove a single trunk group, simply use the minus sign (-) in front of the trunk group name and context. Do not use a Space between the minus sign and the trunk group name and context.

For example, you might have already configured a list of trunk groups with the following entries:

**tgrpA:contextA**, **tgrpB:contextB**, **tgrpC:contextC**, and **tgrp1:context1**. To delete **tgrp1:context1** from the list, you would make the following entry:

```
ACMEPACKET(session-agent) # trunk-group -tgrp1:context1
```

Your list would then contain: **tgrpA:contextA**, **tgrpB:contextB**, and **tgrpC:contextC**.

When you add multiple trunk groups, simple enclose your entry in quotation marks (") or in parentheses (()). While you put spaces between the trunk group name and context entries, you do not use spaces with the plus sign, parentheses or quotation marks.

```
ACMEPACKET(session-agent) # trunk-group -tgrp1:context1 tgrp2:context2
```

- You overwrite (replace) the entire list of a session agent's trunk groups by entering a list that does not use either the plus (+) or the minus (-) sign syntax.

**26. ping-in-service-response-codes**—SIP only. Enter the list of response codes that keep a session agent in service when they appear in its response to the Oracle Communications Session Border Controller's ping

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request. The Oracle Communications Session Border Controller takes the session agent out of service should a response code be used that does not appear on this list. Default is **none**.

**27. out-service-response-codes**—SIP only. Enter the list defines the response codes that take a session agent out of service when they appear in its response to the Oracle Communications Session Border Controller's ping request or any in-dialog creating request (such as an INVITE, SUBSCRIBE, etc.). The Oracle Communications Session Border Controller ignores this list if an in-service list exists.

**28. options**—Optional. You can add your own features and/or parameters by using the options parameter. You enter a comma-separated list of either or both of the following:

- feature=<value feature>

For example:

You can include the original address in the SIP message from the Oracle Communications Session Border Controller to the proxy in the Via header parameter by entering the following option:

```
via-origin=<parameter-name>
```

The original parameter is included in the Via of the requests sent to the session agent. The via origin feature can take a value that is the parameter name to include in the Via. If the value is not specified for via origin, the parameter name is origin.



**Note:** If the feature value itself is a comma-separated list, enclose it within quotation marks.

**29. media-profiles**—Optional and for H.323 only. You can enter a list of media profiles to open logical channels when starting an outgoing call as a Fast Start H.323 call.

Values you enter here must start with either an alphabetical character from A through Z (A-Za-z) or with an underscore (\_). After the first character, each list entry can contain any combination of alphabetical or numerical characters (0-9A-Za-z), as well as the period (.), the dash (-), and the underscore (\_). For example, netnet\_mediaprofile1.

You can enter 1 to 24 characters.



**Note:** The values you enter here must correspond to a valid name you entered when you configure the media profile.

**30. in-translationid**—Optional. Enter the In Translation ID for a configured session translation (group of address translation rules with a single ID) if you want to apply session translation to incoming traffic.

**31. out-translationid**—Optional. Enter the Out Translation ID for a configured session translation (group of address translation rules with a single ID) if you want to apply session translation to outgoing traffic.

Address translations attached to session agents take precedence over address translations attached to realms. If no address translation is applied to a session agent, then the Oracle Communications Session Border Controller will use the address translation applied to a realm. If an address translation is applied to both a realm and session agent, the translation attached to the session agent will apply. If the applicable session agent and realm have no associated translations, then the addresses will remain in their original forms and no address translations will be performed.

**32. trust-me**—Indicate whether this session agent is a trusted source, which the Oracle Communications Session Border Controller checks when it receives a message to determine if the source is trusted. The default value is **disabled**. The valid values are:

- enabled | disabled

The following example shows a session agent with an IP address used for the hostname.

```
session-agent
  hostname          192.168.1.10
  ip-address        192.168.1.10
  port              5060
  state             enabled
  app-protocol      SIP
  app-type
```



transport-method		UDP	
realm-id		realm-1	
description		englab	
carriers			
		carrier1	
allow-next-hop-lp		enabled	
constraints		disabled	
max-sessions		355	
max-inbound-sessions	4		
max-outbound-sessions		355	
max-burst-rate		0	
max-inbound-burst-rate		10	
max-outbound-burst-rate		1	
max-sustain-rate		3000	
max-inbound-sustain-rate		0	
max-outbound-sustain-rate		0	
min-seizures		5	
min-asr		0	time-to-resume 60
ttr-no-response		0	
in-service-period		30	
burst-rate-window		60	
sustain-rate-window		3600	
req-uri-carrier-mode		None	
proxy-mode		Proxy	
redirect-action		Recurse	
loose-routing		enabled	
send-media-session		enabled	
response-map			
ping-method			
ping-interval		0	
media-profiles			
in-translationid			
out-translationid			
trust-me		disabled	
request-uri-headers			
stop-recurse			
local-response-map			
ping-to-user-part			
ping-from-user-part			
li-trust-me		disabled	
in-manipulationid			
out-manipulationid			
p-asserted-id			
trunk-group			
max-register-sustain-rate		0	

### Session Agent Group Configuration

To configure session agent groups:

1. Access the **session-agent-group** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# session-group
ACMEPACKET(session-agent-group)#
```

2. **group-name**—Enter a unique name for the session agent group in Name format.
3. **description**—Optional. Enter descriptive information about the session agent group.
4. **state**—Enable or disable the session agent group on the Oracle Communications Session Border Controller. The default value is **enabled**. Valid values are:
  - enabled | disabled

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---

5. **application-protocol**—Indicate the signaling protocol you want to use with the session agent group. The default value is **SIP**. The valid values are:
  - SIP | H.323
6. **strategy**—Indicate the session agent allocation strategy you want to use. The strategy you chose selects the session agents that will be made available by this session agent group. The default value is **hunt**. The valid values are:
  - **hunt**—Selects session agents in the order in which they are listed. For example, if the first agent is online, working, and has not exceeded defined constraints, all traffic is sent to the first agent. If the first agent is offline or if it exceeds a defined constraint, the second agent is selected. If the first and second agents are offline or exceed defined constraints, the third agent is selected. And so on through the list of session agents.
  - **roundrobin**—Selects each session agent in the order in which they are listed in the destination list, selecting each agent in turn, one per session.
  - **leastbusy**—Selects the session agent that has the fewest number of sessions relative to the maximum sessions constraint (for example, lowest percent busy) of the session agent element. The Least Busy Calculation is the result of dividing the number of active calls for a session agent by the max-sessions parameter within the session-agent element configuration. If the default max-session parameter value issued for a session agent (0), the result of the Least Busy Calculation will be 0. The Least Busy SAG Strategy will route a session to the session agent with the lowest resulting Least Busy Calculation percentage. If multiple session agents have the lowest percentage, the foremost session agent in the Session Agent Group dest parameter will be used.
  - **propdist**—Based on programmed, constrained session limits, the Proportional Distribution strategy proportionally distributes the traffic among all of the available session agents. Sessions are distributed among session agents based on the max-outbound-sessions value in each session agent. The sum of max-outbound-sessions for every session-agent within a session group equates to 100% and the max-outbound-sessions value for each session-agent represents a % that total. Sessions are proportionally allocated to session agents based on their individual session agent max-outbound-sessions value, as a % of the total max-outbound-sessions for the group.
  - **lowsusrate**—The Lowest Sustained Rate strategy routes to the session agent with the lowest sustained rate of session initiations/invitations (based on observed sustained session request rate).
7. **destination**—Identify the destinations (session agents) available for use by this session agent group.

A value you enter here must be a valid IP address or hostname for a configured session agent.
8. **trunk-group**—Enter trunk group names and trunk group contexts to match in either IPTTEL or custom format. If left blank, the Oracle Communications Session Border Controller uses the trunk group in the realm for this session agent group. Multiple entries are surrounded in parentheses and separated from each other with spaces.

Entries for this list must one of the following formats: trgp:context or trgp.context.
9. **sag-recursion**—Enable this parameter if you want to use SIP SAG recursion for this SAG. The default value is **disabled**. Valid values are:
  - enabled | disabled
10. **stop-sag-recurse**—Enter the list of SIP response codes that terminate recursion within the SAG. Upon receiving one of the specified response codes, such as 401 unauthorized, or upon generating one of the specified response codes internally, such as 408 timeout, the Oracle Communications Session Border Controller returns a final response to the UAC and stops trying to route the message. This includes not attempting to contact higher-cost SAs.

You can enter the response codes as a comma-separated list or as response code ranges.
11. Type **done** to save your configuration.

### SAG Matching for LRT and ENUM

When this feature is enabled and a match is found, the Oracle Communications Session Border Controller uses the matching SAG for routing. When there is no match for the SAG, the Oracle Communications Session Border Controller processes the result as it would have if this feature had not been enabled: either matching to a session agent hostname, or performing a DNS query to resolve it.

Note that you set the state of this feature in the SIP configuration.

To configure a SAG for ENUM or LRT matching:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

If you are adding support for this feature to a pre-existing SIP configuration, then you must select (using the ACLI **select** command) that configuration to edit it.

4. **enum-sag-match**—Set this parameter to enabled so the Oracle Communications Session Border Controller will match session agent group (SAG) names with the hostname portion in the naming authority pointer (NAPTR) from an ENUM query or LRT next-hop entry. The default value is **disabled**. The valid values are:

- enabled | disabled

5. Save and activate your configuration.

### Configuring Local Policy

To configure local policy:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **local-policy** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# local-policy
ACMEPACKET(local-policy)#
```

4. **from-address**—Indicate the originating address information by entering a From address value. You can use the asterisk (\*) as a wildcard to indicate this policy can be used with all originating addresses.

You can also use complete or partial E.164 addresses (strings that contain telephone keypad characters) here. Number matching works from left to right. Formats include the following:

- SIP From address
- FQDNs
- IP addresses
- H.323 CallingPartyAddress

The Net-Net SBC also supports the asterisk as part of the From address you configure in your local policies.

This means that for the **from-address** parameters of a local policy configuration, you can enter values in which an asterisk appears and match them accordingly. You might enter values that resemble the following examples:

- 123\*456

5. **to-address**—Indicate the destination address by entering a To address value. You can use the asterisk (\*) as a wildcard to indicate all this policy can be used for any destination address.

You can also use E.164 addresses (strings that contain telephone keypad characters) here. Number matching works from left to right. Formats include the following:

- SIP Request-URI
- FQDNs
- IP addresses
- H.323 CalledPartyAddress

The Net-Net SBC also supports the asterisk as part of the To address you configure in your local policies.

This means that for the **to-address** parameters of a local policy configuration, you can enter values in which an asterisk appears and match them accordingly. You might enter values that resemble the following examples:

- 123\*456

6. **source-realm**—Enter the realm, or list of realms, you want the Net-Net SBC to use to determine how to route traffic. This list identifies from what realm traffic is coming and is used for routing by ingress realm by the local policy.

You can use the asterisk (\*) as a wildcard to indicate this local policy can be used with all realms. The default value is \*.Or you can enter a value that corresponds to the identifier of an already configured realm. Formats include the following:

- realm ID
- customer name
- peer name
- subdomain name
- VPN identifier

7. **activate-time**—Set the time you want the local policy to be activated using the following syntax:

```
yyyy:mm:dd hh:mm:ss  
yyyy:mm:dd-hh:mm:ss
```

8. **deactivate-time**—Set the time you want the local policy to be deactivated using the following syntax:

```
yyyy:mm:dd hh:mm:ss  
yyyy:mm:dd-hh:mm:ss
```

9. **state**—Indicate whether you want the local policy to be enabled or disabled on the system. The default value is **enabled**. The valid values are:

- enabled | disabled

10. **policy-attribute**—Configure local policy attributes by following steps 8 through 21.

11. **next-hop**—Identify the next signaling host by entering the next hop value. You can use the following as next hops:

- IPv4 address or IPv6 address of a specific endpoint
- Hostname or IPv4 address or IPv6 address of a configured session agent
- Group name of a configured session agent group



**Note:** The group name of a configured session agent group must be prefixed with SAG:

For example:

policy-attribute

next-hop SAG:appserver

policy-attribute

next-hop lrt:routetable

policy-attribute

next-hop enum:lerg

You can also configure a next hop that has an address of 0.0.0.0, thereby creating a null route. Different from not having a local policy configured (which would trigger Net-Net SBC local policy recursion), this terminates local policy recursion and immediately fails the request. In these cases, the Net-Net SBC responds a request with a 404 Not Found.

- 12. realm**—Identify the egress realm (the realm used to reach the next hop) if the Net-Net SBC must send requests out from a specific realm.

The value you enter here must correspond to a valid identifier you enter when you configured the realm. If you do not enter a value here, and the next hop is a session agent, the realm identified in the session agent configuration is used for egress. In H.323, the next hop address is matched against the realm's address prefix to determine the realm.

- 13. replace-uri**—Indicate whether you want to replace the Request-URI in outgoing SIP requests with the next hop value.

- 14. carrier**—Optional. Enter the name of the carrier associated with this route. The value you enter here must match one or more of the carrier names in the session agent configuration.

Entries in carrier fields can be from 1 to 24 characters in length and can consist of any alphabetical character (Aa-Zz), numerical character (0-9), or punctuation mark (! " # \$ % ^ & \* ( ) + - = < > ? ' | { } [ ] @ / \ ' ~ , . \_ : ; ) or any combination of alphabetical characters, numerical characters, or punctuation marks. For example, both 1-0288 and acme\_carrier are valid carrier field formats.

- 15. start-time**—Indicate the time of day (from the exact minute specified) the local policy attributes go into effect. Enter only numerical characters (0-9) and follow the 4-digit military time format. For example:

```
1400
```

The default value of **0000** implies that the defined policy attributes can be considered in effect any time after 00:00:00. The valid range is:

- Minimum—0000
- Maximum—2400

- 16. end-time**—Indicate the time of day (from the exact minute specified) the local policy attributes are no longer in effect. Enter only numerical characters (0-9) and follow the 4-digit military time format. For example:

```
2400
```

The default value of **2400** implies that the defined policy attributes can be considered in effect any time before midnight. The valid range is:

- Minimum—0000
- Maximum—2400

- 17. days-of-week**—Enter any combination of days of the week (plus holidays) you want the local policy attributes to be in effect. You must enter at least one day or holiday here. A holiday entry must correspond with a configured holiday established in the Session Router.

The default is **U-S**. The valid values are:

- U (Sunday)
- M (Monday)
- T (Tuesday)
- W (Wednesday)
- R (Thursday)
- F (Friday)
- S (Saturday)
- H (Holiday)

You can enter a range of values separated by a hyphen, for example U-S. And you can enter multiple values separated by commas, for example M,W,F. You cannot use spaces as separators.

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**18. cost**—Enter a cost value that acts as a unitless representation of the cost of a route relative to other routes reaching the same destination (To address). This value is used as a way of ranking policy attributes.

The default value is zero (0). The valid values are:

- minimum—zero (0)
- maximum—999999999

**19. app-protocol**—Enter the signaling protocol to use when sending messages to the next hop. The valid values are:

- H.323 | SIP

**20. state**—Indicate whether you want to enable or disable the local policy. The default value is **enabled**. The valid values are:

- enabled | disabled

**21. media-profiles**—Configure a list of media profiles if you want the local policy to route SIP and H.323 traffic by the codecs specified in the SDP. The list of media profiles entered here are matched against the SDP included in SIP or H.323 requests and the next hop is selected by codec.

The values in this list are matched against the rtpmap attribute of passed SDP, and preference weight for route selection is based on the order in which the matching payload type appears in the SDP's media (m=) line.

For example when the following SDP arrives:

```
m=audio 1234 RTP/AVP 0 8 18
```

that contains the following attributes that correspond to three configured local policies with the same cost:

- a=rtpmap:0 PCMU/8000
- a=rtpmap:8 PCMA/8000
- a=rtpmap:18 G729/8000

the following route selection action occurs:

The local policy route that corresponds to the a=rtpmap:0 PCMU/8000 attribute is selected because the payload type of 0 in the attribute line matches the first payload type of 0 listed in the m= line. The codec value of PCMU indicated in this selected attribute is used to find the local policy with the media profiles attribute that includes PCMU in the list.

Because the value you enter here is matched against the codec values included in the actual passed SDP, it must correspond to accepted industry-standard codec values.

The following example shows a local policy with a next hop value of the session agent group called gw-sag2.

```
local-policy
  from-address
  to-address *
  source-realm *
  activate-time 2005-01-20 20:30:00
  deactivate-time N/A
  state enabled
  last-modified-date 2005-01-10 00:36:29
policy-attribute
  next-hop SAG:gw-sag2
  realm
  replace-uri enabled
  carrier
  start-time 0000
  end-time 2400
  days-of-week U-S
  cost 0
  app-protocol
```

```
state          enabled
media-profiles
```

### Local Policy Matching for Parent Realms

For SIP and H.323, you can configure the Oracle Communications Session Border Controller to use the parent realm for routing purposes even when the source realm for an incoming message is a child realm.

With this feature disabled (default), the Oracle Communications Session Border Controller uses the specific source realm to perform a local policy look-up. When the source realm is a child realm and any relevant local policies are configured with the parent realm, there will be no matches and the local policy look-up will fail. To avoid this issue and ensure successful look-ups, you must configure multiple local policies if you want to use a configuration with nested realms.

The Oracle Communications Session Border Controller examines the source realm to determine if it is a parent realm with any child realms when you enable this feature. If the parent, source realm does have child realms, then the Oracle Communications Session Border Controller creates local policy entries for the parent and all of its child realms. This operation is transparent and can save time during the configuration process.

It is possible, then, for a local policy look-up to match the same child realm in two ways:

- Through a match via the parent realm
- Through a direct match for a local policy configured with that specific child realm

In such a case, the child realm must have different costs for each type of match to avoid collisions.

This feature is enabled on a global basis in the session router configuration. Because it applies system-wide, all source realms will use this form of matching when enabled.

To enable local policy matching for parent realms:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-router** and press Enter.

```
ACMEPACKET(session-router)# session-router
```

```
ACMEPACKET(session-router-config)#
```

4. **match-lp-source-parent-realms**—If you want the Oracle Communications Session Border Controller to perform local policy realm matching based on the parent realm (so that there are local policy entries for parent and child realms), set this parameter to **enabled**. The default value is **disabled**. The valid values are:

- enabled | disabled

```
ACMEPACKET(session-router-config)# match-lp-src-parent-realms enabled
```

5. Save and activate your configuration.

## SIP Session Agent DNS-SRV Load Balancing

Prior to Release 6.2.0 the Oracle Communications Session Border Controller provided the ability to specify an FQDN (fully qualified domain name) for a destination session-agent. During DNS lookup the FQDN could resolve to multiple SRV (Resource Record for Servers) records. Each SRV could resolve to a single IP address via A-Record query in IMS or DNS.

With Release 6.2.0 the Oracle Communications Session Border Controller supports load balancing behavior as described in RFC 3263, Session Initiation Protocol (SIP): Locating SIP Servers.

The Oracle Communications Session Border Controller will provide a new parameter **ping-all-addresses** in session-agent configuration mode to enable internal load balancing and RFC 3263 compliance. The Oracle Communications Session Border Controller monitor the availability of the dynamically resolved IP addresses

## Session Routing and Load Balancing

---

obtained from DNS server using OPTIONS ping (ping-per-DNS entry). The ping-method and ping-interval for each resolved IP addresses will be copied from original session-agent.

Status of Session-Agent:

In Service – if any of dynamically resolved IP addresses is in service

Out of service – if all dynamically resolved IP addresses is out of service.

The default of **ping-all-addresses** is disabled, in which case the Oracle Communications Session Border Controller only pings the first available resolved IP addresses.

With status of each resolved IP addresses above, the Oracle Communications Session Border Controller will recurse through the list of these in-service IP addresses dynamically resolved from DNS server on 503 response, and stop recursion based upon a configured list of response values specified by the **stop-recurse** parameter in sip-interface configuration mode. With internal load balancing enabled in the session-agent, the Oracle Communications Session Border Controller provides the ability to select routing destinations based on SRV weights. The priority/weight algorithm is based on RFC 2782, *A DNS RR for specifying the location of services (DNS SRV)*.

The Oracle Communications Session Border Controller will provide the similar functionality as that listed above for A-records, the Oracle Communications Session Border Controller will select first available routing destinations because there is no priority/weight contained in A-records.

## Session Agent DNS-SRV Load Balancing Configuration

To configure the Oracle Communications Session Border Controller to perform Session-Agent DNS-SRV load balancing:

1. From superuser mode, use the following command sequence to access sip-config configuration mode. While in this mode, you configure SAG-based address resolution.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

2. Use the **ping-all-addresses** parameter to enable Session-Agent DNS-SRV load balancing.
3. Use **done**, **exit**, and **verify-config** to complete Session-Agent DNS-SRV load balancing configuration.

The **show agents** CLI command displays the availability of dynamically resolved IP addresses

```
ACMEPACKET# show sip agents acme.egr.com
21:46:05-51-router
Session Agent acme.egr.com(core) [In Service] NO ACTIVITY

... Statistics of ALL IPs associated with this SA ....

Destination: 192.168.200.235 In Service
Destination: 192.168.200.231 In Service
```

## Answer to Seizure Ratio-Based Routing

---

New SIP session agent constraints set a threshold for Answer to Seizure Ratio (ASR) has been implemented. ASR is considered when determining whether session agents are within their constraints to route calls (in addition to session and rate constraints).

The new session agent constraints indicate the minimum acceptable ASR value and computes the ASR while making routing decisions. ASR is calculated by taking the number of successfully answered calls and dividing by the total number of calls attempted (which are known as seizures).

If the ASR constraints are exceeded, the session agent goes out of service for a configurable period of time and all traffic is routed to a secondary route defined in the local policy (next hop with higher cost).

The two session agent constraints are:



- **minimum seizure:** determines if the session agent is within its constraints. When the first call is made to the session agent or the if calls to the session agent are not answered, the minimum seizure value is checked.

For example, if 5 seizures have been made to the session agent and none of them have been answered, the sixth time, the session agent is marked as having exceeded its constraints and the calls will not be routed to it until the time-to-resume has elapsed.

- **minimum ASR:** considered when make routing decisions. If some or all of the calls to the session agent have been answered, the minimum ASR value is considered to make the routing decisions.

For example, if the you set the minimum ASR at 50% and the session agent’s ASR for the current window falls below 50%, the session agent is marked as having exceeded its constraints and calls will not be routed to it until the time-to-resume has elapsed.

## ASR Constraints Configuration

To configure ASR constraints:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

4. If configuring an existing session agent, enter the select command to select the session agent.
5. **min-seizures**—Enter the minimum number of seizures that when exceeded, cause the session agent to be marked as having exceeded its constraints. Calls will not be routed to the session agent until the time-to-resume has elapsed. The default value is **5**. The valid range is:
  - Minimum—1
  - Maximum—999999999
6. **min-asr**—Enter the percentage you want as the minimum. If the session agent’s ASR for the current window falls below this percentage, the session agent is marked as having exceeded its constraints and calls will not be routed to it until the time-to-resume has elapsed. The default value is **0**. The valid range is:
  - Minimum—0
  - Maximum—100
7. Save and activate your configuration.

The following example shows a session agent configuration.

```
session-agent
  hostname                192.168.1.6
  ip-address
  port                    1720
  state                   enabled
  app-protocol            H323
  app-type                H323-GW
  transport-method
  realm-id                external
  description
  carriers
  allow-next-hop-lp      enabled
  constraints            enabled
  max-sessions           0
max-inbound-sessions    4
  max-outbound-sessions  5
  max-burst-rate         0
```

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```
max-inbound-burst-rate      10
max-outbound-burst-rate     1
max-sustain-rate            0
max-inbound-sustain-rate    0
max-outbound-sustain-rate   0
min-seizures                 5
min-asr                      50
time-to-resume              30
ttr-no-response             0
in-service-period           0
burst-rate-window           0
sustain-rate-window         0
req-uri-carrier-mode        None
proxy-mode
redirect-action
loose-routing                enabled
send-media-session          enabled
response-map
ping-method
ping-interval                0
media-profiles
in-translationid
out-translationid
trust-me                     disabled
request-uri-headers
stop-recurse
local-response-map
ping-to-user-part
ping-from-user-part
li-trust-me                  disabled
in-manipulationid
out-manipulationid
p-asserted-id
trunk-group
max-register-sustain-rate   0
early-media-allow
invalidate-registrations     disabled
last-modified-date          2006-05-12 19:48:06
```

## ENUM Lookup

---

Telephone Number Mapping (ENUM from Telephone Number Mapping) is a suite of protocols used to unify the telephone system with the Internet by using E.164 addresses with the Domain Name System (DNS). With ENUM, an E.164 number can be expressed as a Fully Qualified Domain Name (FQDN) in a specific Internet infrastructure domain defined for this purpose (e164.arpa). E.164 numbers are globally unique, language independent identifiers for resources on Public Switched Telecommunication Networks (PSTNs). ITU-T recommendation E.164 is the international public telecommunication telephony numbering plan.

## How ENUM Works

ENUM uses DNS-based architecture and protocols for mapping a complete international telephone number (for example, +1 202 123 1234) to a series of Uniform Resource Identifiers (URIs).

The protocol itself is defined in the document E.164 number and DNS (RFC 3761) that provides facilities to resolve E.164 telephone numbers into other resources or services on the Internet. The syntax of Uniform Resource Identifiers (URIs) is defined in RFC 2396. ENUM uses Naming Authority Pointers (NAPTR) records defined in RFC 2915 in order to identify available ways or services for contacting a specific node identified through the E.164 number.

## Translating the Telephone Number

A telephone number is translated into an Internet address using the following steps:

1. The number is first stored in the following format, +1-202-555-1234. 1 is the country code for the United States, Canada, and the seventeen other countries that make up the North American Numbering Plan (NANP). The + indicates that the number is a complete, international E.164 telephone number.
2. All characters are removed except for the digits. For example, 12025551234.
3. The order of the digits is reversed. For example, 43215552021. The telephone number is reversed because DNS reads addresses from right to left, from the most significant to the least significant character. Dots are placed between each digit. Example: 4.3.2.1.5.5.5.2.0.2.1. In DNS terms, each digit becomes a zone. Authority can be delegated to any point within the number.
4. A domain (for example, e164.arpa) is appended to the end of the numbers in order to create a FQDN. For example, 4.3.2.1.5.5.5.2.0.2.1.e164.arpa.
5. The domain name is queried for the resource records that define URIs necessary to access SIP-based VoIP.

Once the authoritative name server for that domain name is found, ENUM retrieves relevant records and uses that data to complete the call or service. For example, the number 12025551234 returns sip:my.name@bigcompany.com.

### About NAPTR Records

ENUM uses NAPTR records for URI resource records. NAPTR records are used to translate E.164 addresses to SIP addresses. An example of a NAPTR record is:

```
$ORIGIN 4.3.2.1.5.5.5.2.0.2.1.e164.arpa.  
IN NAPTR 100 10 "u" "sip+E2U" "!^.*$!sip:phoneme@example.net!"
```

This example specifies that if you want to use the "sip+E2U" service, you should use sip:phoneme@example.net as the address.

The regular expression can be used by a telephone company to easily assign addresses to all of its clients. For example, if your number is +15554242, your SIP address is sip:4242@555telco.example.net; if your number is +15551234, your SIP address is sip:1234@555telco.example.net.

## About the Oracle Communications Session Border Controller ENUM Functionality

The ENUM functionality lets the Oracle Communications Session Border Controller make an ENUM query for a SIP request. The ENUM lookup capability lets the Oracle Communications Session Border Controller transform E.164 numbers to URIs during the process of routing (or redirecting) a call. During the routing of a SIP call, the Oracle Communications Session Border Controller uses a local policy attribute to determine if an ENUM query is required and if so which ENUM server(s) need to be queried. A successful ENUM query results in a URI that is used to continue routing or redirecting the call.

### Configurable Lookup Length

You can configure a lookup length in the ENUM configuration that provides for more efficient caching of URI lookup results; in it, you can specify the length of the string for the DNS request starting from the most significant digit. This provides more flexibility for length matching, which is useful given the amount of wild card matching available in ENUM services. Specific ENUM groups might only be intended to provide NPANXX or wild card results.

### UDP Datagram Support for DNS NAPTR Responses

The Oracle Communications Session Border Controller's default behavior is to conform to the DNS standard defined in RFC 1035 Domain Names: Implementation and Specification, which sets a maximum size for UDP responses of 512 bytes. This limitation means that responses larger than 512 bytes are truncated (set with the TC, or truncation, bit). In addition, this limitation protects network and system resources because using TCP consumes an undesirable amount of both.

However, you can configure support ENUM queries that manage larger UDP DNS responses as set out in RFC 2671, Extension Mechanisms for DNS (EDNS0), enabling your Oracle Communications Session Border Controller to manage responses beyond 512 bytes. According to RFC 2671, senders can advertise their capabilities using a new resource record (OPT pseudo-RR), which contains the UDP payload size the sender can receive. When you specify a

## Session Routing and Load Balancing

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maximum response size over 512 bytes, then the Oracle Communications Session Border Controller add the OPT pseudo-RR to the ENUM query—without which the ENUM server will truncate the response.

### Custom ENUM Service Type Support

You can configure the ENUM service type that you want to use for an ENUM group. The Oracle Communications Session Border Controller has always supported E2U+sip and sip+E2U by default, and still does. With Release S-C6.1.0, however, you are also able to configure the service type to those supported in RFCs 2916 and 3721.

For example, you can now set the service type in the ENUM configuration to support E2U+sip and E2U+voicemsg:sip. When you configure customer ENUM service types on your system, however, you should note the following:

- New entries in the **service-type** parameter overwrite pre-existing values, including the default values.
- Because of the overwriting noted above, you must include the defaults (if you want them configured) when you are adding additional ENUM service type support. That is, you have to also type in E2U+sip and sip+E2U if you want them to be used in addition to the customized types you are setting.

## ENUM Failover and Query Distribution

---

### ENUM Query Distribution

The Oracle Communications Session Border Controller can intelligently distribute ENUM queries among all configured ENUM servers. By setting the enum config's **query method** parameter to round robin, the Oracle Communications Session Border Controller will cycle ENUM queries, sequentially, among all configured ENUM servers. For example, query 1 will be directed to server 1, query 2 will be directed to server 2, query 3 will be directed to server 3, and so on.

The default query method, hunt, directs all ENUM queries toward the first configured ENUM server. If the first server is unreachable, the Oracle Communications Session Border Controller directs all ENUM queries toward the next configured ENUM server, and so on.

### Failover to New enum-config

When an enum-config's configured servers are unreachable via the network, i.e., no response is received on a query, the Oracle Communications Session Border Controller can failover to a defined ENUM config that contains different enum servers to query. This failover behavior works when all servers in an enum config are unreachable, rather than when the Oracle Communications Session Border Controller receives not-found type responses.

The Oracle Communications Session Border Controller queries each ENUM server once before trying the next configured server, and then ultimately trying the servers listed in the **failover-to** enum config. If the failover-to servers also are unreachable, the Oracle Communications Session Border Controller fails the call; the failover-to behavior does not recurse among enum-configs, it only checks the first, linked enum-config.

### ENUM Server Operation States


After 5 consecutive failed attempts, an ENUM server is considered Out of Service (OOS). All subsequent queries which would be directed to the OOS servers are immediately directed to the first non-OOS server. ENUM servers return to in-service after 600 seconds. If all configured ENUM servers are OOS, the Oracle Communications Session Border Controller fails the call.

After the first failed attempt to reach an ENUM server, it is placed in a Time Out state, which it stays in for 30 seconds. Within this 30 seconds it will not be contacted when an ENUM query is made. After the 30 seconds pass, the ENUM server goes back to an in-service state.

## Server Availability Monitoring

The Oracle Communications Session Border Controller can probe an ENUM server's health by sending it a standard ENUM NAPTR query and receiving a valid answer. The query is for the phone number defined in the **health query number** parameter, which should be one that the ENUM servers can positively resolve. As long as the query succeeds, that ENUM server maintains its in-service state and is available for ENUM queries. Any lack of response, whether network based (time-outs), or application based (DNS error or not found response) is considered a query failure and the server is set to OOS and unavailable for ENUM queries.

The Oracle Communications Session Border Controller continuously checks the health of all configured ENUM servers to determine their current state and monitor for failed servers' return to service. All servers are checked for availability at the **health query interval** parameter, as defined in seconds.

 **Note:** When ENUM server availability monitoring is enabled, ENUM servers can only exist in an in-service or out-of-service states; Without the health query interval defined, server availability monitoring is disabled, and ENUM servers exist in three service states.

## ENUM Server IP Address and Port

You can configure an IP address and port for each enum server listed in the enum-servers parameter. IP address and port are specified in XXX.XXX.XXX.XXX:YYYY format with a port value range of 1024-65535. If the port number is not specified, 53 is assumed.

The Oracle Communications Session Border Controller supports IPv6 ENUM configurations in IPv6 realms. The enumservers parameter in the enum-config configuration parameter can be configured IPv6 addresses in addition to IPv4 addresses. When IPv6 Addresses are used, the realm configured in the realm-id parameter must be an IPv6 realm.

## Unapplicable SNMP Traps and Objects

When only IPv4 ENUM servers are configured, all legacy SNMP object and trap functionality remains the same. When IPv6 addressing is used for ENUM servers, these existing SNMP objects are obsoleted.

```
apSysMgmtENUMStatusChangeTrap      NOTIFICATION-TYPE
apENUMServerStatusTable             OBJECT-TYPE
```

## IPv6 ENUM SNMP Traps and Objects

New SNMP trap notifies operators of ENUM Server Status change.

```
apAppsENUMServerStatusChangeTrap    NOTIFICATION-TYPE
OBJECTS                              { apAppsENUMConfigName,
                                   apAppsENUMServerInetAddressType,
                                   apAppsENUMServerInetAddress,
                                   apAppsENUMServerStatus }
STATUS                                current
DESCRIPTION
" The trap will be generated if the reachability status of an ENUM
server changes."
::= { apAppsNotifications 1 }
```

The following objects support this trap.

```
apAppsENUMServerStatusTable          OBJECT-TYPE
SYNTAX                               SEQUENCE OF ApAppsENUMServerStatusEntry
MAX-ACCESS                            not-accessible
STATUS                                 current
DESCRIPTION
" A read-only table to hold the status of configured ENUM servers,
indexed by the name of the enum server, server address type and server IP.
Please note this table is the replacement of
apENUMServerStatusTable defined in ap-smgmt.mib, where the table was
obsoleted."
```

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```
 ::= { apAppsMIBTabularObjects 1 }
apAppsENUMServerStatusEntry OBJECT-TYPE
    SYNTAX          ApAppsENUMServerStatusEntry
    MAX-ACCESS      not-accessible
    STATUS          current
    DESCRIPTION
        "An entry designed to hold the status of a single ENUM server"
    INDEX { apAppsENUMConfigName,
            apAppsENUMServerInetAddressType,
            apAppsENUMServerInetAddress }
 ::= { apAppsENUMServerStatusTable 1 }
ApAppsENUMServerStatusEntry ::= SEQUENCE {
    apAppsENUMConfigName          DisplayString,
    apAppsENUMServerInetAddressType  InetAddressType,
    apAppsENUMServerInetAddress    InetAddress,
    apAppsENUMServerStatus        INTEGER
}
apAppsENUMConfigName OBJECT-TYPE
    SYNTAX          DisplayString
    MAX-ACCESS      read-only
    STATUS          current
    DESCRIPTION
        "The name of the enum-config element that contains this
        ENUM server."
 ::= { apAppsENUMServerStatusEntry 1 }
apAppsENUMServerInetAddressType OBJECT-TYPE
    SYNTAX          InetAddressType
    MAX-ACCESS      read-only
    STATUS          current
    DESCRIPTION
        "The IP address of this ENUM server."
 ::= { apAppsENUMServerStatusEntry 2 }
apAppsENUMServerInetAddress OBJECT-TYPE
    SYNTAX          InetAddress
    MAX-ACCESS      read-only
    STATUS          current
    DESCRIPTION
        "The IP address of this ENUM server."
 ::= { apAppsENUMServerStatusEntry 3 }
apAppsENUMServerStatus OBJECT-TYPE
    SYNTAX          INTEGER {
                        inservice(0),
                        lowerpriority(1),
                        oosunreachable(2)
                    }
    MAX-ACCESS      read-only
    STATUS          current
    DESCRIPTION
        "The status of this ENUM server."
 ::= { apAppsENUMServerStatusEntry 4 }
apAppsENUMServerStatusGroup OBJECT-GROUP
    OBJECTS {
        apAppsENUMConfigName,
        apAppsENUMServerInetAddressType,
        apAppsENUMServerInetAddress,
        apAppsENUMServerStatus
    }
    STATUS          current
    DESCRIPTION
        "Report the status of configured ENUM servers."
 ::= { apAppsObjectGroups 1 }
apAppsENUMServerNotificationsGroup NOTIFICATION-GROUP
    NOTIFICATIONS {
        apAppsENUMServerStatusChangeTrap
    }
```

```

    }
    STATUS          current
    DESCRIPTION
    "A collection of traps to extend reporting capabilities."
 ::= { apAppsNotificationGroups 1 }

```

## Caching ENUM Responses

As DNS responses often lead to further DNS queries, a DNS server can send additional multiple records in a response to attempt to anticipate the need for additional queries. The Oracle Communications Session Border Controller can locally cache additional NAPTR, SRV, and A records returned from an ENUM query to eliminate the need for unnecessary external DNS requests by enabling the **cache addl records** parameter. These cached records can then be accessed by internal ENUM and DNS agents.

The unprompted NAPTR, SRV, or A record returned to the Oracle Communications Session Border Controller must include complete information to resolve a call to be added to the local DNS/ENUM cache, otherwise the Oracle Communications Session Border Controller will preform an external query to find the address it is looking to resolve.

Cached entries are per ENUM config. That means if one ENUM config has a number of cached entries, and an ENUM request is directed through a different ENUM config, the second configuration is not privy to what the first configuration has cached.

The Oracle Communications Session Border Controller uses the shorter lifetime of the DNS response's TTL or the server dns attribute's transaction-timeout to determine when to purge a DNS record from the local cache.

## Source URI Information in ENUM Requests

ENUM queries can be configured to include the source URI which caused the ENUM request by enabling the **include source info** parameter. The Oracle Communications Session Border Controller can add the P-Asserted-ID URI (only if not in an INVITE) or the From URI into an OPT-RR Additional Record to be sent to the ENUM server. It can be useful to specify the originating SIP or TEL URI from a SIP request which triggered the ENUM query, so the ENUM server can provide a customized response based on the caller.

This feature implements the functionality described in the Internet Draft, DNS Extension for ENUM Source-URI, draft-kaplan-enum-source-uri-00.

When a P-Asserted-ID is blocked or removed before the ENUM query is made, the Oracle Communications Session Border Controller only sends the URI in the From header.

Note that to support this feature, according to the Internet draft, ENUM clients must support 1220 bytes in UDP responses. Therefore, if this feature is enabled, and the max response size parameter is not set i.e., with a 512 byte default, the Oracle Communications Session Border Controller will set the size to 1200 on the OPT-RR records sent.

## Operation Modes

There are four modes of ENUM operation that are selected on a global basis:

- stateless proxy
- transaction stateful proxy
- session stateful proxy
- B2BUA with or without media

### Stateless Proxy Mode

The stateless proxy mode is the most basic form of SIP operation. The stateless proxy mode:

- Has the least number of messages per call. No record route header is added and there are no 100 Trying or BYEs.
- Does not keep transaction state (timers and retransmission). There are no session counters and no session stop time. No session stop time means no RADIUS STOP records.
- Has no limits on session state.

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- Can restrict functionality by specification. This can mean no media management, limited potential for RADIUS accounting, and no CALEA (no Release/BYE messages for CDC).
- Acts primarily as a routing device, with local policy routing and ENUM routing.

### Transaction Stateful Proxy

In the transaction stateful proxy mode:

- Adds state to the proxy (not dialogs).
- Has lower number of messages per call. No Record Route header added and no BYES.
- Keeps transaction state (timers and retransmissions).
- Enforces session restrictions (32k) because of state management. These restrictions can be increased.
- Can restrict functionality by specification. This can mean no media management, limited potential for RADIUS accounting, and no CALEA (no Release/BYE message for CDC).
- Acts as routing device with transaction timers, with local policy routing and ENUM routing.
- Can off-load some transactions across unreliable links.

### Session Stateful Proxy

The session stateful proxy mode:

- Maintains dialog state as a proxy.
- Includes BYES (though cannot be inserted)
- Keeps transaction state (timers and retransmission)
- Provides per-session information such as session counters per session agent, RADIUS STOP accounting record generation, CALEA CDC generation.
- Enforces session restrictions (32k) because of state management.
- Does not provide media management. There is no CALEA CCC.
- Routes full sessions with transaction timers with local policy routing and ENUM routing.

### B2BUA

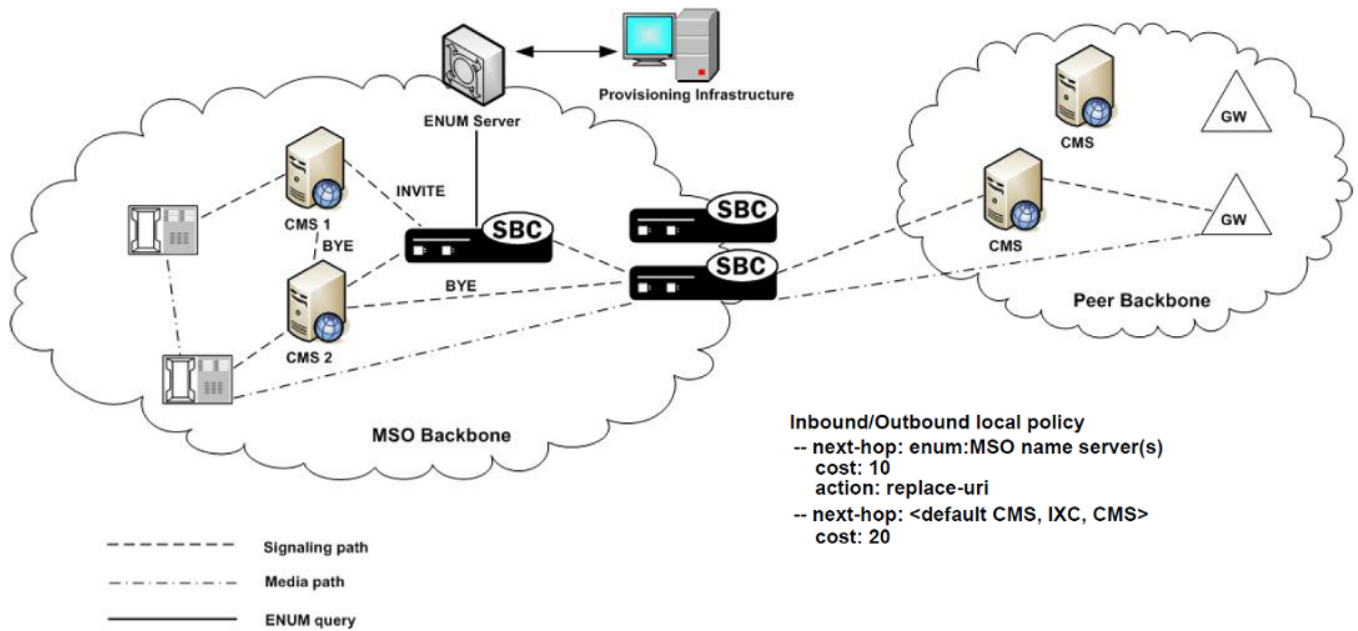
The B2BUA mode:

- Acts as UAS and UAC within call flow.
- Includes BYES (can be inserted).
- Keeps transaction state (timers and retransmissions)
- Provides per-session information such as session counters per session agent, RADIUS STOP accounting record generation, CALEA CDC generation.
- Enforces session restrictions (32k) because of state management.
- Can provide media management, including media routing through a single IP address with topology masking, CALEA CCC, media watchdogs for state management.
- Routes full sessions with topology masking. Includes rewriting Via, Route, Contact headers, full NATing with SIP NAT or header manipulation, direct bridging, local policy routing, and ENUM routing.

### Example ENUM Stateless Proxy

The following diagram shows the Oracle Communications Session Border Controller using ENUM to query a local subscriber database. The Oracle Communications Session Border Controller serves as the inbound and outbound routing hub and performs media management. Calls are routed throughout the MSO network using ENUM lookup results.





## ENUM Configuration

This section shows you how to configure ENUM on your Oracle Communications Session Border Controller.

To configure ENUM:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **enum-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # enum-config
ACMEPACKET (enum-config) #
```

4. **name**—Enter a string that uniquely identifies this ENUM configuration. You use this name in other areas of the Oracle Communications Session Border Controller configuration to refer to this ENUM configuration. For example, in the local policy attributes.

5. **top-level-domain**—Enter the domain extension to be used when querying the ENUM servers for this configuration. For example, e164.arpa. The query name is a concatenation of the number and the domain.

For example the number is +17813334444 and the domain is e164.arpa, the query name would be 4.4.4.4.3.3.1.8.7.1.e164.arpa.com.

6. **realm-id**—Enter the realm where the ENUM servers can be reached. The realm ID is used to determine on which network interface to issue the ENUM query.

7. **enum-servers**—Enter the list of ENUM servers (an ENUM server and corresponding redundant servers) to be queried. Separate each server address with a space and enclose list within parentheses.

The first server on this list is the first one to be queried. If the query times out (including retransmissions) without getting a response, the next server on the list is queried and so on.

8. **service-type**—Enter the ENUM service types you want supported in this ENUM configuration. Possible entries are E2U+sip and sip+E2U (the default), and the types outlines in RFCs 2916 and 3721.

This parameter defaults to the following service types: E2U+sip and sip+E2U.

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You can enter multiple services types in the same entry, as in this example:

```
ACMEPACKET (enum-config) # service-type E2U+sip,sip+E2U,E2U+voicemail
```

**9. query-method**—Set the strategy the Oracle Communications Session Border Controller uses to contact ENUM servers. Valid values are:

- **hunt**—Directs all ENUM queries toward the first configured ENUM server. If the first server is unreachable, the Oracle Communications Session Border Controller directs all ENUM queries toward the next configured ENUM server, and so on.
- **round-robin**—Cycles all ENUM queries, sequentially, among all configured in-service ENUM servers. Query 1 will be directed to server 1, query 2 will be directed to server 2, query 3 will be directed to server 3.

**10. timeout**—Enter the total time in seconds that should elapse before a query sent to a server (and its retransmissions) will timeout. If the first query times out, the next server is queried and the same timeout is applied. This process continues until all the servers in the list have timed out or until one of the servers responds.

The retransmission of ENUM queries is controlled by three timers. These timers are derived from this timeout value and from underlying logic that the minimum allowed retransmission interval should be 250 milliseconds; and that the Oracle Communications Session Border Controller should retransmit 3 times before timing out to give the server a chance to respond. The valid values are:

- **Init-timer**—Is the initial retransmission interval. If a response to a query is not received within this interval, the query is retransmitted. To safeguard from performance degradation, the minimum value allowed for this timer is 250 milliseconds.
- **Max-timer**—Is the maximum retransmission interval. The interval is doubled after every retransmission. If the resulting retransmission interval is greater than the value of max-timer, it is set to the max-timer value.
- **Expire-timer**—Is the query expiration timer. If a response is not received for a query and its retransmissions within this interval, the server will be considered non-responsive and the next server in the list will be tried.

The following examples show different timeout values and the corresponding timers derived from them.

timeout >= 3 seconds

```
Init-timer = Timeout/11  
Max-Timer = 4 * Init-timer  
Expire-Timer = Timeout
```

**timeout = 1 second**

```
Init-Timer = 250 ms  
Max-Timer = 250 ms  
Expire-Timer = 1 sec
```

**timeout = 2 seconds**

```
Init-Timer = 250 ms  
Max-Timer = 650 ms  
Expire-Timer = 2sec
```

**11. cache-inactivity-timer**—Enter the time interval in seconds after which you want cache entries created by ENUM requests deleted, if inactive for this interval. If the cache entry gets a hit, the timer restarts and the algorithm is continued until the cache entry reaches its actual time to live.

Setting this value to zero disables caching. For optimal performance, set this to one hour. Rarely used cache entries are purged and frequently used entries are retained. The default value is **3600**. The valid range is:

- Minimum—0
- Maximum—999999999

**12. lookup-length**—Specify the length of the ENUM query, starting from the most significant digit. The default is 0. The valid range is:

- Minimum—1
- Maximum—255

13. **max-response-size**—Enter the maximum size in bytes for UDP datagrams in DNS NAPTR responses. This parameter takes values from 512 (default) to 65535. Although the maximum value you can set is 65535, Oracle recommends configuring values that do not exceed 4096 bytes.
14. **health-query-number**—Set this parameter to a standard ENUM NAPTR query that will consistently return a positive response from the ENUM server.
15. **health-query-interval**—Set this parameter to the number of seconds to perpetually probe ENUM servers for health.
16. **failover-to**—Set this parameter to the name of another ENUM-config which to failover to under appropriate conditions.
17. **cache-addl-records**—Set this parameter to **enabled** for the Oracle Communications Session Border Controller to add additional records received in an ENUM query to the local DNS cache.
18. **include-source-info**—Set this parameter to **enabled** for the Oracle Communications Session Border Controller to send source URI information to the ENUM server with any ENUM queries.
19. Save your work.

### Example


The following example shows an ENUM configuration called enumconfig.

```
enum-config
  name                enumconfig
  top-level-domain
  realm-id            public
  enum-servers        10.10.10.10:3456
                    10.10.10.11
  service-type        E2U+sip,sip+E2U
  query-method        hunt
  timeout             11
  cacheInactivityTimer 3600
  max-response-size   512
  health-query-number +17813245678
  health-query-interval 0
  failover-to         enumconfig2
  cache-addl-records  enabled
  include-source-info disabled
```

## Configuring the Local Policy Attribute

You can specify that an ENUM query needs to be done for the routing of SIP calls. You do so by configuring the local policy's next-hop attribute with the name of a specific ENUM configuration, prefixed with the enum: tag. For example: enum:test

You can configure multiple next-hops with different ENUM servers or server groups (possibly with different top-level-domains). If the first ENUM server group you enter as the next hop is not available, one of the others can be used.

 **Note:** A new parameter called action has replaced the policy attribute's replace-uri parameter available prior to build 211p19.

To configure local policy:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **local-policy** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

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```
ACMEPACKET(session-router) # local-policy
ACMEPACKET(local-policy) #
```

- 4. next-hop**—Enter the name of the ENUM configuration with the prefix enum:. For example, enum:test.
- 5. action**—Set to **redirect** if you want to send a REDIRECT message back to the calling party with the information returned by ENUM in the Contact. The calling party then needs to send a REDIRECT using that information. The default value is **none**. Valid values are:
  - **none**—No specific actions requested.
  - **replace-uri**—To replace the next Request-URI with the next hop.
  - **redirect**—To send a redirect response with this next hop as contact.
- 6.** Save and activate your configuration.

### Local Policy Example

The following example shows one local policy with the next-hop configured to use enum:test and a second with the next-hope configured to use enum:test\_alternate.

```
local-policy
  from-address          *
  to-address            *
  source-realm          public
  activate-time         N/A
  deactivate-time       N/A
  state                 enabled
  last-modified-date    2006-03-09 09:18:43
  policy-attribute
    next-hop            enum:test
    realm               public
    action              none
    terminate-recursion disabled
    carrier
    start-time          0000
    end-time            2400
    days-of-week        U-S
    cost                 1
    app-protocol        SIP
    state               enabled
  media-profiles
  policy-attribute
    next-hop            enum:test_alternate
    realm               public
    action              none
    terminate-recursion disabled
    carrier
    start-time          0000
    end-time            2400
    days-of-week        U-S
    cost                 2
    app-protocol        SIP
    state               enabled
```

## CNAM Subtype Support for ENUM Queries

CNAM, calling name, data is a string up to 15 ASCII characters of information associated with a specific calling party name. The *Internet-draft, draft-ietf-enum-cnam-08.txt*, registers the Enumservice 'pstndata' and subtype 'cnam' using the URI scheme 'pstndata:' to specify the return of CNAM data in ENUM responses. The Oracle Communications Session Border Controller recognizes CNAM data returned via this mechanism. CNAM data is then inserted into the display name of the From: header in the original Request. If a P-Asserted-ID header is present in the original request, the CNAM data is inserted there as well.

CNAM data is identified by an ENUM response with service-type: E2U+pstndata:cnam

CNAM support is configured in the sip profile configuration element, which can then be applied to either a session agent, realm, or SIP interface.

The Oracle Communications Session Border Controller can preform CNAM queries on the signaling message's ingress or egress from the system by setting the cnam lookup direction parameter to either ingress or egress. If the CNAM lookup direction parameters are configured on both the ingress and egress sides of a call, the Oracle Communications Session Border Controller will only preform the lookup on the ingress side of the call.

## CNAM Unavailable Response

A CNAM response can include a Calling Name Privacy Indicator parameter ('unavailable=p') or Calling Name Status Indicator parameter ('unavailable=u') in responses. The Oracle Communications Session Border Controller can insert a custom reason string into the SIP message's From and P-Asserted-ID header in the original requires.

Configuring the **cnam unavailable ptype** parameter inserts the specified text into the From and P-Asserted-ID headers when a CNAM response contains the unavailable=p parameter.

Configuring the **cnam unavailable utype** parameter inserts the specified text into the From and P-Asserted-ID headers when a CNAM response contains the unavailable=u parameter.

## SIP Profile Inheritance

CNAM features, via the SIP Profile configuration element can be applied to session agents, realms, and SIP interfaces. The more generalized object inherits the more specific object's values. For example, if CNAM support via a SIP profile is configured on a session agent, the expected processing will override any SIP profile configuration on the downstream realm or SIP interface. Likewise, if CNAM support is unconfigured on the receiving session agent, but configured in the realm, CNAM configuration on the SIP interface will be ignored.

## CNAM Subtype Support Configuration

To enable the Oracle Communications Session Border Controller to preform CNAM subtype ENUM queries, you must configure a SIP profile with an enum-config object (that points to valid ENUM servers). The referenced enum-config configuration element lists the servers to contact for CNAM type queries (and other general ENUM server interaction parameters).

To configure CNAM subtype support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-profile
ACMEPACKET(sip-profile)#
```

4. **name**—Enter a string that uniquely identifies this SIP profile configuration. You use this name in other areas of the Oracle Communications Session Border Controller configuration to refer to this SIP profile in session agents, realms, or SIP interfaces.
5. **cnam-lookup-server**—Set this parameter to the name of an ENUM-config to that will query ENUM servers for CNAM data.
6. **cnam-lookup-dir**—Set this parameter to **ingress** or **egress** to identify where the Oracle Communications Session Border Controller performs a CNAM lookup with respect to where the call traverses the system. The default value is **egress**.

7. **cnam-unavailable-ptype**—Set this parameter to a string, no more than 15 characters, to indicate that the `unavailable=p` parameter was returned in a CNAM response.
8. **cnam-unavailable-utype**—Set this parameter to a string, no more than 15 characters, to indicate that the `unavailable=u` parameter was returned in a CNAM response.
9. Save your work.

## Local Route Tables

---


Adding onto the Oracle Communications Session Border Controller's existing routing capabilities, the Local Route Tables feature gives the Oracle Communications Session Border Controller the ability to determine next hops and map E.164 to SIP URIs locally. This ability provides extensive flexibility for routing.

This feature introduces the concept of a local route cache, which is populated by a local gzipped XML file on the Oracle Communications Session Border Controller. Each table/cache is populated from one defined XML file. For routing, the local route cache operates in a way similar to the ENUM model in that a local policy next hop specifies the local route table that the Oracle Communications Session Border Controller looks up. For example, you can configure one next hop to use one table, and another next hop to use a different one.

Similar to the ENUM model, the Oracle Communications Session Border Controller typically performs a local route table lookup using the telephone number (TN) of the SIP Request-URI. This is the user portion of the URI, and the Oracle Communications Session Border Controller ignores user parameters or non-digit characters. The local route table XML file defines the matching number and the resulting regular expression replacement value—as ENUM NAPTR entries would. The Oracle Communications Session Border Controller uses the resulting regular expression to replace the Request-URI, and it uses the hostname or IP address portion to determine the next hop. If the hostname or IP address matches a configured session agent, the request is sent to that session agent. If the Oracle Communications Session Border Controller does not find a matching session agent for the hostname/IP address, it either performs a DNS query on the hostname to determine its IP address or sends the request directly to the IP address.

When the next hop is defined as a user-parameter lookup key, such as a routing number (RN) or carrier identification code (CIC), the defined key is used for the local route table lookup.

Multiple (up to 10) next hops per LRT entry are tried in the order in which they appear in the XML file. If the chosen next hop fails (for example, because it matches an out-of-service session agent or the next hop responds with a failure response), then the Oracle Communications Session Border Controller will try the next in the ordered list.

 **Note:** Entering XML comments on the same line as LRT XML data is not currently supported.

The Oracle Communications Session Border Controller supports a new system task for local route tables, and can perform local route table lookups for SIP requests. It is also responsible for communicating the results to the SIP task. The new task processes the new local routing configuration objects.

When a SIP call is being routed, the Oracle Communications Session Border Controller uses local policy attributes to determine if a local route table lookup is required. If one is needed, it also selects which local routing configuration to use. Successful local route table lookups result in URIs that can be used to continue routing or redirecting calls.

## Local Route Table (LRT) Performance

### Capabilities

- Loads approximately 500 LRT tables during boot time
- Loads 100,000 entries per LRT file
- Loads 2,000,000 LRT entries total per system

### Constraints

- You cannot configure the Oracle Communications Session Border Controller with 500 LRT files each with 100,000 entries.

- Actual performance that affects the interaction among the three performance attributes varies with system memory and configuration.

## Local Routing Configuration

This section shows you how to:

- Set up local route configuration
- Specify that a set of local policy attributes needs to use local routing

### Setting Up a Local Routing Configuration

The local routing configuration is a new element in the CLI `session-router` path. This is where you configure a name for the local route table, the filename you want to give to the database corresponding to this table, and the prefix length (significant digits/bits) to be used for lookup.

To set up a local routing configuration:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type `local-routing-config` and press Enter.

```
ACMEPACKET(session-router)# local-routing-config
ACMEPACKET(local-routing-config)#
```

4. **name**—Enter the name (a unique identifier) for the local route table; this name is used for reference in the local policy attributes when to specify that local routing should be used. There is no default for this parameter, and it is required.
5. **file-name**—Enter the name for the file from which the database corresponding to this local route table will be created. You should use the `.gz` format, and the file should be placed in the `/code/lrt/` directory. There is no default for this parameter and it is required.
6. **prefix-length**—Enter the number of significant digits/bits to used for lookup and cache storage. The default value is `0`. The valid range is:
  - Minimum—`0`
  - Maximum—`999999999`
7. Save and activate your configuration.

Your configured local routing configuration will resemble the following sample.

```
local-routing-config
  name                lookup
  file-name           abc.xml.gz
  prefix-length       3
```

### Applying the Local Routing Configuration

Apply the local routing configuration by calling it to use in the local policy attributes. You do this by setting a flag in the `next-hop` parameter along with the name of the local routing configuration that you want to use.

To apply the local routing configuration:

1. In Superuser mode, type `configure terminal`, and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router`, and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type `local-policy`, and press Enter.

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---

```
ACMEPACKET(session-router) # local-policy
ACMEPACKET(local-policy) #
```

4. Type **policy-attributes**, and press Enter.

```
ACMEPACKET(local-policy) # policy-attributes
ACMEPACKET(local-policy-attributes) #
```

5. **next-hop**—In the **next-hop** parameter, type in **lrt:** followed directly by the name of the local routing configuration to be used. The **lrt:** tag tells the Oracle Communications Session Border Controller that a local route table will be used.

```
ACMEPACKET(local-policy-attributes) # next-hop lrt:lookup
```

6. Save and activate the configuration.

## Local Route Table Support for H.323 and IWF

Local Route Table (LRT) support for H.323 and IWF is compatible with that currently offered for SIP. LRT and ENUM provide the Oracle Communications Session Border Controller with the ability to perform routing based on ENUM queries to a DNS server or local to an onboard database.

For the LRT feature, this means that entries in the local routing table now include those prefixed with the h323: URI scheme, indicating that H.323 is the next hop protocol.

### IWF Considerations

When the system performs a local policy lookup for an incoming SIP or H.323 call and determines an ENUM/LRT server is the next hop, it queries that ENUM/LRT server. The response will include the URI scheme, indicating the next hop protocol and the hostname/IP address representing the next hop. For cases where the incoming call signaling protocol and the URI scheme of the ENUM/LRT response are the same, the call requires no interworking. The Oracle Communications Session Border Controller can simply route the egress call leg to the specified next hop.

Interworking is required when the incoming signaling protocol and the URI scheme of the ENUM/LRT response do not match. When the responses do not match, the Oracle Communications Session Border Controller interworks between SIP and H.323 to route the call to the appropriate next hop.

The Oracle Communications Session Border Controller also compares the URI scheme returned in the ENUM/LRT response to the application protocol specified in the policy attributes. If the URI scheme is SIP, but the policy attributes indicate H.323, the route is deemed invalid. The same is true for an H.323 URI scheme and SIP route.

### ENUM LRT Responses

No special configuration is required for LRT to work for H.323 and IWF calls. You can configure the system to match ENUM/LRT responses against session agent groups, and then use those SAGs for routing.

To enable matching ENUM/LRT responses for H.323 SAG routing:

1. In Superuser mode, type **configure terminal**, and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

2. Type **session-router**, and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

3. Type **h323-config**, and press Enter.

```
ACMEPACKET(session-router) # h323-config
ACMEPACKET(h323-config) #
```

4. **enum-sag-match**—Set this parameter to enabled if you want the Oracle Communications Session Border Controller to perform matching against the hostnames in ENUM/LRT lookup responses and session agent groups. If there is a match, the Oracle Communications Session Border Controller uses the matching SAG for routing. If no match is found, normal ENUM/LRT routing proceeds.



## LRT Entry Matching

When searching an LRT for a matching route, the Oracle Communications Session Border Controller can be configured with one of three match modes with the match mode parameter in the local routing config. These modes are:

- **exact**—When searching the applicable LRT, the search and table keys must be an exact match.
- **best**—The longest matching table key in the LRT is the chosen match.
- **all**—The all mode makes partial matches where the table's key value is a prefix of the lookup key. For example, a lookup in the following table with a key of 123456 returns entries 1, 3, and 4. The 'all' mode incurs a performance penalty because it performs multiple searches of the table with continually shortened lookup keys to find all matching entries. This mode also returns any exact matches too.

Entry#	Key	Result
1	1	<sip:\0@host1.example.com>
2	122	<sip:\0@host22.example.com>
3	123	<sip:\0@host3.example.com>
4	1234	<sip:\0@host4.example.com>
5	1234567	<sip:\0@host7.example.com>
6	1235	<sip:\0@host5.example.com>

## LRT Entry Matching Configuration

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **local-routing-config** and press Enter.

```
ACMEPACKET(session-router)# local-routing-config
ACMEPACKET(local-routing-config)#
```

4. **match-mode**—Set this parameter to either **best**, **all**, or leave it as **exact** which is the default. This indicates to the Oracle Communications Session Border Controller how to determine LRT lookup matches.
5. Save your work using the **done** command.

## LRT String Lookup

The Oracle Communications Session Border Controller can search an LRT for either E.164 or string table keys. This selection is on a global basis. When the string-lookup parameter is **disabled** (default) in the local routing configuration, all lookups will be E.164 type, except when:

- If `eloc-str-lookup` is enabled in a matching local policy's policy-attribute, E-CSCF procedures are applied and the resulting lookup type is 'string'.
- The Oracle Communications Session Border Controller also performs string lookups exclusively when a compound lookup key is specified.

When the lookup type is 'E.164', the lookup is skipped if the lookup key is not a valid telephone number (i.e. it must contain only digits).

### LRT String Lookup Configuration

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **local-routing-config** and press Enter.

```
ACMEPACKET (session-router) # local-routing-config  
ACMEPACKET (local-routing-config) #
```

4. **string-lookup**—Set this parameter to **enabled** for the Oracle Communications Session Border Controller to perform LRT lookups on table keys of a string data type. Leave this parameter to its default as disabled to continue using E.164 type lookups.
5. Save your work using the **done** command.

### Directed Egress Realm from LRT ENUM

---

A message can be sent into a specific egress realm identified in an ENUM query or LRT lookup. The egress realm is noted by a configurable parameter in the result URI. The Oracle Communications Session Border Controller is configured with the name of this parameter, that indicates an egress realm name, and looks for it in the returned URI.

To configure the parameter name, the egress-realm-param option is added to the sip config and/or the h323 config using the following format:

```
egress-realm-param=<name>
```

Where <name> is the parameter name to extract the egress realm name from.

When the egress realm param is defined, the ENUM and LRT results will always be checked for the presence of the URI parameter. The sip config options will apply for received SIP requests. The h323 config option will apply for received H.323 messages.

For example, if egress-realm-param=egress is added to the sip config, a matching entry in the LRT that specifies the egress realm core will look like this:

```
<route>  
<user type="E164">+17815551212</user>  
<next type="regex">!^.*$!sip:\0@core.example.com;egress=core!</next>  
</route>
```

If the URI does not contain the parameter or the parameter identifies a realm that is not configured on the system, the egress realm that is normally applicable (from local policy, SIP-NAT, or session-agent data) will be used.

### Directed Egress Realm Configuration

To add an egress parameter to look for in a sip-config:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal  
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **sip-config** and press Enter. If you are adding this feature to a pre-existing SIP configuration, you will need to select and edit it.

```
ACMEPACKET (session-router) # sip-config  
ACMEPACKET (sip-config) #
```


4. **egress-realm-param**—Configure this option with the parameter to parse for in a returned ENUM or LRT result: For example

- **options egress-realm-param=egress**

In order to append the new option to the sip-config's options list, you must prepend the new option with a plus sign. For example:

```
ACMEPACKET(sip-config) # options +egress-realm-param=egress
```

5. Save your work using the ACLI **done** command.

 **Note:** The egress-realm-param option can be configured similarly in the h323-config.

## SIP Embedded Route Header

The Oracle Communications Session Border Controller examines the ENUM and LRT lookup result for embedded Route headers. In the LRT or as returned in an ENUM query a URI including an embedded route header would look like:

```
<sip:user@example.com?Route=%3Csip:host.example.com;lr%3E>
```

Using embedded Route headers is the Oracle Communications Session Border Controller's default behavior. This can be overridden by adding the sip-config option use-embedded-route.

When the ENUM or LRT result becomes the top Route header, any embedded Route headers extracted are inserted just after that top Route (which will always be a loose route and include the "lr" URI parameter). In this case, the request will be sent to the top Route.

When the ENUM or LRT results become the Request-URI, any embedded Route headers extracted from the result are inserted before any Route headers already in the outgoing request. After that, if there are any Route headers in the outgoing request and the top Route header has an "lr" URI parameter, the request is sent to the top Route header. Otherwise, the request is sent to the Request URI.

## SIP Embedded Route Header Configuration

To set the Oracle Communications Session Border Controller's default behavior of using embedded route headers from ENUM queries or LRT lookups:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

3. Type **sip-config** and press Enter. If you are adding this feature to a pre-existing SIP configuration, you will need to select and edit it.

```
ACMEPACKET(session-router) # sip-config
ACMEPACKET(sip-config) #
```

4. **use-embedded-route**—Configure this as an option with one of the following arguments:

- **all** = use embedded routes from both ENUM and LRT results (default)
- **none** = do not use embedded routes
- **enum** = use embedded routes from ENUM results only
- **lrt** = use embedded routes from LRT results only

In order to append the new option to the sip-config's options list, you must prepend the new option with a plus sign. For example:

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---

Set the options parameter by typing **options**, a Space, the option name **use-embedded-route**, and then press Enter.

```
ACMEPACKET(sip-config)# options +use-embedded-route=none
```

5. Save your work using the ACLI **done** command.

## LRT Lookup Key Creation

---

This section describes the Oracle Communications Session Border Controller's LRT lookup key creation capability.

### Arbitrary LRT Lookup Key

In addition to the standard From, To, and P-Asserted-Identity header fields the Oracle Communications Session Border Controller can now use the values from any arbitrary SIP header as an LRT or ENUM lookup key. This is performed by prepending a dollar sign \$ by the header name whose value's userinfo portion of the URI will be used as the lookup key. For example, `key=$Refer-To` would use the userinfo portion of the URI in the Refer-To header of the request as the lookup key.

An ampersand & followed by a header name will use the whole value of the header as the lookup key. For example, `key=&X-Route-Key` would use the whole value of the X-Route-Key as the lookup key. As a shortcut, an ampersand is not required for a "hidden" header. For example, `key=@LRT-Key` would use the value of the @LRT-Key header as the lookup key.

### Hidden Headers for HMR and LRT lookup

When an LRT lookup key is more complex than just the URI's userinfo or a Tel-URI, HMR can be used to extract the data and build a special header.

By using a header name that begins with the at-sign "@" (e.g. @lrt-key), the header can be hidden and not included in outgoing SIP message, thus eliminating the need for an extra HMR rule to remove it.

Since '@' is not a valid character in a header name as defined by RFC 3261, there is no possibility of a collision between a header name defined in the future and a hidden header name beginning with @.

### Compound Key LRT Lookup

LRT lookup keys can be combinations of more than one key value. For example, `key=$FROM,$TO` would construct a compound key with the userinfo of the From URI followed by a comma followed by the userinfo of the To URI.

If the request message contained:

```
From: <sip:1234@example.com>  
To: <sip:5678@example.com>
```

The compound key to match this From/To pair is "1234,5678".

In the table lookup, the compound key is a single key value and there is no special treatment of the comma in key matching. The comma is simply an ordinary additional character that is matched like any letter or digit (i.e. the comma must appear in the LRT entry's "type" element data). For example, if the table were configured for "best" match-mode, the lookup key "1234,5678" would match a table entry of "1234,567", but it would not match a table entry of "123,5678".

## Retargeting LRT ENUM-based Requests

---

Request re-targeting is when a target or a request as indicated in the Request-URI, is replaced with a new URI.

This happens most commonly when the "home" proxy of the target user replaces the Request-URI with the registered contact of that user. For example, the original request is targeted at the Address-of-Record of bob (e.g. sip:bob@example.net). The "home" proxy for the domain of the original target, example.net, accesses the location

service/registration database to determine the registered contact(s) for the user (e.g. sip:bob@192.168.0.10). This contact was retrieved in a REGISTER request from the user's UA. The incoming request is then re-targeted to the registered contact. When `retarget-requests` is **enabled**, or the original Request-URI is the Oracle Communications Session Border Controller itself, the URI from the LRT lookup is used as the new Request-URI for the outgoing request.

When a request is routed rather than re-targeted, the Request-URI is not changed, but one or more Route headers may be inserted into the outgoing request. Sometimes a request which already contains Route headers will be routed without adding additional Route headers.

When the Oracle Communications Session Border Controller routes requests and the original Request-URI was not the Oracle Communications Session Border Controller itself, the URI from the LRT /ENUM lookup is added as the top Route: header including the "lr" parameter. The Request-URI then remains unchanged.

Whether the Oracle Communications Session Border Controller re-targets or routes a request depends on the following:

- The target (Request-URI) of the received request
- The presence of Route headers
- Local Policy Attributes,
- Registration Cache matching.

If the original target is the Oracle Communications Session Border Controller itself (i.e. the Request-URI contains the IP Address in the SIP interface the request was received on), the request is always re-targeted. When the original target is not the Oracle Communications Session Border Controller and Local Policy is applied, the request will be re-targeted when the policy attribute action parameter is **replace-uri**. The request will also be re-targeted when the policy attribute specifies an ENUM or LRT lookup.

Retargeting requests can be configured in either the ENUM or LRT config depending on the request URI retrieval method chosen.

## Re-targeting LRT ENUM-based Requests Configuration

This section shows you how to configure the Oracle Communications Session Border Controller to re-target/re-route request message when performing an LRT or an ENUM lookup.

To configure the Oracle Communications Session Border Controller to re-target or route request messages when performing an LRT lookup:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **local-routing-config** and press Enter.

```
ACMEPACKET (session-router) # local-routing-config  
ACMEPACKET (local-routing-config) #
```

4. **retarget-requests**—Leave this parameter set to **enabled** for the Oracle Communications Session Border Controller to replace the Request-URI in the outgoing request. Change this parameter to **disabled** for the Oracle Communications Session Border Controller to route the request by looking to the Route header to determine where to send the message.

5. Save your work using the **done** command.

To configure the Oracle Communications Session Border Controller to re-target or route request messages when performing an ENUM lookup:

6. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

7. Type **session-router** and press Enter.

## Session Routing and Load Balancing

---

```
ACMEPACKET (configure) # session-router
```

8. Type **local-routing-config** and press Enter.

```
ACMEPACKET (session-router) # enum-config
ACMEPACKET (enum-config) #
```

9. **retarget-requests**—Leave this parameter set to **enabled** for the Oracle Communications Session Border Controller to replace the Request-URI in the outgoing request. Change this parameter to **disabled** for the Oracle Communications Session Border Controller to route the request by looking to the Route header to determine where to send the message.
10. Save your work using the **done** command.

## Recursive ENUM Queries

---

If the Oracle Communications Session Border Controller receives an A-record in response to an ENUM query, it will reperform that ENUM query to the server received in the A-record.

If the Oracle Communications Session Border Controller receives an NS record in response to an ENUM query, it will resend the original ENUM query to the DNS server defined in the realm of the FQDN in the NS record. It will use the response to perform a subsequent ENUM query.

This behavior is configured by setting the **recursive query** parameter in the enum config to **enabled**.

## Recursive ENUM Queries Configuration

To configure the Oracle Communications Session Border Controller to query a DNS server for a hostname returned in an ENUM lookup result:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **local-routing-config** and press Enter.

```
ACMEPACKET (session-router) # enum-config
ACMEPACKET (enum-config) #
```

4. **recursive-query**—Set this parameter to **enabled** for the Oracle Communications Session Border Controller to query a DNS server for a hostname returned in an ENUM result.
5. Save your work using the **done** command.

## Multistage Local Policy Routing


---

Multistage local policy routing enables the Oracle Communications Session Border Controller to perform multiple stages of route lookups where the result from one stage is used as the lookup key for the next routing stage.

### Routing Stages

A routing stage signifies a re-evaluation of local policy based on the results of a local policy lookup. In the simplest, single stage case, the Oracle Communications Session Border Controller performs a local policy lookup on a SIP message's Request URI. The result of that local policy lookup is a next hop FQDN, IP address, ENUM lookup, or LRT lookup; that result is where the Oracle Communications Session Border Controller forwards the message. In the multistage routing model, that resultant next hop is used as the lookup key for a second local policy lookup.

The results of each stage do not erase the results of the previous stage. Thus, previous results are also possible routes to use for recursion, but the next stage results are tried first.

 **Note:** Setting a next hop to a SAG in a multistage scenario constitutes an error.

### Multi-stage Routing Source Realm

By default, the Oracle Communications Session Border Controller uses the realm within which a message was received as the source realm through all stages of a multistage local policy routing lookup. You can change this by setting the **multi-stage-src-realm-override** parameter in the session router config to enabled. Enabling this setting causes the Oracle Communications Session Border Controller to use the next-hop realm from the current local policy stage as the source realm for the next stage of the lookup. This source realm selection process also repeats for each stage of a multistage routing scenario.

## Network Applications

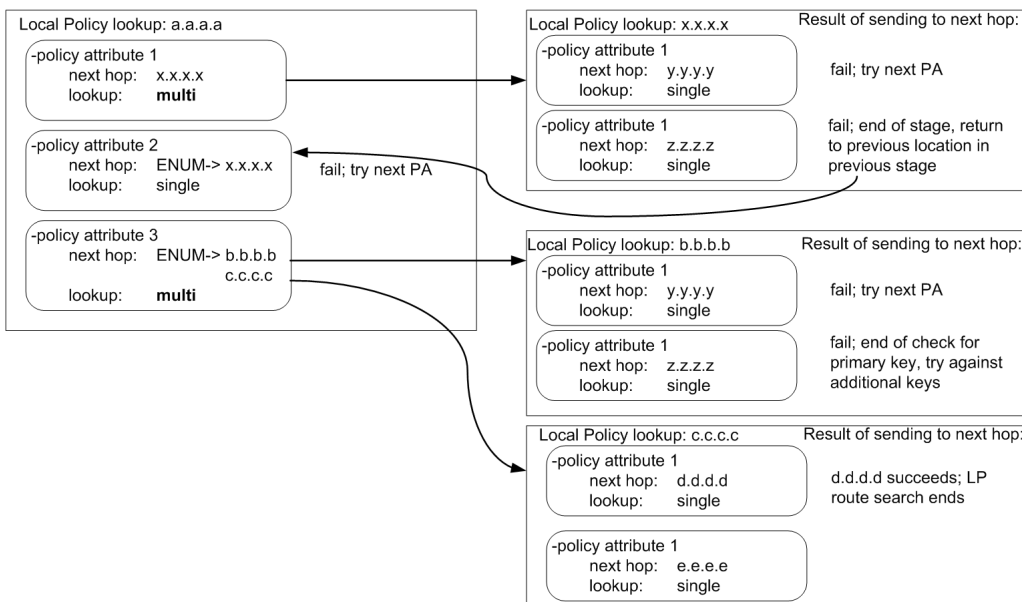
The following are typical applications of multistage routing:

- An operator might need to query an ENUM server for a destination number. Based on the NAPTR result of the ENUM query, the Oracle Communications Session Border Controller performs a local policy lookup to decide how to route the request, perhaps based on a LRT table lookup.
- An operator might need to query one ENUM server for a number portability lookup, then based on the routing number perform a second ENUM query to a different server to learn which carrier to use for the routing number. Then, then based on the identified carrier perform a LRT lookup for what next-hop(s) to use for that carrier.
- An operator might query an LRT table to confirm the allowed source number. Then, based on the result, query an ENUM server for destination routing.

## Multistage Routing Conceptual Example

Multistage routing is enabled by setting a policy attribute’s lookup parameter to multi. Instead of replacing the SIP message’s request URI with the policy attribute’s next hop address or response from an ENUM or LRT lookup, the system uses that next hop or ENUM or LRT lookup response to reconstruct the SIP message. The reconstructed SIP message is fed again through all configured local policy configuration elements (and policy attribute sub elements). Each time the Oracle Communications Session Border Controller re-evaluates a SIP message against local policies, it is considered an additional routing stage. When multiple records are returned from an ENUM or LRT lookup, the Oracle Communications Session Border Controller evaluates the first response against all applicable local policies. If unsuccessful, the Oracle Communications Session Border Controller evaluates all additional responses, in turn, against all applicable local policies.

For example:



### Multistage Routing Example 2

The following three local policy configuration elements are configured in the Oracle Communications Session Border Controller:

Local Policy 1		Local Policy 2		Local Policy 3	
from-address	*	from-address	*	from-address	*
to-address	159	to-address	192.168.1.49	to-address	215680000002
source-realm	private	source-realm	private	source-realm	private
policy-attribute		policy-attribute		policy-attribute	
next-hop	lrt:default-lrt	next-hop	lrt:carrier-lrt	next-hop	192.168.200.98
lookup	multi	lookup	multi	lookup	single
policy-attribute		policy-attribute		policy-attribute	
next-hop	192.168.200.50	next-hop	lrt:emergency	next-hop	192.168.200.97
lookup	single	lookup	single	lookup	single
				policy-attribute	
				next-hop	192.168.200.44
				lookup	multi

```
<route>
  <user type="E164">159</user>
  <next type="regex">!^.*$!sip:11568000000@192.168.200.47!</next>
  <next type="regex">!^.*$!sip:215680000002@192.168.200.99!</next>
  <next type="regex">!^.*$!sip:11578000000@192.168.200.44!</next>
</route>
```

```
INVITE sip:159@192.168.1.49:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.48:5060
From: sipp <sip:sipp@192.168.1.48:5060>;tag=1
To: sut <sip:159@192.168.1.49:5060>
Call-ID: 1-4576@192.168.1.48
CSeq: 1 INVITE
Contact: sip:sipp@192.168.1.48:5060
Max-Forwards: 70
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 135
```

The local route table in default-lrt appears as follows:

```
<route>
  <user type="E164">159</user>
  <next type="regex">!^.*$!sip:11568000000@192.168.200.47!</
next>
  <next type="regex">!^.*$!sip:215680000002@192.168.200.99!</next>
  <next type="regex">!^.*$!sip:11578000000@192.168.200.44!</next>
</route>
```

1. The Oracle Communications Session Border Controller receives an INVITE on realm, private (SDP is omitted below):

```
INVITE sip:159@192.168.1.49:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.48:5060
From: sipp <sip:sipp@192.168.1.48:5060>;tag=1
To: sut <sip:159@192.168.1.49:5060>
Call-ID: 1-4576@192.168.1.48
CSeq: 1 INVITE
Contact: sip:sipp@192.168.1.48:5060
Max-Forwards: 70
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 135
```

2. The Oracle Communications Session Border Controller performs a local policy search based on the following parameters:

```
from-address: sipp <sip:sipp@192.168.1.48:5060>;tag=1
to-address: sip:159@192.168.1.49:5060
Source Realm: private
```

3. The local policy search returns the four following routes to try:

```
lrt:default-lrt
192.168.200.50
```



```
lrt:emergency
lrt:carrier-lrt
```

The first next-hop route will be an LRT query. In addition, this policy attribute is configured with lookup=multi, meaning the results of the LRT query should be used for another local policy query, i.e., a second stage. More specifically, the request-uri that was received in response to the LRT query will be used as the to-uri in the next LP query.

The Oracle Communications Session Border Controller performs the LRT lookup in the default-lrt configuration element and is returned the following:

```
sip:11568000000@192.168.200.47
sip:215680000002@192.168.200.99
sip:11578000000@192.168.200.44
```

The Oracle Communications Session Border Controller attempts to use the results from the LRT query for the next stage Local Policy lookup(s). Beginning with the first route and continuing in sequential order, the Oracle Communications Session Border Controller will try to route the outgoing INVITE message by performing additional Local Policy lookups on the remaining LRT query results, until the INVITE is successfully forwarded.

The Oracle Communications Session Border Controller performs a local policy query on:

```
sip:11568000000@192.168.200.47
```

Which equates to a local policy lookup on:

```
from-URI=sipp <sip:sipp@192.168.1.48:5060>;
to-URI=sip:11568000000@192.168.200.47
Source Realm: private
```

The query fails because there is no Local Policy entry for 11568000000.

The Oracle Communications Session Border Controller performs a second query on request-uri

```
sip:215680000002@192.168.200.99
```

Which equates to a local policy lookup on:

```
from-URI=sipp <sip:sipp@192.168.1.48:5060>;
to-URI=sip:215680000002@192.168.200.99
Source Realm: private
```

The LP query is successful and returns the following next- hops:

```
192.168.200.98
192.168.200.99
192.168.200.44
```

The three routes shown above represent the next stage of the multistage routing for this INVITE. The policy attributes' lookup parameter is set to single for these next-hops. Therefore, the Oracle Communications Session Border Controller will attempt to send the outgoing INVITE message to one or more of these next-hops; there are no more stages to check.

**4. The Oracle Communications Session Border Controller sends an INVITE to 192.168.200.98:**

```
INVITE sip:215680000002@192.168.200.98;lr SIP/2.0
Via: SIP/2.0/UDP 192.168.200.49:5060
From: sipp <sip:sipp@192.168.1.48:5060>
To: sut <sip:159@192.168.1.49:5060>
Call-ID: SDnhae701-76e8c8b6e168958e385365657faab5cb-v3000i1
CSeq: 1 INVITE
Contact: <sip:sipp@192.168.200.49:5060;transport=udp>
Max-Forwards: 69
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 140
```

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5. If the INVITE is sent to 192.168.200.98 successfully, the local policy routing will conclude and the call will continue processing. Otherwise the Oracle Communications Session Border Controller will try the other next hops until a route succeeds or all next-hops have been exhausted

### Customizing Lookup Keys

When the **next hop** parameter points to perform an ENUM or LRT lookup, it can be provisioned with a "key=" attribute in order to specify a parameter other than the username to perform the lookup on. The following table lists the header, key value, and corresponding syntax to configure the Oracle Communications Session Border Controller with.

Username from Header:	Key Value	Example
To-URI	\$TO	key=\$TO
From-URI	\$FROM	key=\$FROM
P-Asserted-Identity	\$PAI	key=\$PAI

For a subsequent stage in multistage local policy routing, the lookup key to use for the next stage can be explicitly specified by configuring the **next key** parameter. By default, multistage lookups use the modified Request-URI returned from the ENUM/LRT response as the to-address key for the next local policy lookup. When the **next key** parameter is configured, its value will be used for the to-address key in the subsequent local policy lookup regardless if an ENUM or LRT lookup is configured for that policy attribute. The key syntax for this parameter is the same as with the Routing-based RN and CIC feature.

### Multistage Routing Lookup Termination

It is important for the Oracle Communications Session Border Controller to have a mechanism to stop performing additional stages of route lookups and limit the number of attempts and results to be tried. Routing termination can be performed at in the non-multistage way or at the global session router level.

#### Global Local Policy Termination

The Oracle Communications Session Border Controller can be configured to limit local policy lookups based several aspects of the route lookup process:

- Limiting the number of stages per message lookup—The Oracle Communications Session Border Controller can limit to the number of additional local policy lookup stages it will perform received message to a maximum of 5. This is configured with the **additional lp lookups** parameter. Leaving this parameter at its default value of 0 essentially disables multistaged local policy lookups.
- Limiting the number of routes per Local Policy lookup—The Oracle Communications Session Border Controller can limit the number of route results to use as returned for each Local-Policy lookup. This is configured with the **max lp lookups routes per lookup** parameter. Leaving this parameter at its default value of 0 places no limit on the number of returned routes the Oracle Communications Session Border Controller can try.
- Limiting the total number of routes for all local policy lookups per message request—The Oracle Communications Session Border Controller can limit the number of route returned in total across all lookups for a given request, including additional stages. This is configured with the **total lp routes** parameter. Leaving this parameter at its default value of 0 places no limit on the number of returned routes the Oracle Communications Session Border Controller can try. This parameter overrides any configured options.

Additionally, the Oracle Communications Session Border Controller monitors for local policy lookup loops which could cause a significant deterioration in performance. If a loop is found, the Oracle Communications Session Border Controller stops trying the looping route list and proceeds to try any remaining routes..

### Multistage Local Policy Routing Configuration

To set up your local policy attributes for routing using the TO header:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **local-policy** and press Enter. If you are adding this feature to a pre-existing local policy configuration, you will need to select and edit a local policy.

```
ACMEPACKET (session-router) # local-policy
ACMEPACKET (local-policy) #
```

4. Type **policy-attributes** and press Enter. If you are adding this feature to a pre-existing local policy configuration, you will need to select and edit your local policy.

```
ACMEPACKET (local-policy) # policy-attributes
ACMEPACKET (local-policy-attributes) #
```

5. **next-hop**—This is the next signaling host and/or object to query. This parameter can be configured as an IP address, ENUM server, or LRT. You can also add a lookup key to an ENUM server or LRT lookup with the following syntax:

```
next-hop enum:ENUM-object;key=$TO
```

6. **terminate-recursion**—Set this parameter to **enabled** to terminate local policy route recursion when the current stage completes.
7. **lookup**—Leave this parameter at the default **single** for single stage local policy routing or set it to **multi** to enable multistage local policy routing.
8. **next-key**—Set this parameter to \$TO, \$FROM, or \$PAI if you wish to override the recently-returned lookup key value for the next stage.
9. Save and activate your configuration.

## Maintenance and Troubleshooting

The **show sipd policy** command includes four additional counters that refer to single and multistage local policy lookups. All counters are reported for the recent period, and lifetime total and lifetime period maximum. These counters are:

- Local Policy Inits—Number of times the Oracle Communications Session Border Controller makes an initial local policy lookup.
- Local Policy Results Max—Number of times the Oracle Communications Session Border Controller truncated the number of routes returned for a local policy lookup because the maximum number of routes per local policy lookup (**max lp lookups routes per lookup**) threshold was reached.
- Local Policy Exceeded—Number of times the Oracle Communications Session Border Controller truncated the number of routes returned for a local policy lookup because the maximum number of routes per message request (**total lp routes**) threshold was reached.
- Local Policy Loops—Number of times the Oracle Communications Session Border Controller detected a loop while performing a multistage local policy lookup.

### Traps

An SNMP trap is generated to notify that the limit on the **additional lp lookups** threshold has been reached during the recent window period. This trap occurs a maximum of once during a window period.

```
apSysMgmtLPLookupExceededTrap NOTIFICATION-TYPE
STATUS current
DESCRIPTION
    " The trap will be generated the first time the additional Local
    Policy Lookups limit is reached is in the recent window period. The trap will
    only occur once during a window period."
 ::= { apSystemManagementMonitors 65}
```

### Routing-based RN and CIC

---

When the Oracle Communications Session Border Controller performs local policy routing, it selects local policy entries based on from addresses, to addresses, and source realms. All three are configurable in the local policy configuration. The to addresses can either be the username in a Request-URI (if it is an E.164/phone number format), or the request-URI's hostname or IP address. The Oracle Communications Session Border Controller sorts matching local policies based on policy attribute entries. A policy attribute defines a next hop, which can be a session agent or a session agent group. Alternatively, the next hop might define an ENUM server group or local route table to use to find the next hop.

If the routing-based RN and CIC feature is not enabled, the Oracle Communications Session Border Controller performs the subsequent ENUM query or local route table lookup using the Request-URI's username, if it is a telephone number (TN). The TN is the normalized user part of the Request-URI, ignoring any user parameters or non-digit characters.

If the routing-based RN and CIC feature is enabled, the Oracle Communications Session Border Controller instead performs the ENUM or local route table lookup based on a user parameter, which is useful for lookups based on routing number (RN) or carrier identification code (CIC):

- An RN is a number that identifies terminating switch nodes in Number Portability scenarios when the original TN has been moved to the switch defined by the RN.
- A CIC is the globally unique number of the terminating carrier to which a ported number has been moved.

In applications where the Oracle Communications Session Border Controller is given the RN or the CIC in the Request-URI, this feature is useful because the Oracle Communications Session Border Controller can perform an additional ENUM or local route table lookup to find the next hop to the RN or the CIC. Typically, ENUM servers have imported Number Portability data with which to respond to the Oracle Communications Session Border Controller query, and (for example) the Oracle Communications Session Border Controller can use local route tables for storing CIC values for direct carrier hand-off.

Even with this feature enabled, the Oracle Communications Session Border Controller still performs local policy match selection based on the TN. This feature only uses the RN or CIC user-parameter for the ENUM or local route table lookup after the local policy and policy attributes have been selected.

### Routing-based RN Configuration

This section shows you how to specify that a set of local policy attributes should use an RN for lookup. You can also set this value to CIC, or to any value you require.

You can set the lookup key to an RN in the local policy attributes' **next-hop** parameter.

To set the lookup key to RN:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure)# session-router
```

3. Type **local-policy** and press Enter.

```
ACMEPACKET (session-router)# local-policy  
ACMEPACKET (local-policy)#
```

4. Type **policy-attributes** and press Enter.

```
ACMEPACKET (local-policy)# policy-attributes  
ACMEPACKET (local-policy-attributes)#
```

5. **next-hop**—In the **next-hop** parameter—after the kind of ENUM service used—type a colon (;). Then, without spaces, type in **key=rn** and press Enter.

```
ACMEPACKET (local-policy-attributes)# next-hop lrt:lookup;key=rn
```

6. Save and activate your configuration.

## Codec Policies for SIP

The Oracle Communications Session Border Controller has the ability to add, strip, and reorder codecs for SIP sessions. This builds on the Oracle Communications Session Border Controller’s pre-existing abilities to route by codec and reorder one codec in an SDP offer by allowing you to configure the order of multiple codecs and to remove specific codecs within the media descriptions in SDP offers.

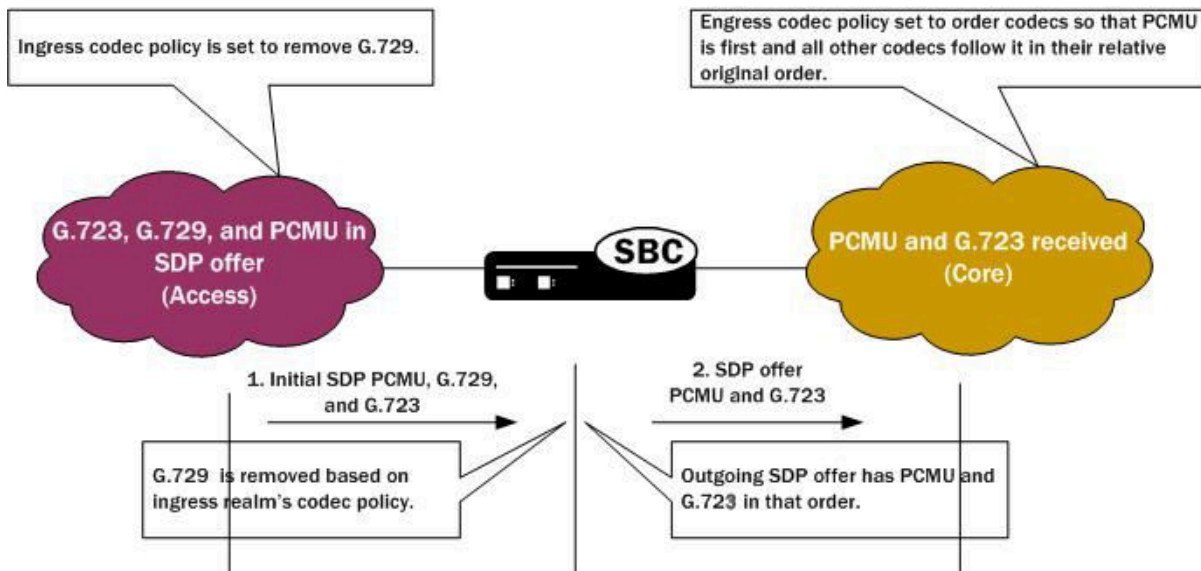
You can enable the Oracle Communications Session Border Controller to perform these operations on SDP offers by configuring codec policies. Codec policies are sets of rules that specify the manipulations to be performed on SDP offers. They are applied on an ingress and egress basis using the realm and session agent configurations.

Oracle Communications Session Border Controller supports three types of codec policies:

- Ingress policy—Codec policy that the Oracle Communications Session Border Controller applies to the SDP offer for incoming traffic
- Egress policy—Codec policy that the Oracle Communications Session Border Controller applies to the SDP offer for traffic leaving the Oracle Communications Session Border Controller
- Conditional policy—Codec policy that the Oracle Communications Session Border Controller applies to the SDP offer for traffic leaving the Oracle Communications Session Border Controller. A conditional policy differs from an egress policy in providing the capability to perform standard codec manipulations (add and strip) dynamically, based on the codec list and associated parameters contained in the original SDP offer.

The Oracle Communications Session Border Controller applies codec policies during the offer phase of media format negotiation. If codec manipulation is enabled, then the Oracle Communications Session Border Controller performs the modification according to the specific policy and forwards on the traffic.

For example, when the Oracle Communications Session Border Controller receives a SIP INVITE with SDP, it refers to the realm through which the INVITE arrived and performs any manipulations specified by an ingress codec policy that may have been assigned to the ingress realm. With the media description possibly changed according to the ingress codec policy, the Oracle Communications Session Border Controller passes the SDP offer to the outgoing realm so that the an egress codec policy can be applied. Note that the SDP to be evaluated by the egress codec policy may match the original SDP, or it may have been changed during transit through the ingress realm. After applying the egress coded policy, the Oracle Communications Session Border Controller forwards the INVITE.



Since the offer-answer exchange can occur at different stages of SIP messaging, the assigned ingress and egress roles follow the media direction rather than the signaling direction. It might be, for example, that the offer is in an OK that the Oracle Communications Session Border Controller modifies.

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You can apply codec policies to realms and to session agents; codec policies configured in session agents take precedence over those applied to realms. However, it is not required that there be both an ingress and an egress policy either for realms or for session agents. If either one is unspecified, then no modifications take place on that side. If neither ingress nor egress policies specified, SDP offers are forwarded as received.

### Relationship to Media Profiles

For each codec that you specify in a codec policy, there must be a corresponding media profile configuration on the Oracle Communications Session Border Controller. You configure media profiles in the ACLI via the session-router path. In them, you can specify codec type, transport protocol, required bandwidth, and a number of constraints.

### Manipulation Modes

You can configure a codec policy to perform several different kinds of manipulations:

- Allow—List of codecs that are allowed for a certain codec policy; if a codec does not appear on this list, then the Oracle Communications Session Border Controller removes it. You can wildcard this list with an asterisk (\*) so that all codecs are allowed. Further, you can create exceptions to a wildcarded allow list.
  - You make an exception to the wildcarded list of codecs by entering the codec(s) that are not allowed with a **no** attribute. This tells the Oracle Communications Session Border Controller to allow all codecs except the one(s) you specify.

```
ACMEPACKET(codec-policy) # allow-codecs (* PCMA:no)
```

- You can also create exceptions to allow lists such that audio or video codecs are removed. However, when the allow list specifies the removal of all audio codecs and an INVITE arrives at the Oracle Communications Session Border Controller with only audio codecs, the Oracle Communications Session Border Controller behaves in accordance with RFC 3264. This means that the resulting SDP will contain one attribute line, with the media port for the media line set to 0. The terminating side will need to supply new SDP in its reply because the result of the manipulation is the same as an INVITE with no body.

```
ACMEPACKET(codec-policy) # allow-codecs (* audio:no)
```

- Order—List of the codecs where you specify their preferred order in the outgoing media offer. The Oracle Communications Session Border Controller arranges matching codecs according to the rule you set, and any remaining ones are added to the list in the same relative order they took in the incoming media offer. If your list specifies a codec that is not present, then the ordering proceeds as specified but skips the missing codec.

You can use an asterisk (\*) as a wildcard in this list, too. The placement of the asterisk is key, as you can see in the following examples:

- For an order rule set this way

```
ACMEPACKET(codec-policy) # order (A B C *)
```

codecs A, B, and C will be placed at the front of the codec list in the order specified; all other codecs in the offer will follow A, B, and C in the same relative order they had in the original SDP offer.

- For an order rule set this way:

```
ACMEPACKET(codec-policy) # order (* A B C)
```

codecs A, B, and C will be placed at the end of the codec list in the order specified; all other codecs in the offer will come before A, B, and C in the same relative order they had in the original SDP offer.

- For an order rule set this way

```
ACMEPACKET(codec-policy) # order (A * B C)
```

codec A will be placed at the beginning of the codec list, to be followed by all other codecs in the offer in the same relative order they had in the original SDP offer, and then B and C will end the list.

- Force—An attribute you can use in the allow list with one codec to specify that all other codecs should be stripped from the outgoing offer. You can specify multiple forced codecs in your rules.
  - If you set multiple codecs in the allow list and one of them is forced, then the outgoing offer will contain the forced codec.

- If you set multiple codecs in the allow list and the one that is forced is not present in the offer, then the Oracle Communications Session Border Controller will select a non-forced codec for the outgoing offer.

```
ACMEPACKET(codec-policy) # allow (PCMU G729:force)
```

You cannot use the force attribute with a wildcarded allow list.

- No—An attribute that allows you to strip specified codecs or codec types from a wildcarded allow list.

```
ACMEPACKET(codec-policy) # allow (* PCMA:no)
```

## In-Realm Codec Manipulation

In addition to being able to apply codec policies in realms, the realm configuration supports a setting for determining whether codec manipulation should be applied to sessions between endpoints in the same realm.

In-realm codec manipulation can be used for simple call flows that traverse two realms. If the originating and terminating realms are the same, the Oracle Communications Session Border Controller checks to see if you have enabled this capability. If you have enabled it, then the Oracle Communications Session Border Controller performs the specified manipulations. If this capability is not enabled, or if the realm's media management in realm (**mm-in-realm**) setting is disabled, then the Oracle Communications Session Border Controller does not perform codec manipulations.

For more complex calls scenarios that involve call agent or reinitiation of a call back to the same realm, the Oracle Communications Session Border Controller does not perform in-realm codec manipulation.

## Codec Policy Configuration

This section gives instructions and examples for how to configure codec policies and then apply them to realms and session agents. It also shows you how to configure settings for in-realm codec manipulation.

### Creating a Codec Policy

To create a codec policy:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the signaling-related configurations.

```
ACMEPACKET(configure) # media-manager
ACMEPACKET(media-manager) #
```

3. Type **codec-policy** and then press Enter.

```
ACMEPACKET(media-manager) # codec-policy
ACMEPACKET(codec-policy) #
```

4. **name**—Enter the unique name for the codec policy. This is the value you will use to refer to this codec policy when you apply it to realms or session agents. This parameter is required and is empty by default.
5. **allow-codecs**—Enter the list of media format types (codecs) to allow for this codec policy. In your entries, you can use the asterisk (\*) as a wildcard, the force attribute, or the no attribute so that the allow list you enter directly reflects your configuration needs. Enclose entries of multiple values in parentheses ( ( ) ).

The codecs that you enter here must have corresponding media profile configurations.

6. **add-codecs-on-egress**—Enter the codecs that the Oracle Communications Session Border Controller adds to an egress SDP offer if that codec is not already there. This parameter applies only to egress offers.

The codecs that you enter here must have corresponding media profile configurations.

**add-codecs-on-egress** can be used to construct ingress, egress, or conditional codec policies.

7. **order-codecs**—Enter the order in which you want codecs to appear in the outgoing SDP offer. Remember that you can use the asterisk (\*) as a wildcard in different positions of the order to directly reflect your configuration needs. Enclose entries of multiple values in parentheses ( ( ) ).

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The codecs that you enter here must have corresponding media profile configurations.

8. Save and activate your configuration.

Your codec policy configuration will resemble the following example:

```
codec-policy
  name          private
  allow-codecs  g723:no pcmu video:no
  order-codecs  pcmu
```

### Applying a Codec Policy to a Realm

Note that codec policies defined for session agents always take precedence over those defined for realms.

To apply a codec policy to a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
```

3. Type **realm-config** and press Enter.

```
ACMEPACKET(media-manager)# realm-config
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI **select** command) the realm that you want to edit.

4. **codec-policy**—Enter the name of the codec policy that you want to apply to this realm. By default, this parameter is empty.
5. Save and activate your configuration.

### Applying a Codec Policy to a Session Agent

Note that codec policies that are defined for session agents always take precedence over those that are defined for realms.

To apply a codec policy to a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET(session-router)# session-agent
```

If you are adding support for this feature to a pre-existing session agent, then you must select (using the ACLI **select** command) the realm that you want to edit.

4. **codec-policy**—Enter the name of the codec policy that you want to apply to this realm. By default, this parameter is empty.
5. Save and activate your configuration.

### In-Realm Codec Manipulations

To enable in-realm codec manipulations:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
```



3. Type **realm-config** and press Enter.

```
ACMEPACKET(media-manager) # realm-config
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI **select** command) the realm that you want to edit.

4. **codec-manip-in-realm**—Enter the name of the codec policy that you want to apply to this realm. The default value is **disabled**. The valid values are:
  - enabled | disabled
5. Save and activate your configuration.

## QoS Based Routing

In addition to configuring your system for routing based on certain session constraints, you can also set up routing based on QoS. QoS based routing uses the R-Factor on a per-realm basis to either cut back on the traffic allowed by a specific realm, or to shut that traffic off altogether.

To use this feature, you set up QoS constraints configurations and apply one per realm. The QoS constraints configuration allows you to set up two thresholds:

- **Major**—The major threshold sets the R-Factor limit beyond which the Oracle Communications Session Border Controller rejects a certain percentage (that you configure) of calls. That is to say, it rejects inbound calls at the rate you set with a 503 Service Unavailable status code, and rejects outbound calls if there are no alternative routes.
- **Critical**—The critical threshold, when exceeded, causes the Oracle Communications Session Border Controller to behave the same way it does when any of the session constraints (set in the session-constraints configuration) are exceeded. All inbound calls to the realm are rejected with a 503 Service Unavailable status code, and (if there is no alternate route) outbound calls are rejected, too. Until the R-Factor falls within acceptable means and the session constraint's time-to-resume value has elapsed, the realm remains in this state.

## Management

This feature is supported by MIBs and traps. Historical data recording (HDR) also supports this feature by providing the following metrics in the session realm statistics collection group:

- Average QoS RFactor (0-93)
- Maximum QoS RFactor (0-93)
- Current QoS Major Exceeded
- Total QoS Major Exceeded
- Current QoS Critical Exceeded
- Total QoS Critical Exceeded

## QoS Constraints Configuration

This section shows you how to configure a QoS constraints configuration and then how to apply it to a realm.

### Configuring QoS Constraints

Your first step to enabling QoS based routing is to set up a QoS constraints configuration. This configuration is where you enter major and critical thresholds, as well as the load reduction for the realm should the R-Factor exceed the major threshold.

To set up a QoS constraints configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

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2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **qos-constraints** and press Enter.

```
ACMEPACKET(session-router)# qos-constraints
ACMEPACKET(qos-constraints)#
```

4. **name**—Enter the name of this QoS constraints configuration. This parameter uniquely identifies the configuration, and you use this value when applying the configuration to a realm. This parameter has no default and is required.
5. **state**—Set the state of this QoS constraints configuration. The default is **enabled**, but you can set this parameter to **disabled** if you want to stop applying these constraints.
6. **major-rfactor**—Enter a numeric value between **0** (default) and **9321** to set the threshold that determines when the Oracle Communications Session Border Controller applies the call reduction rate. If you leave this parameter set to **0**, then the Oracle Communications Session Border Controller will not apply a major threshold for any realm where you apply this QoS constraints configuration.

Note that this value must be greater than that you set for the **critical-rfactor**, except when the **major-rfactor** is 0.

7. **critical-rfactor**—Enter a numeric value between **0** (default) and **9321** to set the threshold that determines when the Oracle Communications Session Border Controller rejects all inbound calls for the realm, and rejects outbound calls when there is no alternate route. If you leave this parameter set to **0**, then the Oracle Communications Session Border Controller will not apply a critical threshold for any realm where you apply this QoS constraints configuration.

Note that this value must be less than that you set for the **major-rfactor**, except when the **major-rfactor** is 0.

8. **call-load-reduction**—Enter a number from **0** (default) to **100** representing the percentage by which the Oracle Communications Session Border Controller will reduce calls to the realm if the **major-rfactor** is exceeded. If you leave this parameter set to 0, then the Oracle Communications Session Border Controller will not reduce call load for the realm—even when the major-rfactor is configured.

This is the percentage of inbound and outbound calls the Oracle Communications Session Border Controller will reject. For example, if you set this parameter to 50 and the major threshold is exceeded, then the Oracle Communications Session Border Controller rejects every other call to the realm.

9. Save and activate your configuration.

### Applying QoS Constraint to a Realm

You apply QoS constraints to realms using the **qos-constraint** parameter.

To apply a QoS constraint to a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **realm-config** and press Enter. If you adding this feature to a pre-existing realm, then you need to select and edit that realm.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. **qos-constraints**—Enter the name value from the QoS constraints configuration you want to apply to this realm.

Save and activate your configuration.



---

# Number Translation

---

## About Number Translation

---

Oracle Communications Session Border Controller number translation is used to change a layer-5 endpoint name according to prescribed rules. Number translations can be performed on both the inbound and the outbound call legs independently, before and after routing occurs. Number translation is used for SIP, H.323, and SIP/H.323 interworking configurations.

Number translation takes place twice for both H.323 and SIP calls. The first number translation is applied to the incoming leg of the call, before the outgoing route is selected. The second number translation is applied to the outgoing leg of the call after the outgoing route is selected.

Number translation can be used to strip address prefixes added by external gateways. It can also be used to add a string tag to an address in order to implement a local policy routing scheme, and then remove the tag upon egress from the Oracle Communications Session Border Controller. The most common use of number translation is to add or remove a “1” or a + from a phone number sent from or addressed to a device.

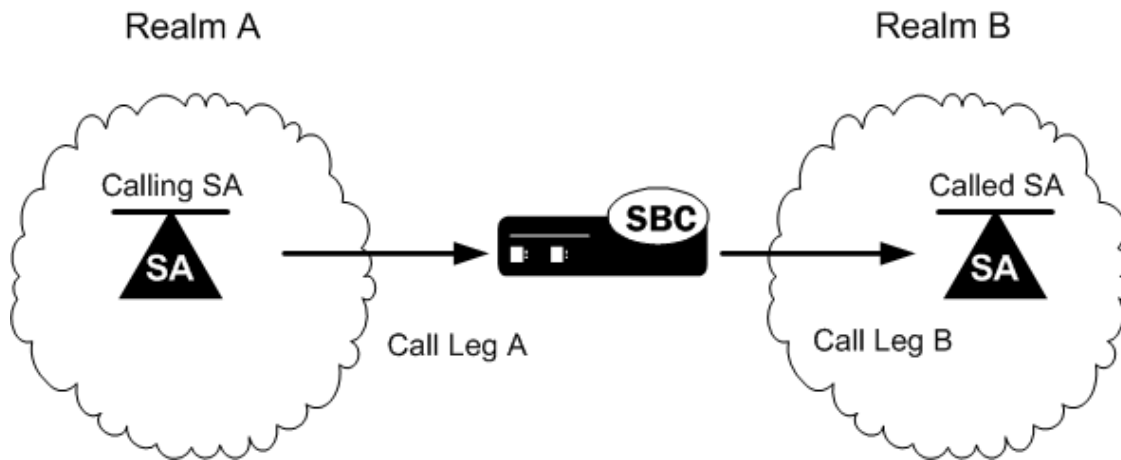
## Number Translation Implementation

Oracle Communications Session Border Controller number translations are implemented in three steps. First, the individual number translation rules are defined in the translation rules subelement. Next, the established rules are grouped in a specified order to apply to calling and called numbers. This second step occurs in the session translation element. Finally, session translations are attached to either session agents or realms in the session agent element or realm configuration element.

Number translations attached to session agents take precedence over number translations attached to realms. If no number translation is applied to a session agent, then the Oracle Communications Session Border Controller will use the number translation applied to a realm. If a number translation is applied to both a realm and session agent, the translation attached to the session agent will apply. If session agents and realms have no associated translations, then all numbers will remain in their original forms as they pass through the Oracle Communications Session Border Controller.

Within each realm or session agent, the number translation is applied to either the incoming or outgoing call leg. This distinction between incoming and outgoing calls is made from the point of view of the Oracle Communications Session Border Controller. The following diagram illustrates the number translation concept.

## Number Translation



The following table shows you which parameters to apply a session translation ID in order to affect the corresponding leg of the call as shown in the illustration.

Leg	Calling SA	Called SA	Realm A	Realm B
A	IN Translation ID		IN Translation ID	
B		OUT Translation ID		OUT Translation ID

## Number Translation in SIP URIs

Number translations only change the user portion of the URI. A typical SIP URI looks like `sip:user@hostname`. The user portion can take the form of either a phone number or any other string used for identification purposes.

Within the SIP header exists a Request URI, a To URI, and a From URI. The session translation element's rules calling parameter modifies the From URI, while the rules called parameter modifies the Request URI and the To URI.

## Number Translation Configuration Overview

Configuring the number translation feature requires the following steps:

1. Configure individual translation rules in the translation rules element.
2. Group these rules for use in the session translation element.
3. Apply these groups of rules on a per session agent or per realm basis using the appropriate fields in the session agent or realm configuration elements.

## Translation Rules

The translation rules subelement is where the actual translation rules are created. The fields within this element specify the type of translation to be performed, the addition or deletion to be made, and where in the address that change takes place. Translations are not applied to any realm or session agent in this element.

When creating translation rules, first determine the type of translation to perform. The following table lists and describes the three types of number translations.

Field Value	Description
add	This translation type adds a character or string of characters to the address.
delete	This translation type deletes a character or string of characters from the address.

Field Value	Description
replace	This translation type replaces a character or string of characters within the address. Replace works by first applying the delete parameter then by applying the add parameter.

After you set the translation type, you define the string to add or delete. The wildcard term for a string to delete is the at-sign, @. Finally, you specify the character position in the address to make the addition or deletion.

The character position where an add or delete occurs is called an index. The index starts at 0 (immediately before the leftmost character) and increases by 1 for every position to the right you move. In order to specify the final position in an address, use the dollar-sign, \$.

To create a translation rule that deletes a string:

## Translation Rules for Deleting Strings

To create a translation rule that deletes a string:

1. Enter a descriptive name for this translation in the ID field.
2. If you are deleting a specific string, enter it in the delete string field.
3. If you are deleting a portion of the address string, enter the index number in the delete index field. For this type of deletion, remember to enter the number of characters you are deleting in the form of at-signs @ in the delete string field.

The first matched string will be deleted, any remaining strings that match will remain. For example, if the address is 187521865 and the string to delete is "18," only the first instance of 18 will be deleted. The second instance will remain after translation.


## Translation Rules for Adding Strings

To create a translation rule that adds a string:

1. Enter a descriptive name for this translation in the ID field.
2. Enter the string you want to add in the add string field.
3. Enter the index of the string insertion in the add-index parameter. If you want to add a string at the end of an number, enter a dollar-sign \$ in the add index field.

## Translation Rules for Replacing Strings

To create a translation rule that replaces a string:

 **Note:** A string replacement involves deleting a string followed by adding a string in the removed string's place. The index is not used when replacing a string.

1. Enter a descriptive name for this translation in the ID field.
2. Enter the string you want to delete in the delete string field.
3. Enter the string you want to add in the add string field.

## Session Translation

A session translation defines how translation rules are applied to calling and called numbers. Multiple translation rules can be referenced and applied using this element, which groups rules together and allows them to be referenced by one identifier.

There are two parameters in the session translation element. The rules calling parameter lists the translation rules to be applied to the calling number. The rules called parameter lists of translation rules to be applied to the called number.

## Number Translation

---

The Oracle Communications Session Border Controller applies the translation rules in the order in which they are entered. They are applied cumulatively. For example, if this field is configured with a value of rule1 rule2 rule3, rule1 will be applied to the original number first, rule2 second, and rule3 last.

To configure the session translation element:

1. Enter a descriptive name for this session translation in the ID field.
2. Enter the IDs of existing translation rules in the rules calling parameter. Multiple rules can be entered in this field. The order you enter them in is the order in which they are applied.
3. Enter the IDs of existing translation rules in the rules called parameter. Multiple rules can be entered in this field. The order you enter them in is the order in which they are applied.

## Applying Session Translations

---

Session translations can be applied to both session agents and realms. Both session agents and realms contain the two parameters that denote incoming and outgoing call legs—in translation ID and out translation ID. These two fields are populated with session translation element IDs.

If none of these fields are populated, no number translation will take place and the original address will remain unchanged as it traverses the Oracle Communications Session Border Controller. Further, any session translation applied to a session agent takes precedence over one applied to a realm.

### Session Agent

To configure number translation for a session agent:

In the session agent element, set the in translation ID and/or the out translation ID to the appropriate ID you configured in the session translation element. There can be only one entry in each of these fields.

### Realm

To configure number translation for a realm:

In the realm configuration element, set the in translation ID and/or the out translation ID to the appropriate ID you configured in the session translation element. There can be only one entry in each of these fields.

## Number Translation Configuration

---

To create a translation rule:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **translation-rules** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# translation-rules  
ACMEPACKET(translation-rules)#
```

From this point, you can configure translation rules parameters. To view all translation rules parameters, enter a ? at the system prompt. The following is an example of what a translation rule configuration might look like. Parameters not described in this section are omitted below.

```
translation-rules  
    id                addplus1  
    type              add  
    add-string        +1
```



add-index	0
delete-string	
delete-index	0

## Translation Rules

Set the following parameters to configure a translation rule:

1. **ID**—Set a descriptive ID name for this translation rule.
2. **type**—Set the type of translation rule you want to configure. The default value is **none**. The valid values are:
  - **add**—Adds a character or string of characters to the address
  - **delete**—Deletes a character or string of characters from the address
  - **replace**—Replaces a character or string of characters within the address
  - **none**—Translation rule is disabled
3. **add-string**—Enter the string to be added during address translation to the original address. The value in this field should always be a real value; i.e., this field should not be populated with at-signs (@) or dollar-signs (\$). The default value is a blank string.
4. **add-index**—Enter the position, 0 being the left most position, where you want to add the string defined in the **add-string** parameter. The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—999999999
5. **delete-string**—Enter the string to be deleted from the original address during address translation. Unspecified characters are denoted by the at-sign symbol (@).

The @ character only works if the type parameter is set to delete. This parameter supports wildcard characters or digits only. For example, valid entries are: delete-string=@@@@, or delete-string=123456. An invalid entry is delete-string=123@@@. When the type is set to replace, this value is used in conjunction with the add-string value. The value specified in the delete-string field is deleted and the value specified in the add-string field is inserted. If no value is specified in the delete-string parameter and the type field is set to replace, then nothing will be inserted into the address. The default value is a blank string.

6. **delete-index**—Enter the position, 0 being the left most spot, where you want to delete the string defined in the **delete-string** parameter. This parameter is only used if the **delete-string** parameter is set to one or more at-signs. The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—999999999

## Session Translation

To configure session translations:

1. Exit out of the translation rules element and enter the session translation element.

```
ACMEPACKET(translation-rules)# exit
ACMEPACKET(session-router)# session-translation
ACMEPACKET(session-translation)#
```

From this point, you can configure the session translation element. To view all session translation parameters, enter a ? at the system prompt. The following is an example of what a session translation configuration might look like:

```
session-translation
  id                    lrules-out
  rules-calling         rule1 rule2 rule3
  rules-called          addplus1
```

2. **ID**—Set a descriptive ID name for this session translation.

## Number Translation

---

3. **rules-calling**—Enter the rules calling in the order in which they should be applied. Multiple rules should be included in quotes and separated by spaces.

```
ACMEPACKET(session-translation) # rules-calling "rule1 rule2 rule3"
```

4. **rules-called**—Enter the rules called in the order in which they should be applied. Multiple rules should be included in quotes and separated by spaces.

## Number Translation Application

To complete your number translation configuration, you must enter into a **realm-config** or **session-agent** element and assign **session-translations** there.

1. Navigate to the chosen element.

- To move from the session-translation element to the session-agent element, exit out of the session translation element and enter the session agent element.

```
ACMEPACKET(session-translation) # exit  
ACMEPACKET(session-router) # session-agent  
ACMEPACKET(session-agent) #
```

- To move from the session-translation element to the realm-config element, exit from the session translation element to the configuration path.

```
ACMEPACKET(session-translation) # exit  
ACMEPACKET(session-router) # exit
```

Navigate to the realm-config element located in the media-manager path.

```
ACMEPACKET(configure) # media-manager  
ACMEPACKET(media-manager) # realm-config  
ACMEPACKET(realm-config) #
```

2. In both **realm-config** or **session agent** elements, you must specify an **in-translationid** and/or an **out-translationid** in order to configure the number translation.

```
session-agent  
    in-translationid  
    out-translationid                lrules-out  
realm-config  
    in-translationid  
    out-translationid                lrules-out
```

Set the following parameters to configure a translation rule:

- **in-translationid**—Enter the configured session translation that you want to affect the incoming traffic in this parameter.
- **out-translationid**—Enter the configured session translation that you want to affect the outgoing traffic in this parameter.

## Other Translations

---

### FQDN Mapping

The Oracle Communications Session Border Controller maps FQDNs that appear in certain headers of incoming SIP messages to the IPv4 address that the Oracle Communications Session Border Controller inserts in outgoing SIP contact headers. The mapped FQDNs are restored in the SIP headers in messages that are sent back to the originator.

This feature is useful to carriers that use IPv4 addresses in the SIP From address to create trunk groups in a PSX for routing purposes. When the carrier's peer uses FQDNs, the carrier is forced to create trunk groups for each possible FQDN that it might receive from a given peer. Similarly, this can apply to SIP Contact and P-asserted-identity headers.

---

## Admission Control and QoS

This chapter describes how to configure the Oracle Communications Session Border Controller for call admission control and Quality of Service (QoS) monitoring. Call admission control lets you manage call traffic based on several different policies. It is aimed at managing call admission rates in the network, enabling you to maintain suitable QoS levels. A new call is admitted only if it meets the requirements

QoS reporting provides you with real-time evaluation of network and route performance. It lets you contrast internal domain and external domain performance and facilitates SLA verification and traffic engineering.

---

### About Call Admission Control

The Oracle Communications Session Border Controller provides call admission control capabilities based on the following policies:

- Bandwidth (single and multi-level policies)
- Session capacity
- Session rate (sustained and burst)



**Note:** In order to provide admission control for networks to which the Oracle Communications Session Border Controller is not directly connected, you need to define multiple realms per network interface.

### Bandwidth-Based Admission Control

The Oracle Communications Session Border Controller is a policy enforcement point for bandwidth-based call admission control. Sessions are admitted or rejected based on bandwidth policies, configured on the Oracle Communications Session Border Controller for each realm.

To manage bandwidth consumption of a network's overall capacity, you can configure aggregate bandwidth policies for each realm.

As the Oracle Communications Session Border Controller processes call requests to and from a particular realm, the bandwidth consumed for the call is decremented from the bandwidth pool for that realm. The Oracle Communications Session Border Controller determines the required bandwidth from the SDP/H.245 information for SIP and from the OLC sent in the SETUP message for H.323. Any request that would cause the bandwidth constraint to be exceeded is rejected with a SIP 503 Service Unavailable or an H.323 Release Complete.

For example, if an incoming SIP message requests PCMU for a payload/encoding name, a zero (0) payload type, and an 8000 cycle clock rate, the Oracle Communications Session Border Controller must determine how much bandwidth is needed.

## Admission Control and QoS

To accomplish this task, the system checks the media profile values and reserves the bandwidth required for flows. If the required bandwidth for the new flow exceeds the available bandwidth at the time of the request, the Oracle Communications Session Border Controller rejects the session.

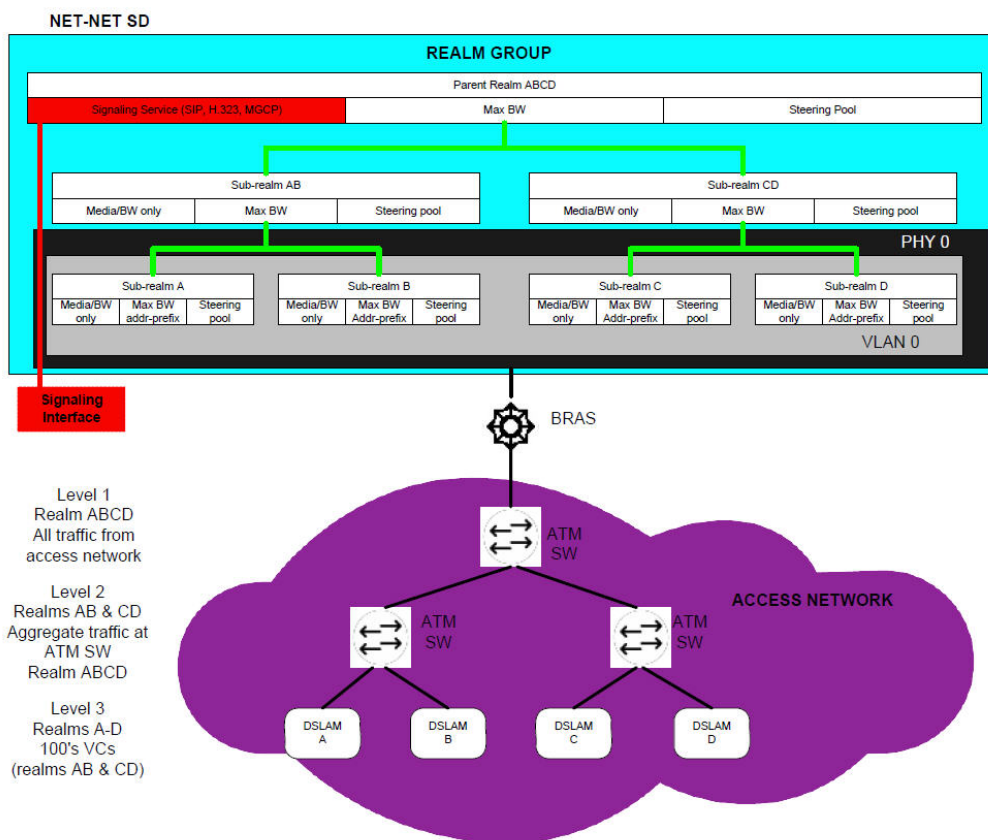
With these mechanisms, the Oracle Communications Session Border Controller provides bandwidth-based admission control.

### Multi-Level Bandwidth Policy Nesting

Multi-level nesting of bandwidth policy enforcement addresses the following issues:

- Bandwidth over-subscription: access or transit transport networks are aggregated and/or oversubscribed. For example, digital subscriber lines (DSL), Frame Relay (FR), and Asynchronous Transfer Mode (ATM). Admission control policies must reflect access network topology.
- Bandwidth partitioning for multiple services: access or transit bandwidth is partitioned among multiple service profiles (SIP, for example) in the same customer network.
- Multi-site VPN environments: admission control must be applied at the site level as well as the VPN level.

The following example illustrates different scenarios; in each there are two or more levels of admission control required. Nested admission control is best depicted by the DSL broadband example.



In DSL access networks, ATM network bandwidth is typically oversubscribed at rates up to 400/1. At Level 3 (above), hundreds of users virtual circuits (VCs) are aggregated to a smaller set of virtual paths (VPs) at each DSLAM. At Level 2, many virtual paths are aggregated at the first ATM switch. Finally, at Level 1, all traffic from all subscribers in the access network is aggregated at the BRAS. Each level of aggregation is oversubscribed, creating the need to perform admission control at each level.

From a Oracle Communications Session Border Controller perspective, multiple tiers of realms are supported, each with its unique bandwidth policy. Only the lowest order realm (Level 3) requires an address prefix (that assigned to the DSLAM) that must be used by the Oracle Communications Session Border Controller to determine in which realm a user resides. When a call request to or from a particular user is received, the Oracle Communications Session

Border Controller checks each realm in the path to determine whether sufficient bandwidth is available to place the call.

## Session Capacity- and Rate-based Admission Control

A session agent defines a signaling endpoint. It is a next hop signaling entity that can be configured to apply traffic shaping attributes. You can define concurrent session capacity and rate attributes for each session agent.

You can configure a set of attributes and constraints for each session agent to support session access control. In this configuration, the Oracle Communications Session Border Controller only accepts requests from configured session agents. And you can set up session admission control so that the Oracle Communications Session Border Controller limits the number of concurrent inbound and outbound sessions for any known service element.

The Oracle Communications Session Border Controller denies a call request to any destination that has exceeded its configured policies for session capacity and session rate. The Oracle Communications Session Border Controller might reject the call request back to the originator. If multiple destinations are available, the Oracle Communications Session Border Controller will check current capacity and rate for each destination and attempt to route the call only to destinations whose policy limits have not been reached.

You assign a media profile to a session agent and indicate whether the transport protocol is SIP or H.323. If the protocol is H.323, you need to indicate whether the session agent is a gateway or a gatekeeper.

### Constraints for Proxy Mode

The Oracle Communications Session Border Controller applies session router and session agent constraints when it is in proxy (transaction or stateless) mode if you enable the CLI **constraints** parameter for a session agent. However, the Oracle Communications Session Border Controller does not track SIP sessions when in transaction mode, so the following session-specific constraints are not applied:

- max-sessions
- max-inbound-sessions
- max-outbound-sessions
- min-seizures
- min-asr

Constraints the Oracle Communications Session Border Controller applies are:

- max-burst-rate
- max-inbound-burst-rate
- max-outbound-burst-rate
- max-sustain-rate
- max-inbound-sustain-rate
- max-outbound-sustain-rate

In order to set the desired time windows for computing burst rates and sustain rates, you also need to configure these parameters in the session agent configuration: **burst-rate-window** and **sustain-rate-window**. You can also set the **time-to-resume** and **in-service-period** parameters to control how long to wait before bringing a session agent back into service after its constraints are no longer exceeded.

## CAC Policing and Marking for non-Audio non-Video Media

The Oracle Communications Session Border Controller supports non-AVT (audio-visual transport) media profile and media policy configurations.

In previous releases, the Oracle Communications Session Border Controller only policed media based on average rate limits configured in media profiles, but these are only applied to AVT. And if there are not required bandwidth or average rate limit values set for the media profile, CAC and policing functions are not applied to media—even if the SDP specifies appropriate bandwidth values. Likewise, ToS markings are not applied for non-AVT media, but only for SIP, H.323, and AVT media types.

With this feature addition, you can now enable your Oracle Communications Session Border Controller to handle non-AVT media types like image and text, and use application and data type for policing purposes. Bandwidth CAC support has also been added for non-AVT media types, as has support for the application specific (AS) bandwidth modifier (`b=AS:<value>`) in the SDP with specification of a defined amount of headroom for that value.

### Bandwidth CAC Fallback Based on ICMP Failure

For networks where backup links (operating in active-standby mode) from CE-routers to the MPLS backbone are provisioned with less bandwidth than the primary links, the Oracle Communications Session Border Controller can:

- Detect remote link failures
- Trigger bandwidth updates at the realm level when using backup links
- Detect remote link fallback to primary

To do so, the Oracle Communications Session Border Controller monitors the primary link status using ICMP echo requests (or pings). It issues the pings at regular intervals, forming a heartbeat mechanism. The CE-router can respond to these pings on the primary link, which is represented by the WAN IP address. When this link fails over, the backup link assumes the same WAN IP address but is not responsive to the pings. This way, the Oracle Communications Session Border Controller determines failover when the ICMP ping fails.

When there is an ICMP ping failure, the Oracle Communications Session Border Controller adjusts the realm's available bandwidth pool from its maximum bandwidth setting to its fallback setting. If the fallback amount is less than the maximum amount, it is possible for the Oracle Communications Session Border Controller to start rejecting calls. It does so until enough calls are released to free adequate bandwidth to stay under the fallback limit and still accept calls.

### Bandwidth CAC Fallback Based on ICMP Failure Configuration

You can set up ICMP heartbeats and fallback bandwidth pools in the realm configuration. Leaving the `icmp-detect-multiplier`, `icmp-advertisement-interval`, or `icmp-target-ip` parameters blank or set to zero turns the feature off.

To enable bandwidth CAC fallback based on ICMP failure:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type `media-manager` and press Enter.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type `realm-config` and press Enter. If you are adding this feature to a pre-existing realm configuration, you will need to select and edit your realm.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

4. `icmp-detect-multiplier`—Enter the multiplier you want to use when determining how long to send ICMP pings before considering a target unreachable. This number multiplied by the time you set for the `icmp-advertisement-interval` determines the length of time. For example, if you set this parameter to 10 and the advertisement interval to 20, the Oracle Communications Session Border Controller will send ICMP pings for 200 seconds before declaring the target unreachable.
5. `icmp-advertisement-interval`—Enter the time in seconds between ICMP pings the Oracle Communications Session Border Controller sends to the target. The default is 0.
6. `icmp-target-ip`—Enter the IP address to which the Oracle Communications Session Border Controller should send the ICMP pings so that it can detect when they fail and it needs to switch to the fallback bandwidth for the realm. There is no default.
7. `fallback-bandwidth`—Enter the amount of bandwidth you want available once the Oracle Communications Session Border Controller has determined that the target is unreachable.

If the fallback amount is less than the **max-bandwidth** value, the Oracle Communications Session Border Controller might start to reject calls. It does so until enough calls are released to free adequate bandwidth to stay under the fallback limit and still accept calls.

8. Save and activate your configuration.

## Bandwidth CAC for Aggregate Emergency Sessions

You can configure the maximum amount of bandwidth on your Oracle Communications Session Border Controller you want used specifically for priority (emergency) calls in the realm configuration's **max-priority-bandwidth** parameter. You set this limit on a per-realm basis, and the limit is enforced for nested realms. Setting a bandwidth limit specifically for priority calls allows the Oracle Communications Session Border Controller to reject calls exceeding the threshold, and also to accept calls that exceed the bandwidth limit for non-priority calls (set in the **max-bandwidth** parameter).

The bandwidth limit for emergency calls operates in conjunction with the bandwidth limits you can set for all other types of calls. When an emergency call comes in, the Oracle Communications Session Border Controller checks the non-priority bandwidth limit. If bandwidth is sufficient, the call goes through and the Oracle Communications Session Border Controller decrements the bandwidth used from the pool of the amount available.


However, if a priority call exceeds the **max-bandwidth** setting, the Oracle Communications Session Border Controller checks the **max-priority-bandwidth** parameter. If it is within the limit for priority calls, the system allows the call and decrements the amount of used bandwidth from what is available.

When there is not enough bandwidth in either the priority or non-priority pool, the Oracle Communications Session Border Controller rejects the call with the corresponding error code and reason phrase.

Any bandwidth subtracted from either pool during a session is returned to that pool as soon as the session ends.

### Bandwidth CAC for Aggregate Emergency Sessions Configuration

You configure bandwidth CAC for priority calls on a per-realm basis.

 **Note:** This parameter honors the hierarchy of nested realms if you have them configured.

To enable bandwidth CAC for aggregate emergency sessions:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **realm-config** and press Enter. If you are adding this feature to a pre-existing realm configuration, you will need to select and edit your realm.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. **max-priority-bandwidth** Enter the amount of bandwidth you want to use for priority (emergency) calls. The system first checks the **max-bandwidth** parameter, and allows the call if the value you set for priority calls is sufficient. If there is not enough priority and non-priority bandwidth allotted for an incoming call, the Oracle Communications Session Border Controller rejects it.

This parameter defaults to 0. You can enter any value between 0 and 999999999.

5. Save and activate your configuration.

## Admission Control for Session Agents

This section explains how to configure session agents for admission control.

### Session Agents Admission Control Configuration

To use admission control based on session rate, you need to configure session agent session rate constraints.

To configure session rates:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # session-agent  
ACMEPACKET (session-agent) #
```

4. Enable session agent constraints and then configure the parameters related to session capacity or session rate to set admission control.

constraints—Enable this parameter. From here you can either configure admission control based on session capacity, session rates, or both. The default value is enabled. The valid values are:

- enabled | disabled

5. **max-sessions**—Set the maximum number of sessions (inbound and outbound) allowed by the session agent. The default value is zero (0). The valid range is:


- Minimum—0
- Maximum—4294967295

6. **max-inbound-sessions**—Enter the maximum number of inbound sessions allowed from this session agent. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

7. **max-outbound-sessions**—Enter the maximum number of concurrent outbound sessions (outbound from the Oracle Communications Session Border Controller ) that are allowed from this session agent. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—4294967295

 **Note:** The number you enter here cannot be larger than the number you entered for max-sessions.

8. **max-burst-rate**—Enter a number to set how many SIP session invitations or H.323 SETUPS this session agent can send or receive (per second) within the configured burst rate window value. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—4294967295

For the sustained rate, the Oracle Communications Session Border Controller maintains a current and previous window size. The period of time over which the rate is calculated is always between one and two window sizes.

For example, if you enter a value of 50 here and a value of 60 (seconds) for the burst rate window constraint, no more than 300 session invitations can arrive at or leave from the session agent in that 60 second time frame (window). Within that 60-second window, any sessions over the limit of 300 are rejected.

9. **max-inbound-burst-rate**—Enter the maximum burst rate (number of session invitations per second) for inbound sessions from this session agent. The default value is zero (0). The valid range is:

- Minimum—0



- Maximum—999999999

**10. max-outbound-burst-rate**—Enter the maximum burst rate (number of session invitations per second) for outbound sessions to this session agent. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

**11. max-sustain-rate**—Enter a number to set the maximum rate of session invitations (per second) this session agent can send or receive within the current window. The default value is zero (0). The valid range is:

- Minimum—zero (0)
- Maximum—4294967295

The number you enter here must be larger than the number you enter for **max-burst-rate**.

For the sustained rate, the Oracle Communications Session Border Controller maintains a current and previous window size. The period of time over which the rate is calculated is always between one and two window sizes.

For example, if you enter a value of 50 here and a value of 36 (seconds) for the sustain rate window constraint, no more than 1800 session invitations can arrive at or leave from the session agent in any given 36 second time frame (window). Within that 36 second window, sessions over the 1800 limit are rejected.

**12. max-inbound-sustain-rate**—Enter the maximum sustain rate (of session invitations allowed within the current window) of inbound sessions from this session agent. This value should be larger than the **max-inbound-burst-rate** value. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

**13. max-outbound-sustain-rate**—Enter the maximum sustain rate (of session invitations allowed within the current window) of outbound sessions to this session agent. This value should be larger than the **max-outbound-burst-rate** value. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999


**14. burst-rate-window**—Enter a number to set the burst window period (in seconds) that is used to measure the burst rate. The term window refers to the period of time over which the burst rate is computed. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—4294967295

**15. sustain-rate-window**—Enter a number to set the sustained window period (in seconds) that is used to measure the sustained rate. The default value is zero (0), which disables the functionality. The valid range is:

- Minimum—10
- Maximum—4294967295

The value you set here must be higher than or equal to the value you set for the burst rate window.

 **Note:** If you are going to use this parameter, you must set it to a minimum value of 10.

The following example shows session agent constraints that are enabled and the session capacity parameters have been configured. Other session agent parameters have been omitted for brevity.

```
session-agent
constraints                enabled
max-sessions                355
max-inbound-sessions       355
max-outbound-sessions      355
```

The following example shows session agent constraints are enabled and the session rate parameters have been configured. Other session agent parameters have been omitted for brevity.

## Admission Control and QoS

```
session-agent
max-burst-rate          0
max-inbound-burst-rate 10
max-outbound-burst-rate 1
max-sustain-rate        3000
max-inbound-sustain-rate 0
max-outbound-sustain-rate 0
burst-rate-window       0
sustain-rate-window     0
```

### Realm Bandwidth Configuration

To configure admission control based on bandwidth, you set the max and min bandwidth parameters in the realm configuration.

To configure realm bandwidth:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

4. Configure the maximum bandwidth.

**max-bandwidth**—Enter a number that sets the maximum bandwidth for dynamic flows to/from the realm in kilobits (Kbps) per second. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—4294967295

The following example shows the maximum bandwidth for the realm has been configured. All other realm parameters have been omitted for brevity.

```
realm-config
max-bandwidth          64000
```

### SIP Admission Control Configuration

You can configure the registered endpoint to accept and process requests from SIP realms. If a request does not meet the criteria of the option you choose here, it is rejected with a 403 (Forbidden) response.

To configure admission control:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. Type **sip-ports** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(sip-interface)# sip-port
ACMEPACKET(sip-port)#
```

## 5. Set the criteria for admission control.

`allow-anonymous`—Enter the anonymous connection mode you want applied when SIP requests are processed. The default value is `all`.

The following are valid values:

- **all**—No ACL is applied and all anonymous connections are allowed.
- **agents-only**—Only requests from configured session agents are processed. The Oracle Communications Session Border Controller responds to all other requests with a forbidden response.
- **realm-prefix**—Only requests from session agents and addresses matching the realm's address prefix are processed. All other requests are rejected with a 403 (Forbidden) response.
- **registered**—Only requests from session agents and registered endpoints are processed. REGISTER allowed from any endpoint.
- **registered-prefix**—Only requests from session agent and registered endpoint addresses that match the realm's realm prefix are processed.

The following example shows the `allow-anonymous` parameter that has been configured to allow only requests from session agents and registered endpoints. All other session agent parameters following the `allow-anonymous` parameters are omitted for brevity.

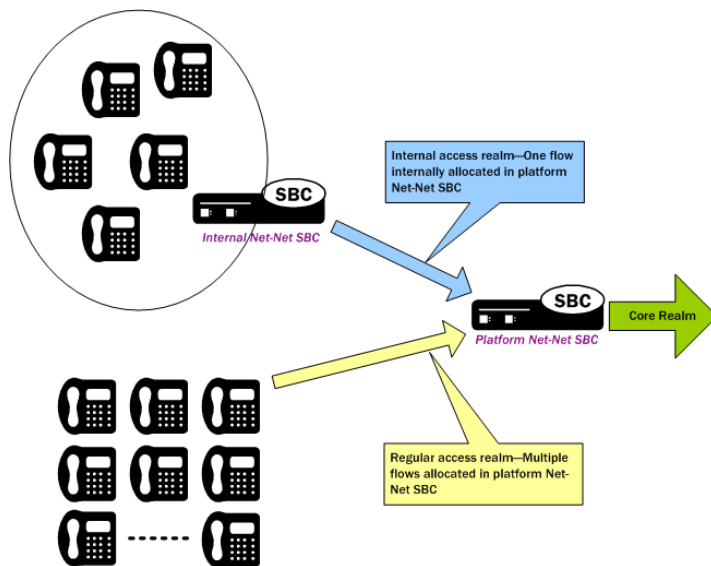
```

sip-port
    address
    port 5060
    transport-protocol UDP
    allow-anonymous registered
  
```

## Session Agent Minimum Reserved Bandwidth

You can assign session agents minimum bandwidth, applicable in accessOracle Communications Session Border Controller deployments. Assigning a session agent minimum bandwidth can prevent overloading other network devices—such as another Oracle Communications Session Border Controller configured as a session agent. Doing so assures signaling bandwidth and availability to the endpoints behind this Oracle Communications Session Border Controller. This feature is only available on the Acme Packet 3820 and Acme Packet 4500.

In the following diagram, the internal Oracle Communications Session Border Controller is configured as a session agent on the platform Oracle Communications Session Border Controller (which conveys traffic to the core realm). Setting up bandwidth reservation allows for the creation of only one allocated flow, and secures bandwidth for all the SIP clients behind the internal Oracle Communications Session Border Controller. Contrast this scenario with the one where the platform Oracle Communications Session Border Controller must allocate multiple flows for many SIP clients.



When you configure minimum reserved bandwidth for session agent to a non-zero value, the Oracle Communications Session Border Controller allocates a separate pipe for per session agent. This is achieved by setting up an access control configuration in a specific way, instructing the Oracle Communications Session Border Controller to use a minimum number of transmission timeslots the individual pipe is guaranteed to receive.

This feature works across both signaling services: SIP and H.323. No more than 4000 session pipes are supported.

### Session Agent Minimum Reserved Bandwidth Configuration

For this feature to work, you must set up an access control configuration as per below.

To configure minimum reserved bandwidth for session agents:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **access-control** and press Enter.

```
ACMEPACKET(session-router)# access-control
ACMEPACKET(access-control)#
```

If you are adding this feature to an existing configuration, you will need to select the configuration you want to edit.

4. **realm-id**—Enter the name of a valid realm.
5. **application-protocol**—Enter a valid application protocol. There is no default for this parameter, and valid values are: **SIP** or **H.323**.
6. **access**—Set this parameter to **permit** (default).
7. **trust-level**—Set this parameter to **high**, changing it from the default (**none**).
8. **minimum-reserved-bandwidth**—Enter the minimum reserved bandwidth you want for the session agent, and that will trigger the creation of a separate pipe for it. Only a non-zero value will allow the feature to work properly, along with the other required values set out in these instructions. The default is 0, and the maximum is 0xffffffff (or 4294967295).
9. Save and activate your configuration.

## Aggregate Session Constraints for SIP

You can set a full suite of session constraints and then apply them to a SIP interface. The session constraints configuration contains many of the same parameters as the session agent, so you can configure a group of constraints and then apply them to a SIP interface/

The SIP interface configuration's **constraint-name** parameter invokes the session constraint configuration you want to apply. Using the constraints you have set up, the Oracle Communications Session Border Controller checks and limits traffic according to those settings for the SIP interface. Of course, if you do not set up the session constraints or you do not apply them in the SIP interface, then that SIP interface will be unconstrained. If you apply a single session-constraint element to multiple SIP interfaces, each SIP interface will maintain its own copy of the session-constraint.

SIP interfaces now have two states: "In Service" and "Constraints Exceeded." When any one of the constraints is exceeded, the status of the SIP interface changes to Constraints Exceeded and remains in that state until the **time-to-resume** period ends. The session constraint timers that apply to the SIP interface are the time-to-resume, burst window, and sustain window.

### Aggregate Session Constraints Configuration

This section shows you how to configure aggregate session constraints and then apply them to a SIP interface.

The session constraints configuration contains many of the same parameters as the session agent does; it also incorporates the changes to the session agent parameters that are described in this section.

To configure the session constraints:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **session-constraints** and press Enter.

```
ACMEPACKET(session-router)# session-constraints
```

4. **name**—Enter the name for this session constraints configuration; this is a unique identifier that you will use in the SIP interface when you want the session constraints applied there. This is a required parameter that has no default.
5. **state**—Enable this parameter to use these session constraints. The default value is **enabled**. The valid values are:
  - enabled | disabled
6. **max-sessions**—Enter the maximum sessions allowed for this constraint. The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—999999999
7. **max-outbound-sessions**—Enter the maximum outbound sessions allowed for this constraint. The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—999999999
8. **max-inbound-sessions**—Enter the maximum inbound sessions allowed for this constraint. The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—999999999

9. **max-burst-rate**—Enter the maximum burst rate (invites per second) allowed for this constraint. This value should be the sum of the **max-inbound-burst-rate** and the **max-outbound-burst-rate**. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

10. **max-sustain-rate**—Enter the maximum rate of session invitations per second allowed within the current window for this constraint. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

For the sustained rate, the Oracle Communications Session Border Controller maintains a current and previous window size. The period of time over which the rate is calculated is always between one and two window sizes.

11. **max-inbound-burst-rate**—Enter the maximum inbound burst rate (number of session invitations per second) for this constraint. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

12. **max-inbound-sustain-rate**—Enter the maximum inbound sustain rate (of session invitations allowed within the current window) for this constraint. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

For the sustained rate, the Oracle Communications Session Border Controller maintains a current and previous window size. The period of time over which the rate is calculated is always between one and two window sizes.

13. **max-outbound-burst-rate**—Enter the maximum outbound burst rate (number of session invitations per second) for this constraint. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

14. **max-outbound-sustain-rate**—Enter the maximum outbound sustain rate (of session invitations allowed within the current window) for this constraint. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

For the sustained rate, the Oracle Communications Session Border Controller maintains a current and previous window size. The period of time over which the rate is calculated is always between one and two window sizes.

15. **time-to-resume**—Enter the number of seconds after which the SA (Session Agent) is put back in service (after the SA is taken OOS (Out Of Service) because it exceeded some constraint). The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

16. **ttr-no-response**—Enter the time delay in seconds to wait before the SA (Session Agent) is put back in service (after the SA is taken OOS (Out Of Service) because it did not respond to the Oracle Communications Session Border Controller). The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

17. **in-service-period**—Enter the time in seconds that elapses before an element (like a session agent) can return to active service after being placed in the standby state. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

**18. burst-rate-window**—Enter the time in seconds that you want to use to measure the burst rate; the window is the time over which the burst rate is calculated, and is used for the over all burst rate as well as the inbound and outbound burst rates. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

**19. sustain-rate-window**—Enter the time in seconds used to measure the sustained rate; the window is the time over which the sustained rate is calculated, and is used for the over all sustained rate as well as the inbound and outbound sustained rates. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—999999999

## Applying Session Constraints in a SIP Interfaces

In the SIP interface, there is a new parameter that allows you to use a set of session constraints for that interface; the parameter is called `constraint-name`.

To apply session constraints to a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET(session-router)# sip-interface
```

4. **constraint-name**—Enter the name of the session constraints configuration that you want to apply to this SIP interface. There is no default for this parameter.
5. Save and activate your configuration.

## Configuring CAC Policing and Marking for non-Audio non-Video Media

In the media profile and the media policy configurations, the following values have been added for the `media-type` parameter:

- **application | data | image | text**

For the media policy, these new values apply to ToS marking.

### Support for the AS Bandwidth Modifier

Two new parameters have been added to the media profile configuration:

- **sdp-bandwidth**—Enable or disable the use of the AS modifier in the SDP if the **req-bandwidth** and **sdp-rate-limit-headroom** parameters are not set to valid values in a corresponding media profile. The default value is **disabled**. The valid values are:
  - **enabled | disabled**
- **sdp-rate-limit-headroom**—Specify the percentage of headroom to be added while using the AS bandwidth parameter while calculating the **average-rate-limit** (rate limit for the RTP flow). The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—100

The following conditions apply to the use and application of these two new parameters:

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- If the amount of required bandwidth is not specified in the media profile (**req-bandwidth**) for the media type in the m= line of the SDP, then the value specified in the AS modifier is used. The Oracle Communications Session Border Controller only uses the AS value if you set the new **sdp-bandwidth** to enabled.
- If the average rate limit value for RTP flows is not specified in the media profile (**average-rate-limit**) for the media type in the m= line of the SDP, then the value specified in the AS modifier is used. The system only uses the AS value if you set the new **sdp-bandwidth** to enabled. When calculating the average rate limit that it will use based on the AS modifier, the Oracle Communications Session Border Controller applies the percentage set in the **sdp-rate-limit-headroom** parameter.
- The Oracle Communications Session Border Controller uses the value specified in the AS modifier (if **sdp-bandwidth** is enabled, and **req-bandwidth** is set to 0) along with the **user-cac-bandwidth** value set in the realm configuration; this works the same way that the **req-bandwidth** parameter does.
- The system uses the value specified in the AS modifier (if **sdp-bandwidth** is enabled, and **req-bandwidth** is set to 0) along with the **max-bandwidth** value set in the realm configuration; this works the same way that the **req-bandwidth** parameter does.

### Media Profile Configuration

To set any of the new media types in the media profile configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **media-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# media-profile
ACMEPACKET(media-profile)#
```

4. **media-type**—Enter the media type that you want to use for this media profile. The valid values are:

- **audio | video | application | data | image | text**

5. Save and activate your configuration.

To set any of the new media types in the media policy configuration:

6. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

7. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

8. Type **media-policy** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# media-policy
ACMEPACKET(media-policy)#
```

9. **media-type**—Enter the media type that you want to use for this media profile. The valid values are:

- **audio | video | application | data | image | text**

10. Save and activate your configuration.

### AS Modifier and Headroom Configuration

To enable AS modifier use and establish the percentage of headroom to use:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```



2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **media-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # media-profile
ACMEPACKET (media-profile) #
```

4. **sdp-bandwidth**—Enable this parameter to use the AS bandwidth modifier in the SDP. The default is **disabled**. Valid values are:
  - enabled | disabled
5. **sdp-rate-limit-headroom**—Specify the percentage of headroom to be added while using the AS bandwidth parameter while calculating the **average-rate-limit** (rate limit for the RTP flow). The default is 0. The valid range is:
  - Minimum—0
  - Maximum—100
6. Save and activate your configuration.

## Offerless Bandwidth CAC for SIP

For SIP sessions offerless INVITES (i.e., INVITES that have no SDP offer), the Oracle Communications Session Border Controller can reserved bandwidth and support the session if you set up applicable media profile associations in the global SIP configuration. Otherwise, the Oracle Communications Session Border Controller terminates these sessions.

You configure support for offerless bandwidth CAC by setting up your global SIP configuration with the options parameters set to **offerless-media-bw-profiles**. The option takes multiple media profile names as values to apply when treating offerless INVITES. When such an INVITE arrives and your configuration supports this option, the Oracle Communications Session Border Controller checks and reserves bandwidth for the session. If there is insufficient bandwidth to reserve, the Oracle Communications Session Border Controller terminates the session. Otherwise, the actual SDP negotiation takes place unaffected while the Oracle Communications Session Border Controller forwards the offerless INVITE. Once the negotiation completes, the Oracle Communications Session Border Controller updates bandwidth reservation.

If the called party's actual bandwidth needs exceed available bandwidth, the Oracle Communications Session Border Controller must terminate the session, even if the session is ringing or answered. To minimize this occurrence as much as possible, you should consider all case scenarios when you select media profiles to use with the **offerless-media-bw-profiles** option.

## Offerless Bandwidth CAC for SIP Configuration

To configure offerless bandwidth CAC for SIP:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-config** and press Enter. If you are editing a pre-existing configuration, you needs to select it before you can make changes.

```
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) #
```

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4. **options**—Your entry will look like this:

```
ACMEPACKET(sip-config)# options offerless-bw-media-profiles=PCMU,G729
```

You can use the plus sign (+) and the minus sign (-) to add and remove values from the options list.

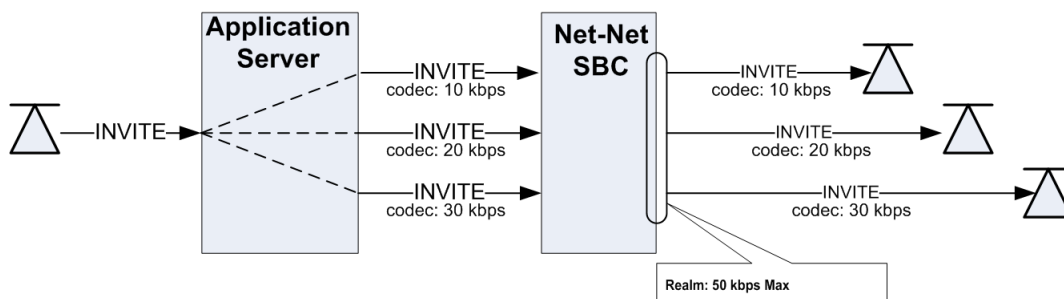
5. Type **done** and continue

## Shared CAC for SIP Forked Calls

A forked call is one which has multiple INVITES for the same call. For example, if an Application Server in the provider core network forks a call attempt, the application server sends several INVITES for the same call toward the Oracle Communications Session Border Controller. Each INVITE is destined for a unique device that belongs to the same user. Ideally, that user will only answer one device. The Oracle Communications Session Border Controller treats each INVITE as a unique call request.

By default, each of the multiple INVITE forks are checked against CAC bandwidth limits, and thus they each consume bandwidth resources when they are received, even though only one of the forks will succeed in establishing a permanent session. Therefore, for many operators the CAC behavior of the SD is too restrictive and results in rejected call attempts which should have been allowed.

The following diagram shows a forked call scenario. The total bandwidth counted against the realm is 60 kbps. If the realm has a bandwidth ceiling of 50 kbps, one of the INVITES will be rejected.



You can, however, enable the system to enforce CAC limits only once for SIP forked calls as long as the calls are identified as such, meaning that they will use the same bandwidth resources. The Oracle Communications Session Border Controller counts the forked call's most bandwidth-hungry codec at the time it arrives at the Oracle Communications Session Border Controller. In the above diagram, with shared bandwidth for forked calls enabled, the Oracle Communications Session Border Controller counts 30 kbps against the realm's total bandwidth after that INVITE arrives, even after the first two INVITES have passed into the final realm.

## Bandwidth Sharing Scenarios

The following table summarizes how bandwidth would be shared given certain ingress and egress realms with this feature enabled. Realms A and C are call ingress realms.; realms B and D are egress realms. For the bandwidth to be shared, Call A and Call B must have the same forked Call-ID in the P-Multiring-Correlator header and be entering or exiting the Oracle Communications Session Border Controller on the same realm.

CALL A					
CALL B		Ingress Realm A	Egress Realm B	Ingress Realm C	Egress Realm D
	Ingress Realm A	bandwidth shared	N/A	bandwidth not shared	N/A
	Egress Realm B	N/A	bandwidth shared	N/A	bandwidth not shared
	Ingress Realm C	bandwidth not shared	N/A	bandwidth shared	N/A
	Egress Realm D	N/A	bandwidth not shared	N/A	bandwidth shared

## Bandwidth Sharing Configuration

To enable bandwidth sharing of forked calls, set the **forked-cac-bw** parameter in the SIP profile configuration to **shared**. Although there are other parameters available in the SIP profile configuration, you only have to set the **name** and the **forked-cac-bw** values to use this feature.

After you set up the SIP profile, you apply it to a realm, SIP interface, or session agent.

### Configuring a SIP Profile

The SIP profile is an element in the ACLI's **session-router** path, and you can configure multiple SIP profiles.

To configure a SIP profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-profile** and press Enter.

```
ACMEPACKET (session-router) # sip-profile
ACMEPACKET (sip-profile) #
```

4. **name**—Enter a name for this SIP profile configuration. This parameter is blank by default, and it is required. You will need the SIP profile's **name** when you want to apply this profile to a realm, SIP interface, or SIP session agent.
5. **forked-cac-bw**—Set this parameter to **shared** if you want forked sessions to share bandwidth resources, or set it to **per-session** if you want bandwidth to be counted for each session individually. There is no default for this parameter, and leaving it blank means:
  - For an ingress session agent without a SIP profile or with a SIP profile where the forked CAC mode is blank, the Oracle Communications Session Border Controller will reference the associated realm.
  - For an ingress realm without a SIP profile or with a SIP profile where the forked CAC mode is blank, the Oracle Communications Session Border Controller will reference the associated SIP interface.
  - For an ingress SIP interface without a SIP profile or with a SIP profile where the forked CAC mode is blank, the Oracle Communications Session Border Controller will not perform bandwidth sharing for forked calls.
6. Save your work.

### Applying a SIP Profile

Once you have configured one or more SIP profiles, you can apply them to realms, SIP interfaces, and SIP session agents. As an example, this section shows you how to apply a SIP profile to a SIP interface. But the parameter name is the same in these configurations:

- realm-config

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- sip-interface
- session-agent

To apply a SIP profile to a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. **sip-profile**—Enter the name of SIP profile configuration that includes the `forked-cac-bandwidth` parameter configured.
5. Save your work.

## RADIUS Accounting Support

VSA 171, `Acme-Session-Forked-Call-Id`, is part of the Oracle RADIUS dictionary. The VSA is a string value, and appears as the header-value without the header parameters from the `P-Multiring-Correlator` header for a session identified as part of a forked call.

## Monitoring

Using the ACLI **show sipd forked** command, you can display the total number of forked sessions the Oracle Communications Session Border Controller received and the total number it rejected. The Oracle Communications Session Border Controller counts forked sessions when it receives a dialog-creating INVITE and is enabled to shared bandwidth. Further, it counts as forked all session with the `P-Multiring-Correlator` header.

```
ACMEPACKET# show sipd forked
11:19:20-116
Forked Sessions          ----- Lifetime -----
                        Recent      Total  PerMax
Forked Sessions          0          0      0
Forked Sessions Rej      0          0      0
```

## Conditional Bandwidth CAC for Media Release

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The Oracle Communications Session Border Controller supports conditional call admission control (CAC) using the SIP profile configuration. With this feature enabled, you can allow the conditional admission of SIP calls that could potentially have their media released instead of risking the possible rejection of those calls due to internal bandwidth limits.

### About Conditional Bandwidth CAC for Media Release

The Oracle Communications Session Border Controller performs bandwidth CAC for SIP per realm, for each Address of Record (AoR) or IP address. The system checks bandwidth limits based on the codecs listed in SDP. If a new SIP INVITE contains codecs in an SDP message that exceed bandwidth available for a given resource, the system rejects that INVITE. This check occurs both on the ingress and egress sides of a call, and both sides must have enough available resources to support the call for it to be admitted.

In the case of calls where media is released, the Oracle Communications Session Border Controller does not count bandwidth consumed by the call. However, this exemption is not given until the media is actually released—and media release conditions are unknown at the time SIP INVITE is admitted. This is because an INVITE received on

one side of the Oracle Communications Session Border Controller is only media-released when that INVITE is routed back through the Oracle Communications Session Border Controller as a hairpin or other multi-system media release. So there has to be enough bandwidth for the initial INVITE; otherwise, and even if the INVITE is a candidate for media release, it will be rejected.

When there is a significant volume of such calls—ones that are candidates for media release, but cannot be admitted because of CAC limits—it becomes important to admit them so long as they truly end in media release. This feature thus allows admission of SIP calls that might otherwise be rejected because of bandwidth limitations when the far-end of the call causes media to be released.

## Details and Conditions

This feature applies in a two system scenario. In order to track a call as a candidate for provisional media release, the access-side Oracle Communications Session Border Controller adds a Require: header with an option tag to the INVITE or UPDATE message on egress. The option tag is configurable in the sip config option. The default is `com.acmepacket.cac`.

The following sections describe when the SIP INVITE or SIP UPDATE are:

- initially received by the Oracle Communications Session Border Controller
- received by the second Oracle Communications Session Border Controller

### INVITES UPDATES Initially Received By Oracle Communications Session Border Controller

When the Oracle Communications Session Border Controller first receives an INVITE or UPDATE message, it considers if it should be admitted provisionally or rejected outright due to CAC bandwidth constraints. If the INVITE or UPDATE is admitted provisionally, a Require: header is inserted on egress from the system.

The Oracle Communications Session Border Controller inserts the Require header on egress under these conditions:

- It receives an INVITE / UPDATE with no or a non-matching Require header.
- The **egress conditional cac admit** parameter in the SIP profile on the egress realm, SIP interface, session agent is set to enabled in the egress realm
- The request would otherwise be rejected because of current bandwidth CAC limits in the ingress OR egress realms
- The call is a candidate for media-release in the ingress realm

A call is considered a candidate for media-release when the ingress realm has any of these parameters set to disabled:

- **mm-in-realm**
- **mm-in-network**
- **mm-same-ip**
- **mm-in-system**

### INVITES UPDATES Received by Second SBC

The second Oracle Communications Session Border Controller receives the INVITE or UPDATE with the newly inserted Require: header. Standard SIP convention indicates that if the UAS receiving the request does not know how to handle the Require header, the request should be rejected.

When the following three conditions are met, the INVITE is permitted into the system for processing:

- The **ingress conditional cac admit** in the SIP profile on the ingress realm, SIP interface, session agent parameter is set to enabled
- The **con-cac-tag** sip config option is configured to the same value as the received Require header's option tag
- The call is a candidate for media-release

The call is considered a candidate for media-release on the second system (indicated by the **ingress conditional cac admit** parameter is set to enabled) when either the ingress or egress realms have any of these parameters set to disabled:

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- `mm-in-realm`
- `mm-in-network`
- `mm-same-ip`
- `mm-in-system`

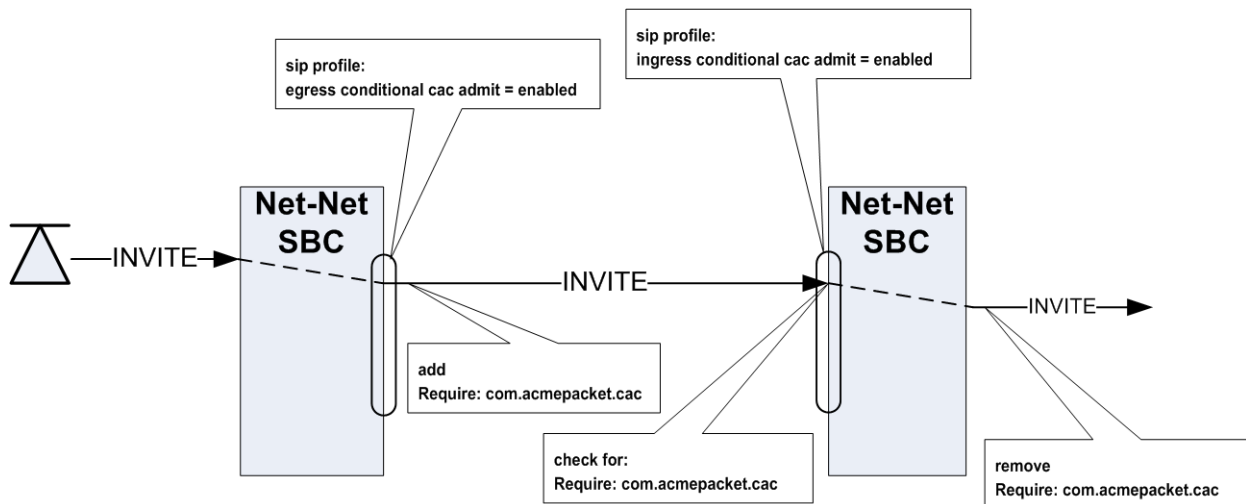
and the following parameter is set to enabled:

- `msm-release`

If the call, as received by the second system is not considered a candidate for release, the INVITE or UPDATE is failed with a 503 Insufficient Bandwidth message.

After the INVITE has been processed by the Oracle Communications Session Border Controller, the Require: header is removed upon egress from the system.

The following diagram shows the two-system scenario:



### Conditional Admission with Per-user CAC

In the event that the per-user CAC feature is also being used, and per-user CAC bandwidth is exceeded, the Oracle Communications Session Border Controller also uses this option tag mechanism. However, if the per-user CAC implementation does count bandwidth regardless of media-release, then the Oracle Communications Session Border Controller will reject calls exceeding the per-user CAC limits when it receives them.

On the second system, when the per-user CAC feature is being used, the Oracle Communications Session Border Controller will perform the same option tag mechanism based on if the `ingress conditional cac admit` parameter is enabled.

### Conditional Bandwidth CAC Configuration

You enable this feature by first configuring a SIP profile, and then applying the profile to any of these:

- realm
- SIP interface
- SIP session agent

#### SIP Profile Configuration

The SIP profile is an element in the ACLI's `session-router` path, and you can configure multiple SIP profiles. Though this configuration contains additional parameters, you do not have to use them for the conditional bandwidth CAC for media release.

To configure a SIP profile:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **account-config** and press Enter.

```
ACMEPACKET (session-router) # sip-profile
ACMEPACKET (sip-profile) #
```

4. **name**—Enter a name for this SIP profile configuration. This parameter is blank by default, and it is required. You will need the SIP profile’s **name** when you want to apply this profile to a realm, SIP interface, or SIP session agent.
5. **ingress-conditional-cac-admit**—Set this parameter to enabled to process an INVITE with a Require tag as received on an ingress interface. You can set this parameter to disabled if you do not want to use this feature on the ingress side. There is no default for this parameter.
6. **egress-conditional-cac-admit**—Set this parameter to enabled if you want to use conditional bandwidth CAC for media release for calls that are first received by this system. This results in option tags being inserted on the INVITE’s egress if the conditional CAC conditions are met. You can set this parameter to disabled if you do not want to use this feature. There is no default for this parameter.
7. Save your work.

## Applying a SIP Profile

Once you have configured one or more SIP profiles, you can apply them to realms, SIP interfaces, and SIP session agents. As an example, this section shows you how to apply a SIP profile to a SIP interface. But the parameter name is the same in these configurations:

- realm-config
- sip-interface
- session-agent

To apply a SIP profile to a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router) # sip-interface
ACMEPACKET (sip-interface) #
```

4. **sip-profile**—Enter the name of SIP profile configuration you want to use for conditional bandwidth CAC for media release for this SIP interface. This value is blank by default, but it must be the value of the **name** parameter from a valid SIP profile.
5. Save your work.

## Configuring Require Header Option Tag

You may change the Require: header’s option tag from the default com.acmepacket.cac to one of your own choosing. Remember that both systems’ option tags must match exactly.

To configure the Require: header’s option tag:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

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2. Type **session-router** and press Enter to access the **session-router** path.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-config
```

4. Use the ACLI **select** command so that you can work with the SIP configuration.

```
ACMEPACKET(sip-config)# select
```

5. **options**—Set the options parameter by typing **+options**, a Space, the option name **con-cac-tag=** your-new-tag, and then press Enter.

```
ACMEPACKET(sip-config)# options +con-cac-tag=com.test.cac
```

6. Save your work.

## CAC Utilization Statistics via SNMP

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The Oracle Communications Session Border Controller allows you to retrieve information on current session utilization and burst rate as a percentage of their configured maximums on per session-agent and/or realm basis. The Oracle Communications Session Border Controller uses the configured max-session and max-burst-rate settings in conjunction with a percentage formula to calculate this value. The system also uses an ACLI configuration setting to establish the threshold at which trap and trap clear messages are sent from the SNMP agent to the configured manager(s).

The user must load the MIB version associated with this software version on all pertinent SNMP managers to query these CAC utilization (occupancy) values and interpret the traps. In addition, the user must configure the threshold at which the system generates the CAC utilization trap. Note that the corresponding clear trap uses the same threshold setting, sending the clear trap when utilization falls below 90% of the threshold.

### SNMP Get for CAC Utilization

Using a MIB browser, the user can query the current percentage utilization values for both max-session and max-burst-rate for any session-agent or realm. The calculations for these utilization levels are:

- Session utilization level = (current session count \* 100) / max-sessions
- Burst rate utilization level = (current burst rate \* 100) / max-burst-rate

The MIB objects associated with these statistics are parallel for session agent and realm and include a table to contain the objects, an object associating the objects containing the values with the applicable table, and objects containing the values themselves. These objects are listed below.

The MIB objects containing CAC utilization data for Session Agents are listed below.

The object establishing the statistics table for session agent CAC utilization follows:

```
--apSip Session Agent Connection Admission Control Stats Table
apSipSaCacStatsTable OBJECT-TYPE
    SYNTAX          SEQUENCE OF ApSipSaCacStatsEntry
    MAX-ACCESS      not-accessible
    STATUS          current
    DESCRIPTION
        "SIP Session Agent Connection Admission Control Stats Table."
    ::= { apSipMIBTabularObjects 5 }
```

The object establishing the session agent CAC utilization statistics objects follows:

```
apSipSaCacStatsEntry OBJECT-TYPE
    SYNTAX          ApSipSaCacStatsEntry
    MAX-ACCESS      not-accessible
    STATUS          current
    DESCRIPTION
```



```

    "Connection Admission Control Statistics."
    AUGMENTS { apSipSessionAgentStatsEntry }
    ::= { apSipSaCacStatsTable 1 }

```

The session agent CAC utilization statistics values include:

```

ApSipSaCacStatsEntry ::= SEQUENCE {
    apSipSaCacSessionUtilLevel          Gauge32,
    apSipSaCacBurstRateUtilLevel       Gauge32
}

```

The above objects, specifying the CAC utilization value for sessions and burst rate utilization for session agents include:

```

apSipSaCacSessionUtilLevel          OBJECT-TYPE
    SYNTAX          Gauge32
    UNITS            "percentage"
    MAX-ACCESS      read-only
    STATUS           current
    DESCRIPTION
        "Current session utilization level."
    ::= { apSipSaCacStatsEntry 1 }

apSipSaCacBurstRateUtilLevel        OBJECT-TYPE
    SYNTAX          Gauge32
    UNITS            "percentage"
    MAX-ACCESS      read-only
    STATUS           current
    DESCRIPTION
        "Current burst rate utilization level."
    ::= { apSipSaCacStatsEntry 2 }

```

The MIB objects containing CAC utilization data for Realms are listed below.

The object establishing the statistics table for realm CAC utilization follows:

```

--apSig Realm Connection Admission Control Stats Table
apSigRealmCacStatsTable OBJECT-TYPE
    SYNTAX          SEQUENCE OF ApSigRealmCacStatsEntry
    MAX-ACCESS      not-accessible
    STATUS           current
    DESCRIPTION
        "Realm Connection Admission Control Stats Table."
    ::= { apSigMIBTabularObjects 6 }

```

The object establishing the realm CAC utilization statistics objects follows:

```

apSigRealmCacStatsEntry OBJECT-TYPE
    SYNTAX          ApSigRealmCacStatsEntry
    MAX-ACCESS      not-accessible
    STATUS           current
    DESCRIPTION
        "Connection Admission Control Statistics."
    AUGMENTS { apSigRealmStatsEntry }
    ::= { apSigRealmCacStatsTable 1 }

```

The session agent CAC utilization statistics values include:

```

ApSigRealmCacStatsEntry ::= SEQUENCE {
    apSigRealmCacSessionUtilLevel      Gauge32,
    apSigRealmCacBurstRateUtilLevel    Gauge32
}

```

The above objects, specifying the CAC utilization value for sessions and burst rate utilization for realms include:

```

apSigRealmCacSessionUtilLevel        OBJECT-TYPE
    SYNTAX          Gauge32

```

```
UNITS          "percentage"
MAX-ACCESS     read-only
STATUS         current
DESCRIPTION    "Current session utilization level."
::= { apSigRealmCacStatsEntry 1 }

apSigRealmCacBurstRateUtilLevel      OBJECT-TYPE
SYNTAX      Gauge32
UNITS       "percentage"
MAX-ACCESS  read-only
STATUS      current
DESCRIPTION "Current burst rate utilization level."
::= { apSigRealmCacStatsEntry 2 }
```

### CAC Utilization Traps

The Oracle Communications Session Border Controller can issue a trap when either the value of max-session or CAC burst rate exceeds a configured value. The system only sends one trap when the threshold is exceeded. When the value falls back under 90% of this threshold, the Oracle Communications Session Border Controller sends a clear trap.

You configure the value that triggers these traps as a percentage of the max-session and max-burst-rate settings configured for the applicable session agent and/or realm. The system uses the same setting to specify when to send both the sessions and burst rate traps. The name of this parameter is the **cac-trap-threshold**.

For realms, you configure a **session-constraint** element with the **cac-trap-threshold** setting and apply that session constraint to the realm. For a session agent however, you configure the **cac-trap-threshold** directly within the session agent's configuration.

The syntax for the command is the same within session constraint and session agent configurations.

#### **cac-trap-threshold [0-99]**

You must express the value as a number less than 100. There is no default setting; the system does not generate a trap if you have not configured this setting.

The `apSipCACUtilAlertTrap` identifies the threshold exceeded on a per-element and per-value (session count or burst rate) for each trap, including:

- `apSipSaCacSessionUtilLevel`
- `apSipSaCacBurstRateUtilLevel`
- `apSipRealmCacSessionUtilLevel`
- `apSipRealmCacBurstRateUtilLevel`

## CAC utilization threshold trap on a session agent configuration

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The CAC utilization threshold causes the system to generate a trap when session count or CAC max-burst-rate exceeds the configured percentage value of these values maximums. This setting is available within a session agent's configuration.

To configure the CAC trap threshold on a session agent, follow the procedure below.

1. Access the **session-agent** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)
```

2. Select the **session-agent** object to edit.

```
ACMEPACKET(session-agent)# select
<hostname>:
1: 192.168.100.101:1813

selection: 1
ACMEPACKET(session-agent)#
```

3. **cac-trap-threshold**—Set the threshold when reached, expressed as a percentage of **max-sessions**, when the CAC trap is sent.
4. Type **done** to save your configuration.

## Configuring the CAC Utilization Thresholds - realm

To configure the CAC trap threshold on a realm or sip interface, create a session constraint object and apply it to your realm, as shown below.

1. Use the following sequence to navigate to session constraint elements.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# session-constraints
ACMEPACKET(session-constraints)#
```

2. Select or create the desired session constraint element.

```
ACMEPACKET(session-constraints)#name trap-at-90-percent
```

3. Configure the desired value for **cac-trap-threshold** expressed as a percentage value, such as 90%, as follows.

```
ACMEPACKET(session-constraints)#cac-trap-threshold 90
```

4. Navigate to the realm-config to which you want to apply the session constraint.

```
ACMEPACKET(realm-config)#session-constraint trap-at-90-percent
```

5. Execute the **done** and **exit** commands.
6. Save and activate your configuration.

## About QoS Reporting

This section describes the Oracle Communications Session Border Controller QoS reporting. QoS reporting provides you with real-time evaluation of network and route performance. It lets you contrast internal domain and external domain performance and facilitates SLA verification and traffic engineering. Oracle Communications Session Border Controller QoS reporting is a measurement tool that collects statistics on Voice over IP (VoIP) call flows for SIP and H.323. To provide information, the Oracle Communications Session Border Controller writes additional parameters to the Remote Authentication Dial-in User Service (RADIUS) call record. To provide information, the Oracle Communications Session Border Controller writes additional parameters to the Remote Authentication Dial-in User Service (RADIUS) call record and Historical Data Recording (HDR) records.

You can use QoS statistics for SLA customer reporting, fault isolation, SLA verification, and traffic analysis. The Oracle Communications Session Border Controller employs specialized hardware to inspect Real-Time Transport Protocol (RTP) and Real-Time Transport Control Protocol (RTCP) flows while maintaining wire-speed packet forwarding. QoS metrics are collected and reported on a per-session and per call-leg basis. These metrics are reported through real-time RADIUS records along with call accounting data.

### Overview

When a conversation is established between two endpoints, two flows are present in each direction:

- RTP flow carries traffic between endpoints with a predictable packet arrival rate. The packets headers have sequence numbers that are used to determine whether packets are missing or lost.

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- RTCP flow carries information about the RTP flow and keeps a different record. The RTCP packets contain timestamps based on Network Time Protocol (NTP).

### QoS Statistics

Reported QoS data includes the following per-flow statistics:

- RTP and RTCP lost packets—Count of lost packets for both RTP and RTCP based on comparing the sequence numbers since the beginning of the call or the last context memory poll.
- RTP and RTCP average jitter—Incremental number of packets for both RTP and RTCP that have been used to generate the total and max jitter since the beginning of the call or the last context memory poll. The incremental accumulated jitter (in milliseconds) over all the packets received.
- RTP and RTCP maximum jitter—Maximum single jitter value (in milliseconds) for both RTP and RTCP from all the packets since the beginning of the call or the last context memory poll.
- RTCP average latency—Number of RTCP frames over which latency statistics have been accumulated and the incremental total of latency values reported since the beginning of the call or the last context memory poll.
- RTCP maximum latency—Highest latency value measured since the beginning of the call or the last context memory poll.
- RTP packet count
- RTP bytes sent and received
- RTCP lost packets—RTP lost packets reported in RTCP packets.
- ATP lost packets—Lost packets determined by monitoring RTP sequence numbers.
- R-Factor and MOS data—R-Factor and MOS data for the calling and called segments at the end of a session

### RADIUS Support

All the QoS statistics go into the RADIUS CDR. If a RADIUS client is configured on the Oracle Communications Session Border Controller, any time a call occurs a record is generated and sent. Only Stop RADIUS records contain the QoS statistic information.

Only RADIUS Stop records contain QoS information. For non-QoS calls, the attributes appear in the record, but their values are always be zero (0). When you review the list of QoS VSAs, please note that “calling” in the attribute name means the information is sent by the calling party and called in the attribute name means the information is sent by the called party.

The following example shows a CDR that includes QoS data:


```
Wed Jun 13 18:26:42 2007
  Acct-Status-Type = Stop
  NAS-IP-Address = 127.0.0.100
  NAS-Port = 5060
  Acct-Session-Id = "SDgtu4401-c587a3aba59dcae68ec76cb5e2c6fe6f-v3000i1"
  Acme-Session-Ingress-CallId =
"8EDDDC21D3EC4A218FF41982146844310xac1ec85d"
  Acme-Session-Egress-CallId = "SDgtu4401-
c587a3aba59dcae68ec76cb5e2c6fe6f-v3000i1"
  Acme-Session-Protocol-Type = "SIP"
  Calling-Station-Id = ""9998776565" <sip:
9998776565@10.10.170.2:5060>;tag=2ed75b8317f"
  Called-Station-Id = "<sip:7143221099@10.10.170.2:5060>"
  Acct-Terminate-Cause = User-Request
  Acct-Session-Time = 7
  h323-setup-time = "18:24:36.966 UTC JUN 13 2007"
  h323-connect-time = "18:24:37.483 UTC JUN 13 2007"
  h323-disconnect-time = "18:24:44.818 UTC JUN 13 2007"
  h323-disconnect-cause = "1"
  Acme-Session-Egress-Realm = "peer"
  Acme-Session-Ingress-Realm = "core"
  Acme-FlowID_FS1_F = "localhost:65544"
  Acme-FlowType_FS1_F = "PCMA"
  Acme-Flow-In-Realm_FS1_F = "core"
```

```
Acme-Flow-In-Src-Addr_FS1_F = 10.10.170.15
Acme-Flow-In-Src-Port_FS1_F = 49156
Acme-Flow-In-Dst-Addr_FS1_F = 10.10.170.2
Acme-Flow-In-Dst-Port_FS1_F = 31008
Acme-Flow-Out-Realm_FS1_F = "peer"
Acme-Flow-Out-Src-Addr_FS1_F = 10.10.130.2
Acme-Flow-Out-Src-Port_FS1_F = 21008
Acme-Flow-Out-Dst-Addr_FS1_F = 10.10.130.15
Acme-Flow-Out-Dst-Port_FS1_F = 5062
Acme-Calling-RTCP-Packets-Lost_FS1 = 0
Acme-Calling-RTCP-Avg-Jitter_FS1 = 15
Acme-Calling-RTCP-Avg-Latency_FS1 = 0
Acme-Calling-RTCP-MaxJitter_FS1 = 15
Acme-Calling-RTCP-MaxLatency_FS1 = 0
Acme-Calling-RTP-Packets-Lost_FS1 = 0
Acme-Calling-RTP-Avg-Jitter_FS1 = 3
Acme-Calling-RTP-MaxJitter_FS1 = 44
Acme-Calling-Octets_FS1 = 957
Acme-Calling-Packets_FS1 = 11
Acme-FlowID_FS1_R = "localhost:65545"
Acme-FlowType_FS1_R = "PCMA"
Acme-Flow-In-Realm_FS1_R = "peer"
Acme-Flow-In-Src-Addr_FS1_R = 10.10.130.15
Acme-Flow-In-Src-Port_FS1_R = 5062
Acme-Flow-In-Dst-Addr_FS1_R = 10.10.130.2
Acme-Flow-In-Dst-Port_FS1_R = 21008
Acme-Flow-Out-Realm_FS1_R = "core"
Acme-Flow-Out-Src-Addr_FS1_R = 10.10.170.2
Acme-Flow-Out-Src-Port_FS1_R = 31008
Acme-Flow-Out-Dst-Addr_FS1_R = 10.10.170.15
Acme-Flow-Out-Dst-Port_FS1_R = 49156
Acme-Called-RTCP-Packets-Lost_FS1 = 0
Acme-Called-RTCP-Avg-Jitter_FS1 = 13
Acme-Called-RTCP-Avg-Latency_FS1 = 0
Acme-Called-RTCP-MaxJitter_FS1 = 21
Acme-Called-RTCP-MaxLatency_FS1 = 0
Acme-Called-RTP-Packets-Lost_FS1 = 0
Acme-Called-RTP-Avg-Jitter_FS1 = 0
Acme-Called-RTP-MaxJitter_FS1 = 3
Acme-Called-Octets_FS1 = 77892
Acme-Called-Packets_FS1 = 361
Acme-Firmware-Version = "C5.0.0"
Acme-Local-Time-Zone = "Time Zone Not Set"
Acme-Post-Dial-Delay = 110
Acme-Primary-Routing-Number = "sip:7143221099@10.10.170.2:5060"
Acme-Ingress-Local-Addr = "10.10.170.2:5060"
Acme-Ingress-Remote-Addr = "10.10.170.15:5060"
Acme-Egress-Local-Addr = "10.10.130.2:5060"
Acme-Egress-Remote-Addr = "10.10.130.15:5060"
Acme-Session-Disposition = 3
Acme-Disconnect-Initiator = 2
Acme-Disconnect-Cause = 16
Acme-SIP-Status = 200
Acme-Egress-Final-Routing-Number = "sip:7143221099@10.10.130.15:5060"
Acme-CDR-Sequence-Number = 14
Client-IP-Address = 172.30.20.150
Acct-Unique-Session-Id = "0832b03cd3a290b3"
Timestamp = 1181773602
```

### Configuring QoS

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This section explains how to configure QoS. To generate QoS metrics, you need to enable QoS for the realm of the originating caller. The ingress realm determines whether QoS is turned on for a specific flow.

 **Note:** If you run with QoS turned on one side only and disabled on the other you lose the ability to measure latency through the use of RTCP timestamps.

### QoS Configuration

To enable QoS:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure)# media-manager
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager)# realm-config  
ACMEPACKET (realm-config)#
```

4. **qos-enable**—Enable this parameter. The default value is **disabled**.

### Network Management Controls

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The Oracle Communications Session Border Controller supports network management controls for multimedia traffic specifically for static call gapping and 911 exemption handling. These controls limit the volume or rate of traffic for a specific set of dialed numbers or dialed number prefixes (destination codes).

In TDM networks, automatic call/code gapping was developed as part of the advanced intelligent network (AIN) to enable network element load shedding based on destination number (DN) in case of overload. However, since there are as yet no standards for call/code gapping for next generation multimedia networks, the Oracle Communications Session Border Controller provides statically-provisioned network management controls.

To enable network management controls on your Oracle Communications Session Border Controller, you set up the ACLI **net-management-control** configuration and then enable the application of those rules on a per-realm basis. Each network management control rule has a unique name, in addition to information about the destination (IP address, FQDN, or destination number or prefix), how to perform network management (control type), whether to reject or divert the call, the next hop for routing, and information about status/cause codes. Details about the content of control rules and how to set them appear in the instructions and examples section.

When a SIP INVITE or an H.323 Setup for a call arrives at the Oracle Communications Session Border Controller on an ingress realm where network management controls have been enabled, the Oracle Communications Session Border Controller takes the following steps:

- It searches the network management rules you have created (which are stored in tables on the Oracle Communications Session Border Controller) for a control rule that best matches the newly-received call.
- If it does not find a matching control rule, the Oracle Communications Session Border Controller allows the call to proceed normally.
- If it finds a matching control rule, then the Oracle Communications Session Border Controller treats the call according to the specifics of the rule and the treatment method that it designates.

### Matching a Call to a Control Rule

The Oracle Communications Session Border Controller uses the call classification key (specified by the **destination-identifier** parameter) to match calls so that it can apply control rules. The call classification

key specifies information about the destination, which can be an IP address, an FQDN, a destination (called) number, or destination prefix. You configure the classification key as part of the control rule.

Matching is performed from left to right, starting at the left-most character. A wildcard matches any digit.

The Oracle Communications Session Border Controller compares the following information from the SIP INVITE or H.323 Setup for matching:

- SIP INVITE—User part of the Request URI, or the host part of the Request URI
- H.323 Setup—Q.931 Called Party Number IE

With Release 6.0, the Oracle Communications Session Border Controller now normalizes the user-part of the Request-URI prior to performing any matching for NMC based on the dialed number. A departure from this feature's prior implementation, this normalization strips out any of the visual-separator characters.

Note that normalization occurs only for NMC look-up purposes, and it does not alter the actual Request-URI. For previous releases, NMC rule matching based on the dialed number fails when the dialed number has visual separators or additional parameters such as: rn, npdi, cic, postd, etc. If multiple rules match an incoming call, then the Oracle Communications Session Border Controller gives first priority to destination number or the destination prefix. Next, it tries to match based on the IP address, and finally it looks to the domain (lowest priority).

Specifically, the Oracle Communications Session Border Controller supports the following:

- The user-part can contain escaped sequences that the Oracle Communications Session Border Controller normalizes to their unescaped representation. For examples %23(358)555.1234567 would be normalized to #3585551234567.
- The Oracle Communications Session Border Controller parses the user-part of the Request-URI up to the first semicolon (;). For example, the user-part in tel:+358-555-1234567;postd=pp22 will be +358-555-12134567.

## Call Handling Determination

There are three types of control rules from which you can choose; each is a different way for the Oracle Communications Session Border Controller to handle calls matching the classification key:

- Call gap rate—Controls the maximum sustained rate of calls that match the classification key.

Using this type, the Oracle Communications Session Border Controller calculates the time since the last matching call. If that time is equal to or greater than the minimum time gap set in the control rule (i.e., it does not exceed the rate), then the call proceeds normally. If the call is less than the minimum time gap (i.e., it causes the call rate to be exceeded), then the Oracle Communications Session Border Controller either rejects or diverts the call.

To keep the call rate below the control value, the Oracle Communications Session Border Controller ensures a minimum call gap time between the matching calls. For example, if the control value is 10 calls per second, the minimum call gap time would be 0.1 second. And if a matching call were to arrive within a tenth of a second since the last matching call, then the Oracle Communications Session Border Controller applies the treatment method.

- Call gap percentage—Controls the percentage of calls matching the classification key you set for the control rule.

When using this control rule type, the Oracle Communications Session Border Controller applies the treatment method to the percentage of matching calls (that you set in the value parameter) out the total number of matching calls it receives. For example, if you set the value parameter for the control rule to 50 and use this control type, the Oracle Communications Session Border Controller applies the treatment method to every other call it receives (or 50% of the calls it receives) that matches the classification key.

Note that the Oracle Communications Session Border Controller cannot maintain exact percentages for the control value at all times, especially at system start-up when the number of incoming calls is small.

- Priority—Exempts calls to a destination (like 911) from local network management controls such as:
  - Session agent constraints
  - Bandwidth constraints (such as per-realm bandwidth)
  - External policy servers (requests are made to the policy server; calls are admitted and processed regardless of the decision or reachability of the policy server)
  - Per-user call admission control

- CPU constraints

The Oracle Communications Session Border Controller will not bypass licensing constraints, however.

### Treatment Methods

You can choose from two different treatment methods:

- Call rejection—The Oracle Communications Session Border Controller rejects the call.
  - For SIP, the Oracle Communications Session Border Controller sends a response messages with the status code of the control rule. This response message also includes a Reason header with the Q.850 cause code that you configure as part of the control rule; it contains only the Q.850 cause code, and there is no reason-text included. For example:

```
Reason: Q.850; cause=63
```

- For H.323, the Oracle Communications Session Border Controller sends a releaseComplete message with the Q.850 cause code (that you configure as part of the control rule) of the control rule as the Q.931 Cause IE.
- Call diversion—The Oracle Communications Session Border Controller routes the call to the location you specify in the control rule's next hop parameter.

Except for this routing, the call proceeds as normal. Local treatments such as number translation apply to the call, as do local controls such as licensing. Note the following:

- If the next hop is an FQDN, the Oracle Communications Session Border Controller performs DNS queries to resolve the next hop to an IP address so that it can route the call properly.
  - DNS queries only apply to pure SIP or IWF calls that originate in H.323 and are interworked to SIP.
- If the next hop is a session agent group, the Oracle Communications Session Border Controller selects a session agent from the group according to the selection strategy you set for the group. Then the Oracle Communications Session Border Controller uses the IP address of the selected session agent.

### Priority Call Exemption from Policy Server Approval

The Oracle Communications Session Border Controller now identifies priority calls and provides expedited treatment for them, even if these calls use associated realms for which there is an associated policy server handling bandwidth allocation. Instead of waiting for a response for the policy server, the Oracle Communications Session Border Controller immediately processes the call. When and if the policy server responds, the Oracle Communications Session Border Controller handles the response, but in all likelihood the priority calls has already been processed.

### Enhanced Call Gapping

NMC provides flexibility by allowing a desired call-per-second (CPS) threshold to be achieved or surpassed by a predictable amount. Referred to as call gapping, this allows the Oracle Communications Session Border Controller to average the call rate and widen the period of a surge that would invoke NMC rules.

Without call gapping enabled, the NMC carries out a call gapping policy that monitors the arrival times between INVITEs, and then compares the arrival times to with the threshold. To enable this, you set the **type** parameter to gap-rate, and then configure the **value** parameter with the maximum sustained rate of calls. The threshold is equal to 1/gap-rate value. However, this implementation means that if two calls arrive simultaneously at the Oracle Communications Session Border Controller, one of them might be rejected or diverted if it exceeds the threshold and the control rule is applied. This is the case even when the sustained call rate does not exceed the control rule.

To resolve this, call gapping uses two parameters that form part of an calculation the Oracle Communications Session Border Controller performs for applying NMC rules. Using the current time, the time of the last call gapped, the call counter value (tracked internally by the Oracle Communications Session Border Controller), the CPS value for the gap-rate control rule, and the values of the new parameters, the Oracle Communications Session Border Controller performs calculations that determine whether or not to apply the control rule.



## About the Call Gapping Algorithm

The Oracle Communications Session Border Controller employs this leaky bucket algorithm to enforce calls per second. It smooths the call rate over a defined window of time to protect against surges. The values used for the calculation are:

- **A**—Calls per second; configure by setting the **type** parameter to `gap-rate`, and the **value** parameter to the CPS you want enforced
- **m**—Maximum counter value; must be greater than 0
- **W**—Window size; must be greater than
- **deltaT**—Time between allowed calls matching an NM control rule

The calculation is performed as follows, with the noted results:

- $1 + m - m * A * \text{deltaT} / W \leq M$ —Means the call is allowed
- $1 + m - m * A * \text{deltaT} / W > M$ —Means that NMC rules are applied

Note the following:

- Setting the counter value and the window size to the same values guarantees that the processed CPS load will not exceed the desired CPS target.
- As the counter value becomes greater than the window size value, rejection rate will drop and the desired CPS threshold is not guaranteed.
- Increasing the window size results in a lower rejection rate when the attempted CPS is the same as the desired CPS; as the attempted CPS rate increases, rejection rates increase at a steeper rate.
- If either the count rate or the window size is set to 0, then the Oracle Communications Session Border Controller reverts to call gapping behavior it uses when the relevant parameters are not configured.

## Network Management Control Configuration

In order use the network management controls feature, you need to set control rules and then enable their application on a per-realm basis. This section shows you how to set up those configuration.

### Configuring an Individual Control Rule

To configure individual network management control rule:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **net-management-control** and press Enter.

```
ACMEPACKET(session-router)# net-management-control
```

4. **name**—Enter the name of this network management control rule; this value uniquely identifies the control rule. There is no default for this parameter.
5. **state**—Enable or disable this network management control rule. The default value is **enabled**. The valid values are:
  - `enabled` | `disabled`
6. **destination-identifier**—Enter the call classification key. This parameter specifies information about the destination, which can be an IP address, an FQDN, a destination (called) number, or destination prefix. You can wildcard characters in the classification key using the carat symbol (^).

You can enter special characters in the **destination-identifier** parameter. You can enter characters such as the plus-sign (+), the asterisk (\*), the pound sign (#), capital letter A (A), capital letter B (B), capital letter C (C), capital letter D (D), lowercase letter p (p), lowercase letter w (w).

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This parameter can accommodate a list of entries so that, if necessary, you can specify multiple classification keys. You can edit the list of classification keys using the ACLI **add-destination-identifier** and **remove-destination-identifier** commands from within the network management controls configuration.

7. **type**—Enter the control type that you want to use. The valid values are:
  - **GAP-RATE**—Controls the maximum sustained rate of calls that match the classification key.
  - **GAP-PERCENT**—Controls the percentage of calls matching the classification key you set for the control rule.
  - **PRIORITY**—Exempts calls to a destination (like 911) from local network management controls. Use this value if you want to enable Priority Call Exemption from Policy Server Approval.
8. **value**—When you set the control type to either GAP-RATE or GAP-PERCENT, enter the maximum sustained rate of calls or the percentage of calls to which you want the control rule applied. The default value is zero (0). The valid values are:
  - **GAP-RATE**—Maximum is 2147483647 (which you can set by entering -1 as the value, an option provided for ease of use)
    - Using the minimum value (0) means that the Oracle Communications Session Border Controller treats all calls
    - Using the maximum value means that the Oracle Communications Session Border Controller treats no calls
  - **GAP-PERCENT**—Maximum is 100
    - Using the minimum value (0) means that the Oracle Communications Session Border Controller treats no calls
    - Using the maximum value (100%) means that the Oracle Communications Session Border Controller treats all calls
9. **treatment**—Enter the treatment method that you want to use. The default value is **none**. The valid values are:
  - **reject**—The Oracle Communications Session Border Controller rejects the call.
  - **divert**—The Oracle Communications Session Border Controller routes the call to the location you specify in the control rule's next hop parameter.
10. **next-hop**—Enter the next hop for the Oracle Communications Session Border Controller to use when the treatment method is **DIVERT**. The valid values are:
  - hostname(:port)
  - IP address(:port)
  - Name of a valid, configured session agent
  - Name of a valid, configured session agent group—When you set this parameter to a session agent group, you must specify that it is a session agent group by prepending the name of the group with either SAG: or sag:. For example, the entry for a session agent group with Group2 as its name would be SAG:Group2 or sag:Group2.
11. **realm-next-hop**—Enter the realm identifier to designate the realm of the next hop when the treatment type is **DIVERT**.
12. **protocol-next-hop**—Enter the signaling protocol for the next hop when the treatment type is **DIVERT**.
13. **status-code**—Enter the SIP response code that you want the Oracle Communications Session Border Controller to use when the treatment method is **REJECT**. The default value is **503** (Service Unavailable). The valid range is:
  - Minimum—1
  - Maximum—699
14. **cause-code**—Enter the Q.850 cause code that you want the Oracle Communications Session Border Controller to use when the treatment method is **REJECT**. The default value is **63** (Service or option not available). The valid range is:
  - Minimum—1
  - Maximum—999999999

For a SIP call, the Oracle Communications Session Border Controller replaces the cause code in the Reason header of the SIP response.

For a H.323 call, the Oracle Communications Session Border Controller converts the cause code to a Q.931 cause code in the Q.931 Cause IE in the releaseComplete message.

## Enabling Enhanced Call Gapping

Enhanced NMC call gapping uses new configuration parameters to the network management controls configuration:

- **gap-rate-max-count**—Maximum count that triggers the application of network management control rule if it is exceeded. The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—999999999
- **gap-rate-window-size**—Length of time in seconds used for the gapping rate calculation. The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—999999999

For this feature to behave as intended, you also need to set the **type** parameter to **gap-rate**, and set the **value** parameter to the maximum sustained rate of calls that you want to support.

To configure NMC call gapping enhancements:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **net-management-control** and press Enter.

```
ACMEPACKET(session-router)# net-management-control
```

To add support to a pre-existing network management control configuration, use the ACLI **select** command to choose the configuration you want to edit.

- **gap-rate-max-count**—Maximum count that triggers the application of network management control rule if it is exceeded. The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—999999999

Along with the current time, the last time of a gapped call, the call counter value, the CPS value, and the gap-rate-window-size value, the Oracle Communications Session Border Controller uses **gap-rate-max-count** as a measurement to determine if a control rule will be applied.

- **gap-rate-window-size**—Length of time in seconds used for the gapping rate calculation. The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—999999999

Along with the current time, the last time of a gapped call, the call counter value, and the CPS value, the Oracle Communications Session Border Controller uses the **gap-rate-window-size** value to calculate whether the maximum count is within allowable limits.

1. Save and activate your configuration.

## Applying a Network Management Control Rule to a Realm

Once you have configured network management control rules, you can enable their use on a per-realm basis.

To apply a network management control rule to a realm:

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1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager  
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config  
ACMEPACKET (realm-config) #
```

If you are enabling network management controls for a pre-existing realm, then you must select (using the ACLI **select** command) the realm that you want to edit.

4. **net-management-control**—Set this parameter to **enabled** to apply network control rules in this realm. The default value is **disabled**. The valid values are:

- enabled | disabled

5. Save and activate your configuration.

## 3147 - Emergency Call Fallback

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You can enable the rejection of priority calls per a configured SIP interface. When this option is enabled, the Oracle Communications Session Border Controller does not allow priority calls through that SIP interface, and the call is rejected with a 380 response.

If the send-380-response parameter in the SIP interface is configured, this string is provided as a reason in the 380 message. If this parameter is not populated, the default message priority calls not allowed is sent.

### Emergency Call Fallback Configuration

To enable emergency call fallback, you must select an option for the SIP-interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-interface  
ACMEPACKET (sip-interface) #
```

4. **options**—Type **disallow-priority-calls** to send a 380 response to priority calls.

```
ACMEPACKET (session-router) # options disallow-priority-calls
```

5. Save and activate your configuration.

## Accounting Configuration for QoS

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This section explains how to configure the account configuration and account servers so you can use the Oracle Communications Session Border Controller in conjunction with external RADIUS (accounting) servers to generate CDRs and provide billing services requires.

### QoS Accounting Configuration

To configure the account configuration and account servers:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **account-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # account-config
ACMEPACKET (account-config) #
```

4. To configure account server parameters (a subset of the account configuration parameters, type **account-servers** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (account-config) # account-servers
ACMEPACKET (account-server) #
```

The following example shows both the account config and account server parameters.

```
account-config
  hostname                acctserver1
  port                    1813
  strategy                 Hunt
  state                   enabled
  max-msg-delay            60
  max-wait-failover       100
  trans-at-close           disabled
  generate-start           OK
  generate-interim         OK
                          Reinvite-Response
account-server
  hostname                192.168.2.2
  port                    1813
  state                   enabled
  min-round-trip          100
  max-inactivity           100
  restart-delay           100
  bundle-vsa              enabled
  secret                  testing
  NAS-ID                  acme-accounting
last-modified-date       2005-01-15 02:23:42
```

## Account Configuration

You set the account configuration parameters to indicate where you want accounting messages sent, when accounting messages you want them sent, and the strategy you want used to select account servers.

To configure the account configuration:

1. **hostname**—Enter a name for the host associated with the Oracle Communications Session Border Controller in hostname (FQDN) format. The default value is the name of the local host.

The value you enter here must match the configured physical interface's operation type **control** or **maintenance**, to determine on which network to send RADIUS messages.

2. **port**—Enter the number of the UDP port associated with the Oracle Communications Session Border Controller from which RADIUS messages are sent. The default value is **1813**. The valid range is:

- Minimum—1025
- Maximum—65535

3. **strategy**—Indicate the strategy you want used to select the accounting servers to which the Oracle Communications Session Border Controller will send its accounting messages. The default value is **hunt**. The following table lists the available strategies:
- **hunt**—Selects accounting servers in the order in which they are listed.  
If the first accounting server is online, working, and has not exceeded any of the defined constraints, all traffic is sent to it. Otherwise the second accounting server is selected. If the first and second accounting servers are offline or exceed any defined constraints, the third accounting server is selected. And so on through the entire list of configured servers
  - **failover**—Uses the first server in the list of predefined accounting servers until a failure is received from that server. Once a failure is received, it moves to the second accounting server in the list until a failure is received. And so on through the entire list of configured servers.
  - **roundrobin**—Selects each accounting server in order, distributing the selection of each accounting server evenly over time.
  - **fastestrtt**—Selects the accounting server that has the fastest round trip time (RTT) observed during transactions with the servers (sending a record and receiving an ACK).
  - **fewestpending**—Selects the accounting server that has the fewest number of unacknowledged accounting messages (that are in transit to the Oracle Communications Session Border Controller).
4. **state**—Enable this parameter if you want the account configuration active on the system. Disable it if you do not want the account configuration active on the system. The default value is **enabled**. The valid values are:
- enabled | disabled
5. **max-msg-delay**—Indicate the length of time in seconds that you want the Oracle Communications Session Border Controller to continue trying to send each accounting message. During this delay, the Oracle Communications Session Border Controller can hold a generic queue of 4096 messages. The default value is **60**.
- Minimum—zero (0)
  - Maximum—4294967295
6. **max-wait-failover**—Indicate the maximum number of accounting messages the Oracle Communications Session Border Controller can store its message waiting queue for a specific accounting server, before it is considered a failover situation.
- Once this value is exceeded, the Oracle Communications Session Border Controller attempts to send its accounting messages, including its pending messages, to the next accounting server in its configured list. The default value is **100**. The valid range is:
- Minimum—1
  - Maximum—4096
7. **trans-at-close**—Disable this parameter if you do not want to defer the transmission of message information to the close of a session. Enable it if you want to defer message transmission. The default value is **disabled**. The valid values are:
- **disabled**—The Oracle Communications Session Border Controller transmits accounting information at the start of a session (Start), during the session (Interim), and at the close of a session (Stop). The transmitted accounting information for a single session might span a period of hours and be spread out among different storage files.
  - **enabled**—Limits the number of files on the Oracle Communications Session Border Controller used to store the accounting message information for one session. It is easiest to store the accounting information from a single session in a single storage file.
8. **generate-start**—Select the type of SIP event that triggers the Oracle Communications Session Border Controller to transmit a RADIUS Start message. The default value is **is ok**. The valid values are:
- **start**—RADIUS Start message should not be generated
  - **invite**—RADIUS Start message should be generated once the Oracle Communications Session Border Controller receives a SIP session INVITE.

- **ok**—RADIUS Start message is generated once the Oracle Communications Session Border Controller receives an OK message in response to an INVITE.
9. **generate-interim**—Retain the default value **reinvite-response** to cause the Oracle Communications Session Border Controller to transmit a RADIUS Interim message. (A RADIUS Interim message indicates to the accounting server that the SIP session parameters have changed.)

To disable interim message generation, enter a pair of quotes as the value for this parameter. Otherwise, select one or more than one of the following values:

- **ok**—RADIUS Interim message is generated when the Oracle Communications Session Border Controller receives an OK message in response to an INVITE.
  - **reinvite**—RADIUS Interim message is generated when the Oracle Communications Session Border Controller receives a SIP session reINVITE message.
  - **reinvite-response**—RADIUS Interim message is generated when the Oracle Communications Session Border Controller receives a SIP session reINVITE and responds to it (for example, session connection or failure).
  - **reinvite-cancel**—RADIUS Interim message is generated when the Oracle Communications Session Border Controller receives a SIP session reINVITE, and the Reinvite is cancelled before the Oracle Communications Session Border Controller responds to it.
10. **account-server**—Create the account server list to store accounting server information for the account configuration. Each account server can hold 100 accounting messages.

Account server entries are specific to the account configuration. They cannot be viewed or accessed for editing outside of the account configuration.



**Note:** RADIUS will not work if you do not enter one or more servers in a list.

## Account Server

You must establish the list of servers to which the Oracle Communications Session Border Controller can send accounting messages.

1. **hostname**—Name of the host associated with the account server as an IP address.
2. **port**—Enter the number of the UDP port associated with the account server to which RADIUS messages are sent. The default value is **1813**. The valid range is:
  - Minimum—1025
  - Maximum—65535
3. **state**—Enable or disable the account servers on the system. The default value is **enabled**. The valid values are:
  - enabled | disabled
4. **min-round-trip**—Indicate the minimum round trip time of an accounting message in milliseconds. The default value is **250**. The valid range is:
  - Minimum—10
  - Maximum—5000

A round trip consists of the following:

The system sends an accounting message to the account server.

The account server processes this message and responds back to the Oracle Communications Session Border Controller.

If the **fastest RTT** is the **strategy** for the account configuration, the value you enter here can be used to determine an order of preference (if all the configured account servers are responding in less than their minimum RTT).

5. **max-inactivity**—Indicate the length of time in seconds that you want the Oracle Communications Session Border Controller with pending accounting messages to wait when it has not received a valid response from the target account server. The default value is **60**. The valid range is:

- Minimum—1
- Maximum—300

Once this timer value is exceeded, the Oracle Communications Session Border Controller marks the unresponsive **account server** as disabled in its failover scheme. When a server connection is marked as inactive, the Oracle Communications Session Border Controller attempts to restart the connection and transfers pending messages to another queue for transmission. RADIUS messages might be moved between different **account servers** as servers become inactive or disabled.

6. **restart-delay**—Indicate the length of time in seconds you want the Oracle Communications Session Border Controller to wait before resending messages to a disabled account server. The default value is **30**. The valid range is:

- Minimum—1
- Maximum—300

7. **bundle-vs-a**—Retain the default **enabled** if you want the account server to bundle the VSAs within RADIUS accounting messages. Enter **disabled** if you do not want the VSAs to be bundled. (Bundling means including multiple VSAs within the vendor value portion of the message.) The valid values are:

- enabled | disabled

In a bundled accounting message, the RADIUS message type is vendor-specific, the length is determined for each individual message, and the vendor portion begins with a 4-byte identifier, and includes multiple vendor type, vendor length, and vendor value attributes.

8. **secret**—Enter the secret passed from the account server to the client in text format. Transactions between the client and the RADIUS server are authenticated by the shared secret; which is determined by the source IPv4 address of the received packet.

9. **NAS-ID**—Enter the NAS ID in text format (FQDN allowed). The account server uses this value to identify the Oracle Communications Session Border Controller for the transmittal of accounting messages.

The remote server to which the **account configuration** sends messages uses at least one of two potential pieces of information for purposes of identification. The Oracle Communications Session Border Controller accounting messages always includes in the first of these:

- Network Access Server (NAS) IP address (the IP address of the Oracle Communications Session Border Controller's SIP proxy)
- **NAS ID** (the second piece of information) provided by this value. If you enter a value here, the NAS ID is sent to the remote server.

## Whitelists for SIP

---

Oracle Communications Session Border Controller by default ignores and passes-through unknown SIP headers and URI parameters. However, some operators require that the Oracle Communications Session Border Controller only accept messages with headers and URI parameters complying with those supported by their internal equipment. This section describes the use of whitelists to control unknown headers and parameters in request and response traffic.

### What is a Whitelist

A whitelist is an approved list of entities for which equipment provides particular privileges, access, and recognition. The Oracle Communications Session Border Controller can use configured whitelist profiles to control and accept specific inbound SIP headers and URI parameters that are being passed-through the Oracle Communications Session Border Controller. When you configure this feature, the Oracle Communications Session Border Controller rejects requests not matching the configured profile, or removes the unspecified headers or URI parameters not in the configured profile.



## Whitelists Configuration

You can configure whitelist profiles or rules that allow the Oracle Communications Session Border Controller to only accept inbound SIP headers and URI parameters that are configured in this whitelist, using the parameter **allowed-elements-profile**. You can configure the settings for this parameter using the ACLI interface at `session-router>enforcement-profile`. Since the **enforcement-profile** object also pertains to session agents, realms, and SIP interfaces, you can also apply the profiles you configure to these remote entities using the ACLI interface at `session-router>session-agent`, `session-router>sip-interface`, and `media-manager>realm-config`.

In the following configuration example, it is assumed that your baseline configuration passes SIP traffic, with the Oracle Communications Session Border Controller in the role of an Access SBC. Use this procedure to configure a whitelist for the session router and optionally apply the specific whitelists to the session agent and SIP interface, as well as the media manager's realm configuration.

To configure a whitelist for the session router:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **allowed-elements-profile** and press Enter to access the enforcement profile-related parameters.

```
ACMEPACKET(session-router)# allowed-elements-profile
ACMEPACKET(allowed-elements-profile)#
```

4. Type **allowed-elements-profile** and press Enter to access the whitelist-related parameters. The system prompt changes to let you know that you can begin configuring individual parameters for this object.

```
ACMEPACKET(enforcement-profile)# allowed-elements-profile
ACMEPACKET(allowed-elements-profile)#
```

5. **name** — Enter a unique name for the whitelist you are creating. This name can be referenced when configuring the enforcement-profiles for session-agent, SIP interface, and realm-config.

```
ACMEPACKET(allowed-elements-profile)# name whitelist1
```

6. **description** — Enter a description that explains the purpose of creating this whitelist. You can use any alpha-numeric characters when entering the description.

```
ACMEPACKET(allowed-elements-profile)# description Basic Whitelist
```

7. Type **rule-sets** and press Enter to specify the rules to match against specific incoming SIP headers and/or URI parameters. The system prompt changes to let you know that you can begin configuring individual parameters for this object.

```
ACMEPACKET(allowed-elements-profile)# rule-sets
ACMEPACKET(rule-sets)#
```

8. **unmatched-action** — Select the action for the Oracle Communications Session Border Controller to perform when a header does not exist in an incoming message. The default value is **reject**. The valid values are:

- **reject** — Rejects all incoming messages that do not contain a header.
- **delete** — Deletes all incoming message that do not contain a header.



**Note:** This parameter applies to non-matching header names only (not non-matching URI parameters).

```
ACMEPACKET(rule-sets)# unmatched action delete
```

9. **msg-type** — Specify the type of messages for which the Oracle Communications Session Border Controller applies this whitelist configuration. The default value is **any**. The valid values are:

- **any** — Applies to all incoming messages.
- **request** — Applies to only incoming REQUEST messages.

- **response** — Applies to only incoming RESPONSE messages.

```
ACMEPACKET(rule-sets)# msg-type any
```

- 10. methods** — Enter the packet method(s), separated by a comma, for which this whitelist is enforced. Packet methods include, INVITE, OPTIONS, ACK, BYE, etc. If this field is left blank, the whitelist applies to all packet methods. You can enter up to a maximum of 255 characters.

```
ACMEPACKET(rule-sets)# methods INVITE,ACK,BYE
```

- 11. logging** — Select whether or not an incoming message is written to a log file called **matched.log** when the message contains an element not specified in the whitelist. The default value is **disabled**. The valid values are:

- enabled | disabled

```
ACMEPACKET(rule-sets)# logging enabled
```

The **matched.log** contains information about the timestamp, received/sent Oracle Communications Session Border Controller network-interface, IP address/port from which it was received or being sent from, and which peer IP address/port it was received from or sent to. The log also specifies the request URI (if applicable), and the From, To, and Contact headers in the message, as well as which rule triggered the log action. An example of the log output of the **matched.log** file is as follows:

```
Dec 17 14:17:54.526 On [0:0]192.168.1.84:5060 sent to 192.168.1.60:5060
allowed-elements-profile[whitelist1(reject)]
INVITE sip:service@192.168.1.84:5060 SIP/2.0
From: sipp <sip:+2125551212@192.168.1.60:5060>;tag=3035SIPpTag001
To: sut <sip:service@192.168.1.84>
Contact: sip:sipp@192.168.1.60:5060
```

The **header-rule** object consists of 6 parameters that make up the header-rule:

- header-name
- unmatched-action
- allow-header-param
- allow-uri-param
- allow-uri-user-param
- allow-uri-header-name

You can configure an unlimited number of header-rules on the Oracle Communications Session Border Controller that you can apply to your network. Use the following parameters to configure a header-rule that the Oracle Communications Session Border Controller uses to control which incoming messages it allows.

- 12. header-rule** — This object allows you to configure, as part of the whitelist, multiple parameters which make up the header rule that the Oracle Communications Session Border Controller allows from incoming messages. Header-rules do NOT have to be in any specific order. The system prompt changes to let you know that you can begin configuring individual parameters for this object.

```
ACMEPACKET(rule-set)# header-rule
ACMEPACKET(allowed-header-rule)#
```

- 13. header-name** — Enter the name of the header in the whitelist that the Oracle Communications Session Border Controller allows from incoming messages. It is case-insensitive and supports abbreviated forms of header names. For example, “Via”, “via”, or “v” all match against the same header. A header name of “request-uri” refers to the request URI of requests, while a header name of \* applies to any header-type not matched by any other header-rule. The default value is \*. This default value provides the ability to have header-rules for commonly known headers that remove unknown parameters, but leave unknown headers alone.

```
ACMEPACKET(allowed-header-rule)# header-name Contact
```

- 14. unmatched-action** — Select the action for the Oracle Communications Session Border Controller to perform when an incoming header’s parameters do not match the relevant allowed parameters specified for this header-name. The default value is **reject**. The valid values are:

- **reject** — Rejects all incoming messages that have header parameters that do not match the parameters specified in this header-name.

- **delete** — Deletes all incoming messages that have header parameters that do not match the parameters specified in this header-name.



**Note:** This parameter applies to non-matching header names only (not non-matching URI parameters).

```
ACMEPACKET (allowed-header-rule) # unmatched-action delete
```

- 15. allow-header-param** — Enter the header parameter that the Oracle Communications Session Border Controller allows from the headers in incoming messages. You can enter up to 255 characters, including a comma (,), semi-colon (;), equal sign (=), question mark (?), at-symbol (@), backslash (\), or plus sign (+). The default value is \*, which allows all header parameters to pass through. If you leave this field empty, no header parameters are allowed.

```
ACMEPACKET (allowed-header-rule) # allow-header-param *
```

- 16. allow-uri-param** — Enter the URI parameter that the Oracle Communications Session Border Controller allows from the headers in incoming messages. You can enter up to 255 characters, including a comma (,), semi-colon (;), equal sign (=), question mark (?), at-symbol (@), backslash (\), or plus sign (+). The default value is \*, which allows all URI parameters to pass through. If you leave this field empty, no URI parameters are allowed.

```
ACMEPACKET (allowed-header-rule) # allow-uri-param *
```

- 17. allow-uri-user-param** — Enter the URI user parameter that the Oracle Communications Session Border Controller allows from the headers in incoming messages. You can enter up to 255 characters, including a comma (,), semi-colon (;), equal sign (=), question mark (?), at-symbol (@), backslash (\), or plus sign (+). The default value is \*, which allows all URI user parameters to pass through. If you leave this field empty, no URI user parameters are allowed.

```
ACMEPACKET (allowed-header-rule) # allow-uri-user-param *
```

- 18. allow-uri-header-name** — Enter the URI header name that the Oracle Communications Session Border Controller allows from the headers in incoming messages. You can enter up to 255 characters, including a comma (,), semi-colon (;), equal sign (=), question mark (?), at-symbol (@), backslash (\), or plus sign (+). The default value is \*, which allows all URI header name parameters to pass through. If you leave this field empty, no URI header name parameters are allowed.

```
ACMEPACKET (allowed-header-rule) # allow-uri-header-name *
```

- 19.** Save your work using the ACLI **done** command.

## Configuration Exception

There are specific instances in incoming Request-URI messages where the Oracle Communications Session Border Controller ignores specific parameters and automatically adds header-rules.

In a Request-URI, all parameters are URI parameters, and URI headers are not allowed. If you define values for the “allow-header-param”, “allow-uri-header-name”, and “allow-uri-param”, the Oracle Communications Session Border Controller ignores these parameters in the Request-URI. Instead the Oracle Communications Session Border Controller automatically adds header-rules for incoming “Via”, “From”, “To”, “Call-ID”, and “CSeq” messages. These are explicit header rules and cannot be deleted. Each header-rule in a Request-URI has parameters populated with the value of \*. If required, a user can change the header-rule parameter values with the values identified in the following table.

Header Rule	Applicable Parameter	Required Value(s)
Via	allow-header-param	<ul style="list-style-type: none"> <li>• branch</li> <li>• received</li> <li>• rport</li> </ul>
From	allow-header-param	<ul style="list-style-type: none"> <li>• tag</li> </ul>
To	allow-header-param	<ul style="list-style-type: none"> <li>• tag</li> </ul>

Header Rule	Applicable Parameter	Required Value(s)
Call-ID	allow-header-param	No restrictions
CSeq	allow-header-param	No restrictions

### Verify Whitelist Configuration

After you have configured and saved a whitelist on the Oracle Communications Session Border Controller, you can use the existing `verify-config` command at the top level prompt to verify the saved configuration. For example:

```
ACMEPACKET# verify-config
```

This command checks for errors in the Oracle Communications Session Border Controller configuration. Whitelist configuration errors, specifically related to the enforcement-profile object, also display in the output of this command if applicable. The whitelist configuration error(s) display if any references to the allowed-element-profiles are improperly configured. If an error(s) exist, the following message displays:

```
-----  
ERROR: enforcement-profile [ep] contains a reference to an allowed-  
enforcement-profile [abc] that does not exist  
-----
```

### How Whitelists Work

Whitelists allow you to customize which SIP signaling messages to allow into your network and which messages to reject or delete. In the flow of SIP traffic to/from the Oracle Communications Session Border Controller, the Oracle Communications Session Border Controller matches any received request or response, in or out of a dialog against the configured allowed list, and rejects or deletes the non-matching element based on the actions specified in the whitelist configuration.

For responses, the Oracle Communications Session Border Controller does not reject the message if a header or parameter is not found in the allowed list, even if the action is set to reject. Instead it deletes the offending parameter or header. In addition, if the message is a request of the method type ACK, PRACK, CANCEL or BYE, it deletes all unmatched elements, but does not reject the request, even if the action was configured to reject.

The whitelist verification performs for any method; however you can narrow this list to operate only on specific methods by defining them in the **methods** parameter of the configuration.

Whitelist verification occurs when a request or response is received by the Oracle Communications Session Border Controller, but only after the Oracle Communications Session Border Controller has processed the inbound header manipulation rule (HMR), network management controls (NMC), Resource-Priority header (RPH), and monthly-minutes checking.

The Oracle Communications Session Border Controller responds to requests which have non-matching headers or parameters configured with an action of reject, with a "403 Forbidden" response by default. You can use a local-response event, **allowed-elements-profile-rejection**, to override the default reject status code and reason phrase.

The configured whitelist operates transparently on headers that have multiple URIs or multiple header values within a single header (header values separated by a comma).

Parameter parsing operates only on parameters that it can identify. For parameters that can not be parsed, for example an invalid URI (e.g. < sip:user@host.com&hp=val>), the Oracle Communications Session Border Controller ignores this URI header parameter value of "hp" since it is not contained within a valid URI. Even though it would appear to be a URI header parameter, URI headers must come after URI parameters. Parameter matching does not occur if the headers and parameters in the URI are not well-formed. The Oracle Communications Session Border Controller does not remove the parameter since it cannot identify it.

## Whitelist Learning

You can build your whitelist configuration based on the learning capabilities of the Oracle Communications Session Border Controller. When you enable the Oracle Communications Session Border Controller learning mode, it acquires the knowledge of the allowable elements (headers and parameters) currently incoming to your network. The Oracle Communications Session Border Controller collects the information about the headers received and the parameters that exist within each header. The information continues to be gathered until you disable the learning mode.

Once you disable the learning mode, the Oracle Communications Session Border Controller prompts you to enter a name for the allowed-elements-profile. If the profile name you entered does not exist, the captured information is written to the new allowed-elements-profile configuration. The administrator can then make changes to the configuration as applicable, save the configuration, and apply it to a logical remote entity.

The new allowed-elements-profile does not contain any wildcard rules. The Oracle Communications Session Border Controller cannot generate wildcard headers and parameters during the learning mode. The Methods object is populated from the list of methods seen by the Oracle Communications Session Border Controller while learning.



**Note:** Oracle recommends running the learning mode during off-peak and/or light traffic times. This mode can operate in conjunction with the execution of an allowed-elements-profile. The learning occurs just before any configured allowed-elements-profile configuration.

## Whitelist Learning Configuration

The ACLI interface provides two commands that allow a Superuser to start and stop whitelist learning on the Oracle Communications Session Border Controller:

Command	Description
start <argument> <options>	<p>Starts whitelist learning on the Oracle Communications Session Border Controller.</p> <p>You must specify the argument learn-allowed-elements with this command to start the learning operation.</p> <p>Optionally, you can use method, msg-type, and params after the argument.</p>
stop <argument> <identifier>	<p>Stops the whitelist learning on the Oracle Communications Session Border Controller and writes the learned configuration to the editing configuration on the Oracle Communications Session Border Controller where it is saved and activated.</p> <p>You must specify the argument learn-allowed-elements with this command to stop the learning operation.</p> <p>You must specify a unique identifier that identifies the allowed-elements-profile name.</p> <p>If you specify an identifier name that already exists as a profile, the ACLI returns an error message and prompts you to enter a different name.</p>

You can use these commands at the top level ACLI prompt as required on the Oracle Communications Session Border Controller.

You use these commands with the argument, learn-allowed-elements to start/stop the whitelist learning feature. By default, the learning mode creates a single rule-set under which all of the headers and their respective parameters are stored.

For example:

## Admission Control and QoS

---

```
ACMEPACKET# start learn-allowed-elements
Learning mode for allowed-elements-profile started.
```

In the above example, “**start**” is the top level CLI command and `learn-allowed-elements` is the operation being performed.

Optionally, you can specify `[method]`, `[msg-type]`, and `[params]` in any order, for the Oracle Communications Session Border Controller to learn specific rule-set elements from incoming messages and save them to the whitelist configuration.

For example:

```
ACMEPACKET# start learn-allowed-elements method msg-type params
```

The `method` option creates a new rule-set per unique method. The “`msg-type`” option creates a new rule-set per unique message-type seen. The “`params`” option performs URI and header parsing to examine parameters within the message. By default, parameter parsing is disabled.

To start the whitelist learning feature:

In Superuser mode, at the top level CLI prompt, type `start learn-allowed-elements` and press Enter.

```
ACMEPACKET# start learn-allowed-elements
```

The following message displays:

```
Learning mode for allowed-elements-profile started.
```


To specify the elements of rule-sets for whitelists:

In Superuser mode, at the top level CLI prompt, type `start learn-allowed-elements method msg-type params` and press Enter.

```
ACMEPACKET# start learn-allowed-elements method msg-type params
```

The following message displays:

```
Learning mode for allowed-elements-profile started.
```

 **Note:** If you try to start a whitelist learning operation while another learning operation is already running, the following message displays:

```
Learning mode restarted without saving
Learning mode for allowed-elements-profile started.
```

To stop the whitelist learning feature:

In Superuser mode, at the top level CLI prompt, type `stop learn-allowed-elements <identifier>`, where `<identifier>` is the allowed-elements-profile name, and press Enter.

```
ACMEPACKET# stop learn-allowed-elements whitelists1
```

The following message displays:

```
Learning mode for allowed-elements-profile stopped.
```

If you specify an identifier name that already exists as a profile, the CLI returns an error message and prompts you to enter a different name.

## Rejected Messages Monitoring

---

Whitelists, when configured on the Oracle Communications Session Border Controller, control whether or not the Oracle Communications Session Border Controller allows unknown headers and URI parameters to be accepted in incoming request and response traffic. When the Oracle Communications Session Border Controller rejects messages according to the whitelist, the rejected messages are logged to a file called “`matched.log`” if logging is set to enabled. You can open and view the log when required to view the rejected messages.

In addition to the rejected messages being logged to the “matched.log” file, the rejected messages are also sent through a burst counter that keeps track of the amount of messages rejected. You can enter the show sipd to display the number of rejected messages. The counter is titled Rejected Message.






## Static Flows

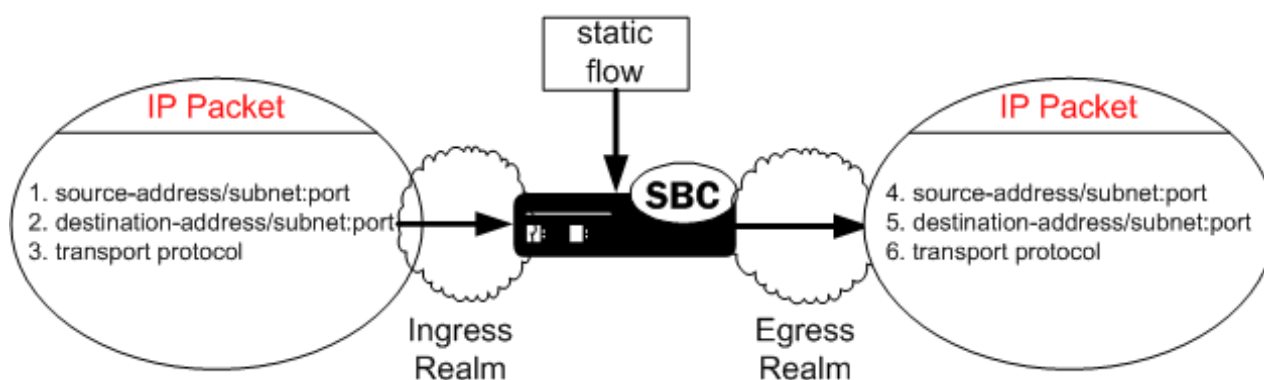
This chapter describes the Oracle Communications Session Border Controller's static flows feature. Static flows allow network traffic that matches specific criteria to pass through the Oracle Communications Session Border Controller unrestricted. Static flows are unidirectional. This feature lets you steer traffic toward a particular destination based on its original characteristics. Static flows can range from being widely accessible to very restrictive, depending on the values you establish. Static flows are used for transporting a variety of signaling messages through the Oracle Communications Session Border Controller to achieve vendor interoperability. The Oracle Communications Session Border Controller supports the following types of Static flows:

- IPv6 to IPv6 flows
- IPv4 to IPv6 flows
- IPv4 to IPv4 flows

 **Note:** Traffic that traverses the Oracle Communications Session Border Controller in two directions, such as ICMP requests and responses, requires static flows configured for both directions.

### About Static Flows

The static flow element explicitly writes entries into the IP routing table. These entries are persistent and are not deleted as calls are set up and broken down. Refer to the following diagram to understand how a static flow works.



A static flow entry watches for traffic with specific criteria on a specified ingress realm; that traffic consists of the following criteria:

- The packet enters the Oracle Communications Session Border Controller on the specified ingress realm.
- The packet contains matching source address, subnet, and port criteria, field 1.

## Static Flows

---

- The packet contains matching destination address, subnet, and port criteria, field 2.
- The packet contains a matching transport protocol, field 3.

If the above conditions are met, then the Oracle Communications Session Border Controller does the following:

- The IPv4 traffic is forwarded out of the Oracle Communications Session Border Controller on the specified egress realm.
- The configured source address, subnet, and port criteria are written to the exiting packet, field 4.
- The configured destination address, subnet, and port criteria are written to the exiting packet, field 5.
- The original transport protocol and its contents remain unchanged as the packet exits into the egress realm.

## IPv6 / IPv4 Translations

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The ingress or egress traffic type, whether IPv4 or IPv6, must match the configuration of the realm where attached, as **in-realm-id** or **out-realm-id**. A realm and IP version configuration mismatch results in an error message and log entry at error level.

IPv6 to IPv4 flows exit the Oracle Communications Session Border Controller with the prefix `::fff:0:0:0/96`. They may be written as `::fff:0:a.b.c.d`, where a.b.c.d refers to an IPv6-enabled node.

While IPv4 addresses can be translated into IPv6 addresses, IPv6 address can not be translated to IPv4.

## About Network Address Translation ALG

---

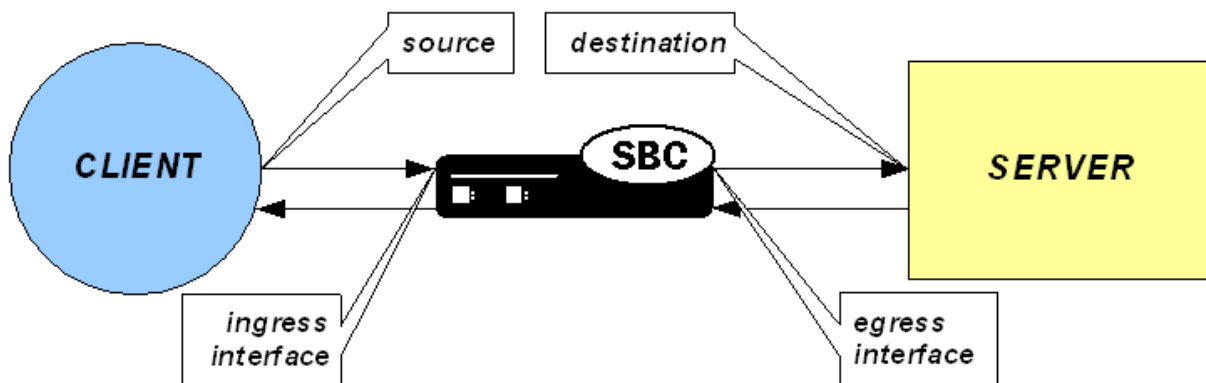
The Oracle Communications Session Border Controller supports Network Address and Port Translation (NAPT) and Trivial File Transfer Protocol (TFTP) functionality over media interfaces, collectively known as Network Address Translation (NAT) ALG. The NAT ALG feature is implemented as an extension of the static flow feature.

In some applications, the Oracle Communications Session Border Controller acts as an intermediary device, positioned between endpoints located in an access network and application servers located in a backbone network. The Oracle Communications Session Border Controller's NAT ALG feature enables these endpoints to use non-VoIP protocols, such as TFTP and HTTP, to access servers in a provider's backbone network to obtain configuration information.

NAT ALG parameters support RTC and can be dynamically reconfigured. The active NAT ALG configuration can be replicated on the standby SD in an HA configuration.

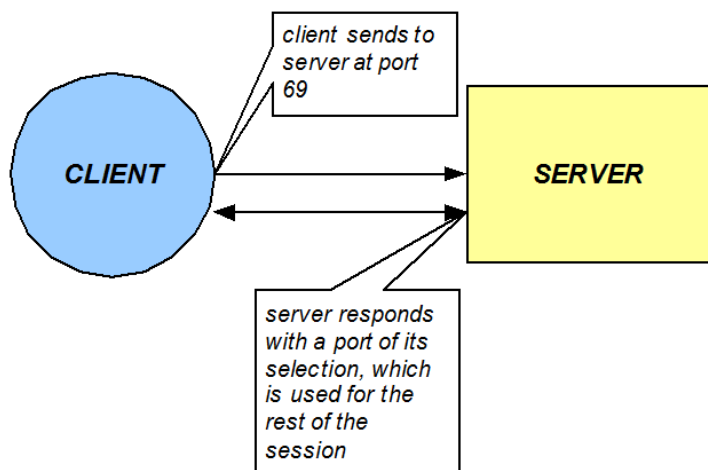
### NAPT

The NAPT ALG functionality is the same as that found in commercially available enterprise and residential NAT devices. The Oracle Communications Session Border Controller watches for packets entering a media interface that match source and destination IP address criteria. Matching packets are then redirected out of the egress interface, through a specified port range, toward a destination address.

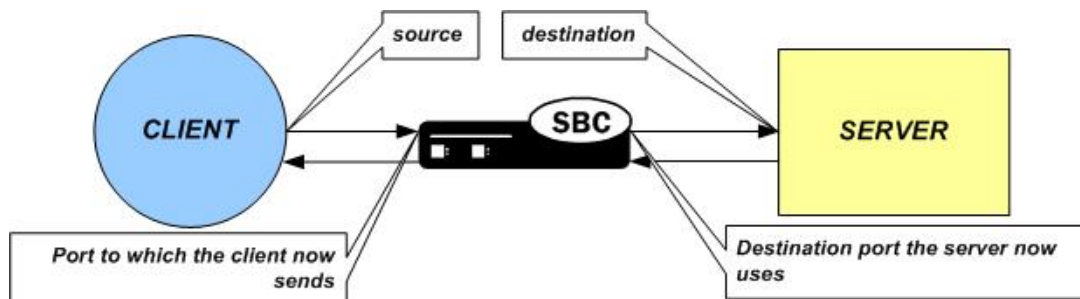


## TFTP

The TFTP ALG is implemented as an extension of the NAT ALG. It works slightly differently than traditional NAT. In a TFTP session, the first packet is sent from a source endpoint to port 69 on the TFTP server. The TFTP server responds from another port. This port, from which the TFTP response originates, is used for the remainder of the TFTP session.



To act as a TFTP ALG, the Oracle Communications Session Border Controller will latch on the first return packet from the server to learn the server's port. The ingress-side destination port of the Oracle Communications Session Border Controller is changed to reflect the new communications port for the TFTP session. This process takes place without any user intervention.



## Configuring Static Flows

This section explains how to configure static flows. It also provides sample configurations for your reference. You can configure static flows with or without NAT ALG. If you configure static flows with NAT ALG, you can choose NAT or TFTP as the ALG type.

### Basic Static Flow Configuration Overview

This section outlines the basic static flow configuration, without NAT ALG. You configure static flows by specifying ingress traffic criteria followed by egress re-sourcing criteria.

When configuring static flows, the following conventions are used:

- An address of 0.0.0.0 matches all addresses. This token is used as the wildcard for both IPv4 and IPv6 static flows
- Enclose the address portion of an IPv6 address in brackets: [7777::11]/64:5000
- Not specifying a port implies all ports.
- Not specifying a subnet mask implies a /32, matching for all 32 bits of the IPv4 address, or a /128 matching for all 128 bits of the IPv6 address.

## Static Flows

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1. Set the static flows' incoming traffic-matching criteria. First set the ingress realm where you expect to receive traffic that will be routed via a static flow. Second, set the traffic's source IP address, source subnet, and source port or port range criteria. Third, set the traffic's destination IP address, destination subnet, and destination port criteria. This is usually an external address on the Oracle Communications Session Border Controller.
2. Set the criteria that describes how traffic should be translated on the egress side of the Oracle Communications Session Border Controller. First set the egress realm where you want to send the traffic to be routed by this static flow. Second, set the traffic's source IP address, source subnet, and source port or port range criteria. This is usually an external address on the Oracle Communications Session Border Controller. Third, set the traffic's destination IP address, destination subnet, and destination port criteria.
3. Set the protocol this static flow entry acts upon. This type of packet, as the payload of the IP packet, remains untouched as traffic leaves the Oracle Communications Session Border Controller. Specifying a layer 4 protocol here acts as another criteria to filter against for this static flow.

The combination of entries in the ingress realm, ingress source address, ingress destination address, and protocol fields must be unique. For bidirectional traffic, you need to define a separate static flow in the opposite direction.

## Static Flow Configuration

This section describes how to configure the **static-flow** element using the ACLI.

The ingress IP address criteria is set first. These parameters are applicable to traffic entering the ingress side of the Oracle Communications Session Border Controller.

- **in-realm-id**—The access realm, where endpoints are located.
- **in-source**—The source network in the access realm where the endpoints exist. This parameter is entered as an IP address and netmask in slash notation to indicate a range of possible IP addresses.
- **in-destination**—The IP address and port pair where the endpoints send their traffic. This is usually the IP address and port on a Oracle Communications Session Border Controller physical interface that faces the access realm.

The egress IP address criteria is entered next. These parameters determine how traffic is re-sourced as it leaves the Oracle Communications Session Border Controller and enters the backbone network.

- **out-realm-id**—The backbone realm, where servers are located.
- **out-source**—The IP address on the physical interface of the Oracle Communications Session Border Controller where traffic exits the Oracle Communications Session Border Controller into the backbone realm. Do not enter a port for this parameter.
- **out-destination**—The IP address and port pair destination of the traffic. This is usually a server in the backbone realm.
- **protocol**—The protocol associated with the static flow. The protocol you choose must match the protocol in the IPv4 header. Valid entries are TCP, UDP, ICMP, ALL.

The type of NAT ALG, if any.

- **alg-type**—The type of NAT ALG. Set this to NAPT, TFTP, or none.

The port range for port re-sourcing as traffic affected by the NAT ALG exits the egress side of the Oracle Communications Session Border Controller is set next. (Not applicable if **alg-type** is set to none.)

- **start-port**—The starting port the NAT ALG uses as it re-sources traffic on the egress side of the Oracle Communications Session Border Controller.
- **end-port**—The ending port the NAT ALG uses as it re-sources traffic on the egress side of the Oracle Communications Session Border Controller.

The flow timers are set next. (Not applicable if **alg-type** is set to none.)

- **flow-time-limit**—Total session time limit in seconds. The default is 0; no limit.



**Note:** Note that the static flow-time-limit must have a value larger than initial-guard-timer and subseq-guard-timer for static flows.

- **initial-guard-timer**—Initial flow guard timer for an ALG dynamic flow in seconds. The default is 0; no limit.
- **subsq-guard-timer**—Subsequent flow guard timer for an ALG dynamic flow in seconds. The default is 0; no limit.

Finally, you can set the optional bandwidth policing parameter for static flows (with or without NAT ALG applied).

- **average-rate-limit**—Sustained rate limit in bytes per second for the static flow and any dynamic ALG flows. The default is 0; no limit.

To configure static flow:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the **media-manager** path.

```
ACMEPACKET(configure)# media-manager
```

3. Type **static-flow** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# static-flow
```

From this point, you can configure media policing parameters.

4. **in-realm-id**—Enter the ingress realm or interface source of packets to match for static flow translation. This in-realm-id field value must correspond to a valid identifier field entry in a **realm-config**. This is a required field. Entries in this field must follow the Name Format.
5. **in-source**—Enter the incoming source IP address and port of packets to match for static flow translation. IP address of 0.0.0.0 matches any source address. Port 0 matches packets received on any port. The port value has no impact on system operation if either ICMP or ALL is the selected protocol. This parameter takes the format:

```
in-source <ip-address>[:<port>]
```

The default value is 0 . 0 . 0 . 0. The valid port range is:

- Minimum—0
- Maximum—65535

6. **in-destination**—Enter the incoming destination IP address and port of packets to match for static-flow translation. An IP address of 0.0.0.0 matches any source address. Port 0 matches packets received on any port. The port value has no impact on system operation if either ICMP or ALL is the selected protocol. The in-source parameter takes the format:

```
in-destination <ip-address>[:<port>]
```

The default value is 0 . 0 . 0 . 0. The valid port range is:

- Minimum—0
- Maximum—65535

7. **out-realm-id**—Enter the defined realm where traffic leaving this NAT ALG exits the Oracle Communications Session Border Controller .
8. **out-source**—Enter the egress IPv4 address. This is the IPv4 address of the network interface where traffic subject to the NAT ALG you are defining leaves the Oracle Communications Session Border Controller . Do not enter a port number for this parameter. The default value is 0 . 0 . 0 . 0.
9. **out-destination**—Enter the IPv4 address and port number of the server or other destination to which traffic is directed. The default value is 0 . 0 . 0 . 0. The valid port range is:

- Minimum—0
- Maximum—65535

10. **protocol**—Enter the protocol this NAT ALG acts upon. The default value is **UDP**. The valid values are:

- TCP | UDP | ICMP | ALL

## Static Flows

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11. **alg-type**—Enter the type of NAT ALG to use. The default value is **none**. The valid values are:
  - **none**—No dynamic ALG functionality
  - **NAPT**—Configure as NAPT ALG
  - **TFTP**—Configure as TFTP ALG
12. **start-port**—Enter the beginning port number of the port range that the Oracle Communications Session Border Controller allocates on the egress side for flows that this NAPT ALG redirects. The default value is **0**. The valid range is:
  - Minimum—0, 1025
  - Maximum—65535
13. **end-port**—Enter the ending port number of the port range that the Oracle Communications Session Border Controller allocates on the egress side for flows that this NAPT ALG redirects. The default value is **0**. The valid range is:
  - Minimum—0, 1025
  - Maximum—65535
14. **flow-time-limit**—Enter the total time limit for a flow in seconds. A value of **0** means there is no limit. The valid range is:
  - Minimum—0
  - Maximum—999999999
15. **initial-guard-timer**—Enter the initial guard timer value in seconds. A value of **0** means there is no limit. The valid range is:
  - Minimum—0
  - Maximum—999999999
16. **subsq-guard-timer**—Enter the subsequent guard timer value in seconds. A value of **0** means there is no limit. The valid range is:
  - Minimum—0
  - Maximum—999999999
17. **average-rate-limit**—Enter a maximum sustained rate limit in bytes per second. The default value is **0**; no limit. The valid range is:
  - Minimum—0
  - Maximum—125000000

The following example shows a **static-flow** configuration element configured for a NAPT ALG.

```
in-realm-id           access
in-source             172.16.0.0/16
in-destination        172.16.1.16:23
out-realm-id          backbone
out-source            192.168.24.16
out-destination       192.168.24.95:23
protocol              TCP
alg-type              NAPT
start-port            11000
end-port              11999
flow-time-limit       0
initial-guard-timer   60
subsq-guard-timer     60
average-rate-limit    0
```

### Example Configuration: Bidirectional Static Flows

The configuration lines below present the configuration of two example static flows to be used for ICMP to a specific host through the Oracle Communications Session Border Controller.

The following lines present the example configuration for the access to core side.

```
static-flow
in-realm-id access
description
in-source 0.0.0.0
in-destination 10.1.215.63
out-realm-id core
out-source 10.2.214.63
out-destination 10.2.214.51
protocol ICMP
alg-type none
start-port 0
end-port 0
flow-time-limit 0
initial-guard-timer 60
subsq-guard-timer 60
average-rate-limit 0
```

The following lines present the example configuration for the core to access side.

```
static-flow
in-realm-id core
description
in-source 10.2.214.51
in-destination 10.2.214.63
out-realm-id access
out-source 10.1.215.63
out-destination 0.0.0.0
protocol ICMP
alg-type none
start-port 0
end-port 0
flow-time-limit 0
initial-guard-timer 60
subsq-guard-timer 60
average-rate-limit 0
```





---

## High Availability Nodes

Oracle Communications Session Border Controller s can be deployed in pairs to deliver high availability (HA). Two Oracle Communications Session Border Controller s operating in this way are called an HA node. Over the HA node, media and call state are shared, keeping sessions/calls from being dropped in the event of a failure.

Two Oracle Communications Session Border Controller s work together in an HA node, one in active mode and one in standby mode.

- The active Oracle Communications Session Border Controller checks itself for internal process and IP connectivity issues. If it detects that it is experiencing certain faults, it will hand over its role as the active system to the standby Oracle Communications Session Border Controller in the node.
- The standby Oracle Communications Session Border Controller is the backup system, fully synchronized with active Oracle Communications Session Border Controller s session status. The standby Oracle Communications Session Border Controller monitors the status of the active system so that, if needed, it can assume the active role without the active system having to instruct it to do so. If the standby system takes over the active role, it notifies network management using an SNMP trap.

In addition to providing instructions for how to configure HA nodes and their features, this chapter explains how to configure special parameters to support HA for all protocols.

### Overview

---

To produce seamless switchovers from one Oracle Communications Session Border Controller to the other, the HA node uses shared virtual MAC and virtual IP addresses for the media interfaces in a way that is similar to VRRP (virtual router redundancy protocol). When there is a switchover, the standby Oracle Communications Session Border Controller sends out gratuitous ARP messages using the virtual MAC address, establishing that MAC on another physical port within the Ethernet switch. To the upstream router, the MAC and IP are still alive, meaning that existing sessions continue uninterrupted.

Within the HA node, the Oracle Communications Session Border Controller s advertise their current state and health to one another in checkpointing messages; each system is apprised of the other's status. Using Oracle's HA protocol, the Oracle Communications Session Border Controller s communicate with UDP messages sent out and received on the rear interfaces.

The standby Oracle Communications Session Border Controller shares virtual MAC and IPv4 addresses for the media interfaces (similar to VRRP) with the active Oracle Communications Session Border Controller . Sharing addresses eliminates the possibility that the MAC and IPv4 address set on one Oracle Communications Session Border Controller in an HA node will be a single point of failure. The standby Oracle Communications Session Border

## High Availability Nodes

---

Controller sends ARP requests using a utility IPv4 address and its hard-coded MAC addresses to obtain Layer 2 bindings.

The standby Oracle Communications Session Border Controller assumes the active role when:

- It has not received a checkpoint message from the active Oracle Communications Session Border Controller for a certain period of time.
- It determines that the active Oracle Communications Session Border Controller 's health score has decreased to an unacceptable level.
- The active Oracle Communications Session Border Controller relinquishes the active role.

## Establishing Active and Standby Roles

Oracle Communications Session Border Controller s establish active and standby roles in the following ways.

- If a Oracle Communications Session Border Controller boots up and is alone in the network, it is automatically the active system. If you then pair a second Oracle Communications Session Border Controller with the first to form an HA node, then the second system to boot up will establish itself as the standby automatically.
- If both Oracle Communications Session Border Controller s in the HA node boot up at the same time, they negotiate with each other for the active role. If both systems have perfect health, then the Oracle Communications Session Border Controller with the lowest HA rear interface IPv4 address will become the active Oracle Communications Session Border Controller . The Oracle Communications Session Border Controller with the higher HA rear interface IPv4 address will become the standby Oracle Communications Session Border Controller .
- If the rear physical link between the two Oracle Communications Session Border Controller s fails during boot up or operation, both will attempt to become the active Oracle Communications Session Border Controller . In this case, processing will not work properly.

### Health Score

HA Nodes use health scores to determine their active and standby status. Health scores are based on a 100-point system. When a Oracle Communications Session Border Controller is functioning properly, its health score is 100.

Generally, the Oracle Communications Session Border Controller with the higher health score is active, and the Oracle Communications Session Border Controller with the lower health score is standby. However, the fact that you can configure health score thresholds builds some flexibility into using health scores to determine active and standby roles. This could mean, for example, that the active Oracle Communications Session Border Controller might have a health score lower than that of the standby Oracle Communications Session Border Controller , but a switchover will not take place because the active Oracle Communications Session Border Controller 's health score is still above the threshold you configured.

Alarms are key in determining health score. Some alarms have specific health score value that are subtracted from the Oracle Communications Session Border Controller 's health score when they occur. When alarms are cleared, the value is added back to the Oracle Communications Session Border Controller 's health score.

You can look at a Oracle Communications Session Border Controller 's health score using the ACLI **show health** command.

## Switchovers

A switchover occurs when the active Oracle Communications Session Border Controller stops being the active system, and the standby system takes over that function. There are two kinds switchovers: automatic and manual.

### Automatic Switchovers

Automatic switchovers are triggered without immediate intervention on your part. Oracle Communications Session Border Controller s switch over automatically in the following circumstances:

- When the active Oracle Communications Session Border Controller 's health score of drops below the threshold you configure.

- When a time-out occurs, meaning that the active Oracle Communications Session Border Controller has not sent checkpointing messages to the standby Oracle Communications Session Border Controller within the allotted time.

The active Oracle Communications Session Border Controller might not send checkpointing messages for various reasons such as link failure, communication loss, or advertisement loss. Even if the active Oracle Communications Session Border Controller has a perfect health score, it will give up the active role if it does not send a checkpoint message or otherwise advertise its status within the time-out window. Then the standby Oracle Communications Session Border Controller takes over as the active system.

When an automatic switchover happens, the Oracle Communications Session Border Controller that has just become active sends an ARP message to the switch. This message informs the switch to send future messages to its MAC address. The Oracle Communications Session Border Controller that has just become standby ignores any messages sent to it.

### Manual Switchovers


You can trigger a manual switchover in the HA node by using the CLI `notify berpd force` command. This command forces the two Oracle Communications Session Border Controller s in the HA node to trade roles. The active system becomes standby, and the standby becomes active.

In order to perform a successful manual switchover, the following conditions must be met.

- The Oracle Communications Session Border Controller from which you trigger the switchover must be in one of the following states: active, standby, or becoming standby.
- A manual switchover to the active state is only allowed on a Oracle Communications Session Border Controller in the standby or becoming standby state if it has achieved full media, signaling, and configuration synchronization.
- A manual switchover to the active state is only allowed on a Oracle Communications Session Border Controller in the standby or becoming standby state if it has a health score above the value you configure for the threshold.

### State Transitions

Oracle Communications Session Border Controller s can experience series of states as they become active or become standby.

 **Note:** Packet processing only occurs on an active Oracle Communications Session Border Controller .

State	Description
Initial	When the Oracle Communications Session Border Controller is booting.
Becoming Active	When the Oracle Communications Session Border Controller has negotiated to become the active system, but is waiting the time that you set to become fully active. Packets cannot be processed in this state.
Active	When the Oracle Communications Session Border Controller is handling all media, signaling, and configuration processing.
Relinquishing Active	When the Oracle Communications Session Border Controller is giving up its Active status, but before it has become standby. This state is very brief.
Becoming Standby	When the Oracle Communications Session Border Controller is becoming the standby system but is waiting to become fully synchronized. It remains in this state for the period of time you set in the becoming-standby-time parameter, or until it is fully synchronized.
Standby	When the Oracle Communications Session Border Controller is fully synchronized with its active system in the HA node.
OutOfService	When the Oracle Communications Session Border Controller cannot become synchronized in the period of time you set in the becoming-standby-time parameter.

### State Transition Sequences

When the active Oracle Communications Session Border Controller assumes its role as the active system, but then changes roles with the standby Oracle Communications Session Border Controller to become standby, it goes through the following sequence of state transitions:

- Active
- RelinquishingActive
- BecomingStandby
- Standby

When the standby Oracle Communications Session Border Controller assumes its role as the standby system, but then changes roles with the active Oracle Communications Session Border Controller to become active, it goes through the following sequence of state transitions:

- Standby
- BecomingActive
- Active

### HA Features

HA nodes support configuration checkpointing, which you are required to set up so that the configurations across the HA node are synchronized. In addition, you can set up the following optional HA node features:

- Multiple rear interface support
- Gateway link failure detection and polling

### Multiple Rear Interfaces

Configuring your HA node to support multiple rear interfaces eliminates the possibility that either of the rear interfaces you configure for HA support will become a single point of failure. Using this feature, you can configure individual Oracle Communications Session Border Controller s with multiple destinations on the two rear interfaces, creating an added layer of failover support.

When you configure your HA node for multiple rear interface support, you can use last two rear interfaces (wancom1 and wancom2) for HA—the first (wancom0) being used for Oracle Communications Session Border Controller management. You can connect your Oracle Communications Session Border Controller s using any combination of wancom1 and wancom2 on both systems. Over these rear interfaces, the Oracle Communications Session Border Controller s in the HA node share the following information:

- Health
- Media flow
- Signaling
- Configuration

For example, if one of the rear interface cables is disconnected or if the interface connection fails for some other reason, all health, media flow, signaling, and configuration information can be checkpointed over the other interface.

Health information is checkpointed across all configured interfaces. However, media flow, signaling, and configuration information is checkpointed across one interface at a time, as determined by the Oracle Communications Session Border Controller 's system HA processes.

### Configuration Checkpointing

During configuration checkpointing, all configuration activity and changes on one Oracle Communications Session Border Controller are automatically mirrored on the other. Checkpointed transactions include adding, deleting, or modifying a configuration on the active Oracle Communications Session Border Controller . This means that you only need to perform configuration tasks on the active Oracle Communications Session Border Controller because the standby system will go through the checkpointing process and synchronize its configuration to reflect activity and changes.

Because of the way configuration checkpointing works, the ACLI **save-config** and **activate-config** commands can only be used on the active Oracle Communications Session Border Controller .

- When you use the ACLI **save-config** command on the active Oracle Communications Session Border Controller , the standby Oracle Communications Session Border Controller learns of the action and updates its own configuration. Then the standby Oracle Communications Session Border Controller saves the configuration automatically.
- When you use the ACLI **activate-config** command on the active Oracle Communications Session Border Controller , the standby Oracle Communications Session Border Controller learns of the action and activates its own, updated configuration.

The ACLI **acquire-config** command is used to copy configuration information from one Oracle Communications Session Border Controller to another.

### Gateway Link Failure Detection and Polling


In an HA node, the Oracle Communications Session Border Controller s can poll for and detect media interface links to the gateways as they monitor ARP connectivity. The front gateway is assigned in the network interface configuration, and is where packets are forwarded out of the originator's LAN.

The Oracle Communications Session Border Controller monitors connectivity using ARP messages that it exchanges with the gateway. The Oracle Communications Session Border Controller sends regular ARP messages to the gateway in order to show that it is still in service; this is referred to as a heartbeat message. If the Oracle Communications Session Border Controller deems the gateway unreachable for any of the reasons discussed in this section, a network-level alarm is generated and an amount you configure for this fault is subtracted from the system's health score.


The Oracle Communications Session Border Controller generates a gateway unreachable network-level alarm if the Oracle Communications Session Border Controller has not received a message from the media interface gateway within the time you configure for a heartbeat timeout. In this case, The Oracle Communications Session Border Controller will send out ARP requests and wait for a reply. If no reply is received after resending the set number of ARP requests, the alarm remains until you clear it. The health score also stays at its reduced amount until you clear the alarm.

When valid ARP requests are once again received, the alarm is cleared and system health scores are increased the appropriate amount.

You can configure media interface detection and polling either on a global basis in the SD HA nodes/redundancy configuration or on individual basis for each network interface in the network interface configuration.

-  **Note:** To improve the detection of link failures, the switchport connected to the NIU should have Spanning Tree disabled. Enabling Spanning Tree stops the switchport from forwarding frames for several seconds after a reset. This prevents the NIU from reaching the gateway and generates a "gateway unreachable" network-level alarm.

## Before Configuring a High Availability (HA) Pair

-  **Note:** When you configure an HA pair, you must use the same password for both Oracle Communications Session Border Controllers.

Before configuring the parameters that support HA, complete the following steps.

1. Establish the physical connections between the Oracle Communications Session Border Controllers. Avoid breaking the physical link (over the rear interfaces) between the Oracle Communications Session Border Controllers in an HA node. If the physical link between the Oracle Communications Session Border Controllers breaks, they will both attempt to become the active system and HA will not function as designed.
2. Confirm that both Oracle Communications Session Border Controllers are set to the same time. Use the ACLI **show clock** command to view the system time. If the Oracle Communications Session Border Controllers show different times, use the **system-timeset** command to change the time.

## High Availability Nodes

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Oracle recommends that you use NTP to synchronize your Oracle Communications Session Border Controllers, so that they have a common stratum time source.

3. HA nodes use ports 1 and 2 as the HA interfaces. Set port 0 on the rear panel of the Oracle Communications Session Border Controller chassis as the boot and management interface. You configure the rear interfaces during the physical interface configuration.
4. For ACLI configuration, you need to know the target names of the Oracle Communications Session Border Controllers making up the HA node. The target name of the system is reflected in the ACLI's system prompt. For example, in the ACMEPACKET# system prompt, ACMEPACKET is the target name.

You can also see and set the target name in the Oracle Communications Session Border Controller boot parameters.



**Note:** The target name is case sensitive.

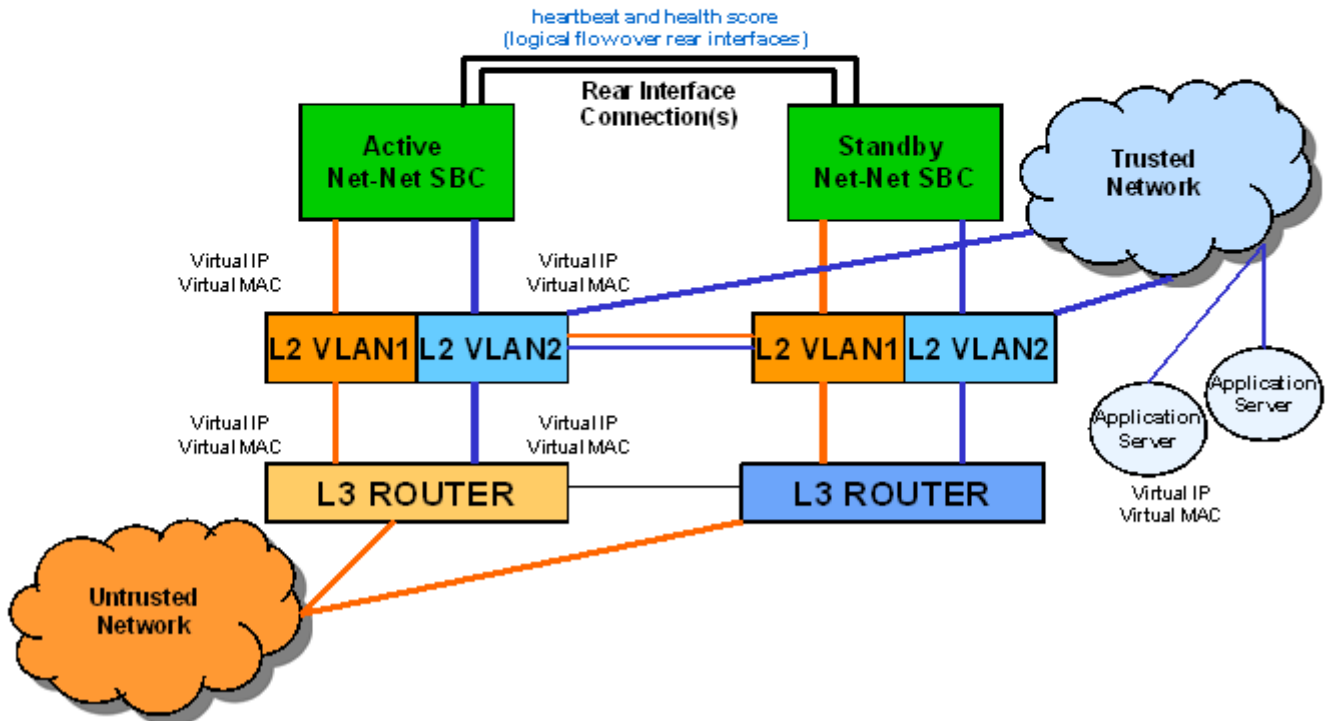
5. Devise virtual MAC addresses so that, if a switchover happens, existing sessions are not interrupted.

## HA Node Connections

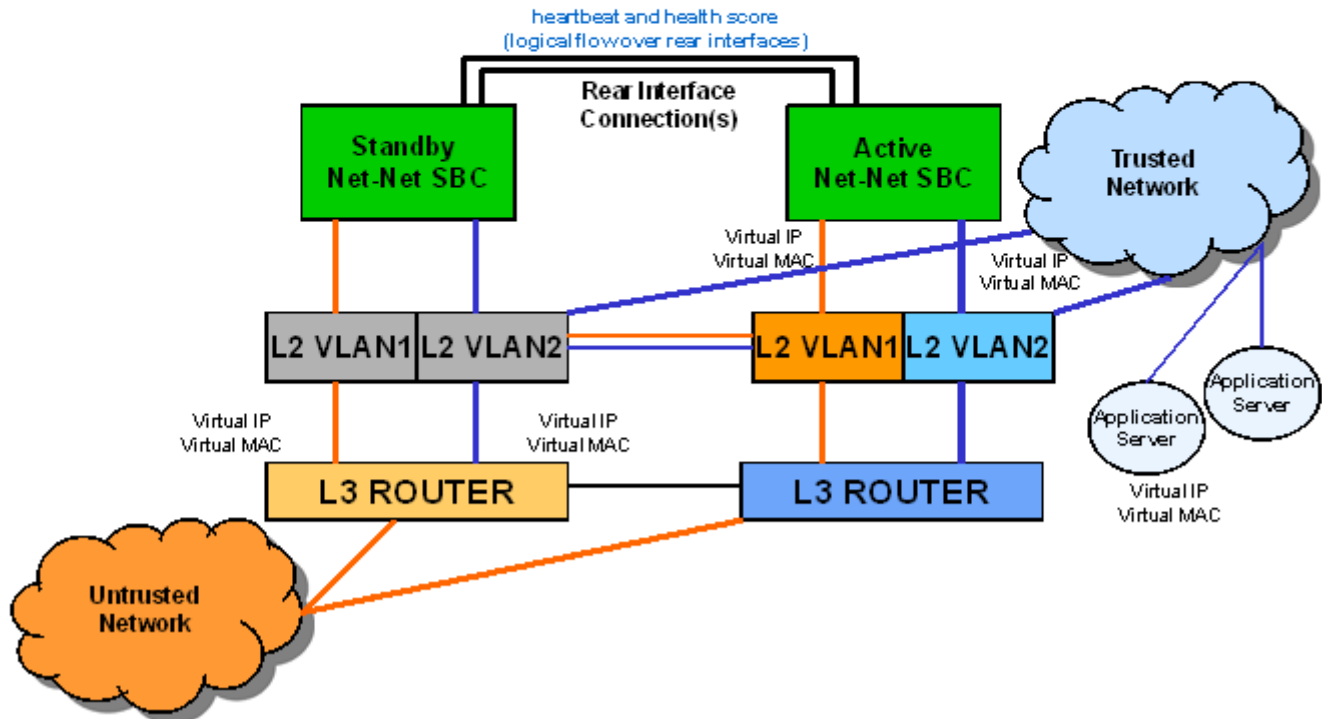
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To use HA, you must establish Layer 2 and Layer 3 networks that interconnect two Oracle Communications Session Border Controllers and support HA with the required physical network connections. The basic network set-up in the following diagram shows an HA node deployment where each system is connected to its own Layer 2 switch. This set-up provides a measure of added redundancy in the event that one of the switches fails.

Here, the active system is using the virtual MAC and IP addresses.



In the second diagram, the same network is shown with the HA node having experienced a switchover. The previously standby Oracle Communications Session Border Controller has taken over the active role in the HA node and is using the virtual IP and MAC addresses.



 **Note:** Switches should never be in master-slave mode. If they are, HA will not work correctly.

The following are hardware set-up and location considerations for placing an HA Node:

- You must set up each Oracle Communications Session Border Controller according to the requirements and safety precautions set out in the *Oracle Communications System Hardware Installation Guide*.
- Each Oracle Communications Session Border Controller's media interfaces must be connected to the same switches (or other network entities), as shown in the diagram above.
- The length of the shielded crossover 10/100 category 5 Ethernet cable that connects the Oracle Communications Session Border Controllers from the rear interfaces must be able to reach from the configured rear interface on one Oracle Communications Session Border Controller to the configured rear interface on the other.

HA nodes use Oracle element redundancy protocol for its tasks. This protocol uses a connection between the rear interfaces of two Oracle Communications Session Border Controllers to checkpoint the following information: health, state, media flow, signaling, and configuration.

We recommend that you use shielded category 5 (RJ45) crossover cables for all 10/100 Ethernet connections used for HA.

You can set up either single or multiple rear interface support for your HA node. For single interface support, one cable connects the two Oracle Communications Session Border Controllers; for multiple interface support, two cables are used. However, the software configurations for each type of connection mode are different.



When you make these connections, do not use port 0 (wancom0) on the rear interface of the Oracle Communications Session Border Controller chassis; that port should only be used for Oracle Communications Session Border Controller management. Instead, use ports 1 and 2 (wancom1 and wancom2).

To cable Oracle Communications Session Border Controllers using single rear interface support:

1. Using a 10/100 category 5 crossover cable, insert one end into either port 1 (wancom1) or port 2 (wancom2) on the rear interface of the first Oracle Communications Session Border Controller.
2. Insert the other end of the cable into port 1 or port 2 on the rear interface of the second Oracle Communications Session Border Controller. We recommend that you use corresponding ports on the two systems. That is, use port 1 on both systems or use port 2 on both systems.
3. Perform software configuration for these interfaces as described in this chapter.

To cable Oracle Communications Session Border Controllers using multiple rear interface support:

4. Using a 10/100 category 5 crossover cable, insert one end into port 1 on the rear interface of the first Oracle Communications Session Border Controller.
5. Insert the other end of that cable into port 1 on the rear interface of the second Oracle Communications Session Border Controller to complete the first physical connection.
6. Using a second 10/100 category 5 cable, insert one end into port 2 on the rear interface of the first Oracle Communications Session Border Controller.
7. Insert the other end of this second cable in port 2 on the rear interface of the second Oracle Communications Session Border Controller to complete the second physical connection.
8. Perform software configuration for these interfaces as described in this chapter.

## Virtual MAC Addresses

In order to create the HA node, you need to create virtual MAC addresses for the media interfaces. You enter these addresses in virtual MAC address parameters for physical interface configurations where the operation type for the interface is media.

The HA node uses shared virtual MAC (media access control) and virtual IP addresses for the media interfaces. When there is a switchover, the standby Oracle Communications Session Border Controller sends out an ARP message using the virtual MAC address, establishing that MAC on another physical port within the Ethernet switch. Virtual MAC addresses are actually unused MAC addresses that based on the Oracle Communications Session Border Controller's root MAC address.

The MAC address is a hardware address that uniquely identifies each Oracle Communications Session Border Controller. Given that, the virtual MAC address you configure allows the HA node to appear as a single system from the perspective of other network devices. To the upstream router, the MAC and IP are still alive, meaning that existing sessions continue uninterrupted through the standby Oracle Communications Session Border Controller.

Depending on the type of physical layer cards you have installed, you can create MAC addresses as follows: Four Ethernet (MAC) address for each configured four-port GigE physical interface card.

## Virtual MAC Address Configuration

To create a virtual MAC address:

1. Determine the Ethernet address of the Oracle Communications Session Border Controller by using the CLI **show interfaces** command. This command only works if you have already set up physical interface configurations. Otherwise, you will get no output.

The example below shows you where the Ethernet address information appears; this sample has been shortened for the sake of brevity. For each type of physical interface card, the Oracle Communications Session Border Controller displays the following:

```
ACMEPACKET# show interfaces
f00 (media slot 0, port 0)
  Flags: UP BROADCAST MULTICAST ARP RUNNING
  Type: GIGABIT_ETHERNET
```

## High Availability Nodes

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```
Admin State: enabled
Auto Negotiation: enabled
Internet address: 10.10.0.10      Vlan: 0
Broadcast Address: 10.10.255.255
Netmask: 0xffff0000
Gateway: 10.10.0.1
Ethernet address is 00:08:25:01:07:64
```

### 2. Identify the root portion of the Ethernet (MAC) address.

Each Oracle Communications Session Border Controller has MAC addresses assigned to it according to the following format: 00:08:25:XX:YY:ZN where:

- 00:08:25 refers to Acme Packet
- XX:YY:ZN refers to the specific Oracle Communications Session Border Controller
- N is a 0-f hexadecimal value available for the Oracle Communications Session Border Controller

In this example, the root part of this address is 00:08:25:XX:YY:Z.

### 3. To create an unused MAC address (that you will use as the virtual MAC address) take the root MAC address you have just identified. Replace this N value with unused hexadecimal values for the Oracle Communications Session Border Controller: 8, 9, e, or f.

In other words, you change the last digit of the MAC address to either 8, 9, e, or f depending on which of those address are not being used.

For example, for an HA node with MAC address bases of 00:08:25:00:00:00 and 00:08:25:00:00:10, the following addresses would be available for use at virtual MAC addresses:

- 00:08:25:00:00:08
- 00:08:25:00:00:09
- 00:08:25:00:00:0e
- 00:08:25:00:00:0f
- 00:08:25:00:00:18
- 00:08:25:00:00:19
- 00:08:25:00:00:1e
- 00:08:25:00:00:1f

Corresponding media interfaces in HA nodes must have the same virtual MAC addresses. Given that you have various physical interface card options, the following points illustrate how virtual MAC address can be shared:

If you are using a four-port GigE physical interface card, both the active Oracle Communications Session Border Controller and the standby Oracle Communications Session Border Controller might have the following virtual MAC address scheme for the slots:

- Slot 0 - 00:08:25:00:00:0e and 00:08:25:00:00:0f
- Slot 1 - 00:08:25:00:00:1e and 00:08:25:00:00:1f



**Note:** Note the virtual MAC addresses you have created so that you can reference them easily when you are configuring the physical interfaces for HA.

## HA Node Connections

---

You can begin software configuration for your HA node after you have:

- Completed the steps for physical set-up and connection.
- Noted the target name of the Oracle Communications Session Border Controllers that make up the HA node.
- Configured the virtual MAC addresses that you need, according to the type of physical interface cards installed on your Oracle Communications Session Border Controller.

## HA Node Connection Configuration

If you are using HA, you need to set the physical interface configuration parameters described in this section to establish successful connections. These parameters are for rear and media interfaces.

To access physical interface menu in the ACLI:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure)# system
```

3. Type **phy-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (system)# phy-interface
ACMEPACKET (phy-interface)#
```

From this point, you can configure physical interface parameters. To view all physical interfaces parameters, enter a **?** at the system prompt.

### Rear Interfaces

You can use port 1 (wancom1) or port 2 (wancom2) as interfaces to support HA. Do not use port 0 (wancom 0) as that port is reserved for carrying management traffic.

Make sure that the physical connections you have made on the rear panel of your Oracle Communications Session Border Controllers correspond to the configurations you enter for physical interfaces. You can connect Oracle Communications Session Border Controllers through multiple rear interfaces. For multiple rear interface connectivity, cable both port 1 and port 2 (wancom1 and wancom2) on one Oracle Communications Session Border Controller to port1 and port 2 on the other Oracle Communications Session Border Controller in the HA node.

The Oracle Communications Session Border Controller's HA function depends heavily on health scores to determine the active and standby roles in an HA node. You can set the amount that will be subtracted from a Oracle Communications Session Border Controller's health score in the event that a management interface fails for any reason. For example, a connection might become invalid or a cable might be removed inadvertently.

The following example shows how a configured physical interface will appear in the ACLI for an HA node:

```
phy-interface
  name wancom1
  operation-type Control
  port 1
  slot 0
  virtual-mac
  wancom-health-score 20
```

To establish rear interfaces for use in an HA node using the ACLI:

1. Access the physical interface menu.
2. **name**—Set a name for the interface using any combination of characters entered without spaces. For example: wancom1.
3. **operation-type**—Set this parameter to **Control**.
4. **slot**—Set this parameter to **0**.
5. **port**—Set this parameter to **1** or **2**.
6. **wancom-health-score**—Enter the number value between 0 and 100. This value will be subtracted from the Oracle Communications Session Border Controller's health score in the event that a rear interface link fails. We recommend that you change this value from its default (**50**), and set it to **20**.

This value you set here is compared to the active and emergency health score thresholds you establish in the Oracle Communications Session Border Controller HA node (redundancy) configuration.

## High Availability Nodes

7. For multiple rear interface support, configure the remaining, unused rear interfaces with the appropriate values.

The following example shows configuration for multiple rear interface support.

```
ACMEPACKET(system)# phy-interface
ACMEPACKET(phy-interface)# name wancom1
ACMEPACKET(phy-interface)# operation-type control
ACMEPACKET(phy-interface)# port 1
ACMEPACKET(phy-interface)# wancom-health-score 20
ACMEPACKET(phy-interface)# done
ACMEPACKET(phy-interface)# name wancom2
ACMEPACKET(phy-interface)# operation-type control
ACMEPACKET(phy-interface)# port 2
ACMEPACKET(phy-interface)# wancom-health-score 20
ACMEPACKET(phy-interface)# done
```

### Media Interface Virtual MAC Addresses

To configure HA for the media interfaces in an HA node, you must set one or more virtual MAC addresses, according to the type of physical layer cards you have installed on your Oracle Communications Session Border Controller.

To set a virtual MAC address using the ACLI:

1. Access the physical interface configuration.
2. Configure all relevant parameters as noted in the Physical Interfaces section of this guide's *System Configuration* chapter.

Since virtual MAC addresses are used for media interfaces only, verify that the operation type is set to media.

3. **virtual-mac**—Enter the virtual MAC address that you have created using the steps in the Virtual MAC Addresses section.

## HA Node Parameters

To establish a pair of Oracle Communications Session Border Controllers as an HA node, you need to configure basic parameters that govern how the Oracle Communications Session Border Controllers:

- Transition on switchover
- Share media and call state information
- Checkpoint configuration data

The following example shows what an HA configuration might look like in the ACLI.

```
redundancy-config
state                enabled
log-level            WARNING
health-threshold     75
emergency-threshold  50
port                 9090
advertisement-time   500
percent-drift         210
initial-time         1250
becoming-standby-time 45000
becoming-active-time 100
```

You need to configure the two Oracle Communications Session Border Controllers to be HA node peers. To enable configuration checkpointing, you must to configure two peers in the ACLI, one for the primary and one for the secondary Oracle Communications Session Border Controller. The HA node peers configuration also allows you to configure destinations for where to send health and state information. Unless you create Oracle Communications Session Border Controller peers and destinations configurations, HA will not work properly.

The following example shows what an HA configuration might look like in the ACLI.

```
peer
name                netnetsd1
state                enabled
```

```

type Primary
destination
address 169.254.1.1:9090
network-interface wancom1:0
peer
name netnetsd2
state enabled
type Secondary
destination
address 169.254.1.2:9090
network-interface wancom1:0

```

## HA Node Parameter Configuration

To configure general HA node parameters using the ACLI:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # system
```

3. Type **redundancy** and press Enter.

```
ACMEPACKET (system) # redundancy
```

From here, you configure basic HA node parameters. To view all basic HA node parameters, enter a **?** at the system prompt.

4. **state**—Leave this parameter set to **enabled** for HA to work. To stop HA operation, set this parameter to **disabled**. The default value is **enabled**. The valid values are:

- enabled | disabled

5. **log-level**—Set the log level you want to use for the HA system process. The value you set in this field overrides any log level value you set for the entire Oracle Communications Session Border Controller in the system configuration process log level parameter. The default value is **INFO** which allows you to receive a moderate amount of detail. The valid values are:

- emergency | critical | major | minor | warning | notice | info | trace | debug | detail

6. **health-threshold**—Enter a value between 0 and 100 to set the health score at which the Oracle Communications Session Border Controllers in the HA node gracefully exchange active-standby roles. The default value is **75**. The valid range is:

- Minimum—1
- Maximum—100

For example, if this field is set to **75** and the active Oracle Communications Session Border Controller's health score falls below that point, the standby Oracle Communications Session Border Controller will take over the active role. However, Oracle Communications Session Border Controller will only take over the active role if its own health score is 75 or better.

7. **emergency-threshold**—Enter the health score for the standby Oracle Communications Session Border Controller to become active immediately. The default value is **50**. The valid range is:

- Minimum—0
- Maximum—100

If the standby Oracle Communications Session Border Controller is initializing and the active Oracle Communications Session Border Controller's health score is below the health threshold, the standby Oracle Communications Session Border Controller will take the active role and there will be a graceful switchover. If the active Oracle Communications Session Border Controller's health score is below the emergency threshold, then the switchover will be immediate.

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If the standby Oracle Communications Session Border Controller has a health score below the emergency threshold and the active Oracle Communications Session Border Controller is unhealthy, the active Oracle Communications Session Border Controller will not give up its active role.

- advertisement-time**—Enter the number of milliseconds to set how often Oracle Communications Session Border Controllers in an HA node inform each other of their health scores.

We recommend you leave this parameter set to its default, **500**. The valid range is:

- Minimum—50
- Maximum—999999999

- percent-drift**—Enter the percentage of the advertisement time that you want one member of the HA node to wait before considering the other member to be out of service. For the standby Oracle Communications Session Border Controller, this is the time it will wait before taking the active role in the HA node. The default value is **210**. The valid range is:

- Minimum—100
- Maximum—65535

- initial-time**—Enter the number of milliseconds to set the longest amount of time the Oracle Communications Session Border Controller will wait at boot time to change its state from initial to either becoming active or becoming standby. The default value is **1250**. The valid range is:

- Minimum—5
- Maximum—999999999

- becoming-standby-time**—Enter the number of milliseconds the Oracle Communications Session Border Controller waits before becoming standby, allowing time for synchronization. If it is not fully synchronized within this time, it will be declared out of service.

We recommend that you do not set this parameter below **45000**. If a large configuration is being processed, we recommend setting this parameter to **180000** to allow enough time for configuration checkpointing. The default value is **45000**. The valid range is:

- Minimum—5
- Maximum—999999999

- becoming-active-time**—Enter the number of milliseconds that the standby Oracle Communications Session Border Controller takes to become active in the event that the active Oracle Communications Session Border Controller fails or has an intolerably decreased health score. The default value is **100**. The valid range is:

- Minimum—5
- Maximum—999999999

### HA Node Peer Configuration

To configure a Oracle Communications Session Border Controller as an HA node peer:

- From the redundancy menu, type **peers** and press Enter.

```
ACMEPACKET(system)# redundancy
ACMEPACKET(redundancy)# peers
```

- state**—Enable or disable HA for this Oracle Communications Session Border Controller. The default value is **enabled**. The valid values are:

- enabled | disabled

- name**—Set the name of the HA node peer as it appears in the target name boot parameter.

This is also the name of your system that appears in the system prompt. For example, in the system prompt ACMEPACKET1#, ORACLE1 is the target name for that Oracle Communications Session Border Controller.

- type**—These values refer to the primary and secondary utility addresses in the network interface configuration. To determine what utility address to use for configuration checkpointing, set the type of Oracle Communications Session Border Controller: primary or secondary.



**Note:** You must change this field from unknown, its default. The valid values are:

- **primary**—Set this type if you want the Oracle Communications Session Border Controller to use the primary utility address.
- **secondary**—Set this type if you want the Oracle Communications Session Border Controller to use the secondary utility address.
- **unknown**—If you leave this parameter set to this default value, configuration checkpointing will not work.

## HA Node Health And State Configuration

To configure where to send health and state information within an HA node:

1. From the peers configuration, type destinations and press Enter.

```
ACMEPACKET (rdncy-peer) # destinations
ACMEPACKET (rdncy-peer-dest) #
```

2. **address**—Set the destination IPv4 address and port of the other Oracle Communications Session Border Controller in the HA node to which this Oracle Communications Session Border Controller will send HA-related messages. This value is an IPv4 address and port combination that you enter as: IPAddress:Port. For example, 169.254.1.1:9090.
  - The IPv4 address portion of this value is the same as the IPv4 address parameter set in a network interface configuration of the other Oracle Communications Session Border Controller in the HA node.
  - The port portion of this value is the port you set in the Oracle Communications Session Border Controller HA Node/redundancy configuration for the other Oracle Communications Session Border Controller in the node.
3. **network-interface**—Set the name and subport for the network interface where the Oracle Communications Session Border Controller receives HA-related messages. Valid names are wancom1 and wancom2. This name and subport combination must be entered as name:subport; for example, **wancom1 : 0**.

The network interface specified in this parameter must be linked to a physical interface configured with rear interface parameters. The physical interface's operation type must be control or maintenance, and so the subport ID portion of this parameter is 0. The subport ID is the VLAN tag.

## Synchronizing Configurations

You can synchronize the Oracle Communications Session Border Controllers (SBC) in your High Availability (HA) node in the following ways:

- Automatically — Set up configuration checkpointing within the HA node.
- Manually — Check whether or not configurations in the HA node are synchronized, and then copy configuration data from one SBC to the other.

When you initially configure a new HA node, copy the configuration data manually from one SBC to the other. When you complete the process, you can configure your HA node to automatically synchronize configurations.

Oracle recommends that you configure the HA node for configuration checkpointing because that is the most reliable way to ensure that both systems have the same configuration.

## Synchronize HA Peers

The process for synchronizing the peers in a High Availability (HA) node for the first time by way of the ACLI includes the following steps.

1. Create a complete configuration on the active Oracle Communications Session Border Controller (SBC). Include all HA node parameters and all rear interface configurations. Confirm that the rear interfaces are configured to send and receive information across the HA node.
2. On the active SBC, save the configuration.
3. On the active SBC, reboot to run the new configuration.

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Use the ACLI **show health** command to see that the active SBC booted without a peer. This changes after you copy the configuration to the standby SBC and activate the configuration.

4. On the standby SBC, perform the ACLI **acquire-config** command to copy the configuration from the active SBC. Use the **acquire-config** command with the IPv4 address of wancom 0 on the active SBC.

```
ACMEPACKET2# acquire-config 192.168.12.4
```

The IPv4 address of wancom 0 on the active SBC is the IPv4 address portion of the value displayed for the inet on ethernet boot parameter. The following codeblock shows an example of the inet on ethernet value that the system displays when you view the boot parameters:

```
inet on ethernet (e) : 192.168.12.4:ffff0000
```

5. When the copying process (**acquire-config**) is complete, reboot the standby SBC to activate the configuration. The system boots and displays start-up information.
6. Confirm that the HA node synchronized the configurations by using the ACLI **display-current-cfg-version** and **display-running-cfg-version** commands:

```
ACMEPACKET# display-current-cfg-version  
Current configuration version is 3  
ACMEPACKET# display-running-cfg-version  
Running configuration version is 3  
ACMEPACKET# display-current-cfg-version  
Current configuration version is 3  
ACMEPACKET# display-running-cfg-version  
Running configuration version is 3
```

In the preceding example, all configuration versions—current and running—are the same number (3).

## Using Configuration Checkpointing

The Oracle Communications Session Border Controller's primary and secondary utility addresses support configuration checkpointing, allowing the standby Oracle Communications Session Border Controller to learn configuration changes from the active Oracle Communications Session Border Controller. This means that you only have to enter configuration changes on the active Oracle Communications Session Border Controller for the configurations across the HA node to be updated.

Configuration checkpointing uses parameters in the network interface and in the Oracle Communications Session Border Controller HA Nodes/redundancy configurations.

If you are using configuration checkpointing, you also need to set up two Oracle Communications Session Border Controller peer configurations: one the primary, and one for the secondary.

### HA Configuration Checkpointing

You need to first set applicable network interface configuration parameters, and then establish applicable parameters in the Oracle Communications Session Border Controller HA node (redundancy) configuration.

We recommend that you do not change the configuration checkpointing parameters in the redundancy configuration. Using the defaults, this feature will function as designed.

 **Note:** Remember to set the appropriate type parameter in the HA node redundancy peers configuration.

For the network interface, these parameters appear as they do in the following example when you use the ACLI. This example has been shortened for the sake of brevity.

```
pri-utility-addr 169.254.1.1  
sec-utility-addr 169.254.1.2
```

For the Oracle Communications Session Border Controller HA node (redundancy) configuration, these parameters appear as they do in the following example when you use the ACLI. This example has been shortened for the sake of brevity. You should not change these values without consultation from Oracle Technical Support or your Oracle Systems Engineer.



```

cfg-port                1987
cfg-max-trans           10000
cfg-sync-start-time    5000
cfg-sync-comp-time     1000

```

To configure HA configuration checkpointing in the ACLI:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # system
```

3. Type **network-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (system) # network-interface
```

```
ACMEPACKET (network-interface) #
```

From here, you can configure network interface parameters. To view all network interfaces parameters, enter a **?** at the system prompt.

4. **pri-utility-addr**—Enter the utility IP address for the primary HA peer in an HA architecture.

This address can be any unused IP address within the subnet defined for the network interface. For example, given a network interface of with the IPv4 address 168.0.4.15/24 (identifying the host associated with the network interface), the possible range of unused IPv4 addresses is 168.0.4.1 to 168.0.4.254. Your network administrator will know which IPv4 addresses are available for use.

5. **sec-utility-addr**—Enter the utility IP address for the secondary Oracle Communications Session Border Controller peer in an HA architecture.

Usually, this IP address is usually the next in the sequence up from the primary utility address. It is also generated from the range of unused IP addresses within the subnet defined for the network interface.

6. Save your work and exit the network interface configuration.

```
ACMEPACKET (network-interface) # done
```

```
ACMEPACKET (network-interface) # exit
```

```
ACMEPACKET (system) #
```

7. Access the system HA node/redundancy configuration by typing **redundancy** at the system prompt and then press Enter.

```
ACMEPACKET (system) # redundancy
```

```
ACMEPACKET (redundancy) #
```



**Note:** We strongly recommend that you keep the default settings for the parameters Steps 8 through 11.

8. **cfg-port**—Enter the port number for sending and receiving configuration checkpointing messages. Setting this to zero (0) disables configuration checkpointing. The default value is **1987**. The valid values are:

- Minimum—0, 1025
- Maximum—65535

9. **cfg-max-trans**—Enter the number of HA configuration checkpointing transactions that you want to store. The active Oracle Communications Session Border Controller maintains the transaction list, which is acquired by the standby Oracle Communications Session Border Controller. Then the standby system uses the list to synchronize its configuration with active system. The default value is **10000**. The valid range is:

- Minimum—0
- Maximum—4294967295

Transactions include: modifications, additions, and deletions. If the maximum number of stored transactions is reached, the oldest transactions will be deleted as new transactions are added.

10. **cfg-sync-start-time**—Enter the number of milliseconds before the Oracle Communications Session Border Controller tries to synchronize by using configuration checkpointing. On the active Oracle

## High Availability Nodes

---

Communications Session Border Controller, this timer is continually reset as the Oracle Communications Session Border Controller checks to see that it is still in the active role. If it becomes standby, it waits this amount of time before it tries to synchronize.

We recommend you leave this field at its default value, **5000**, so that configuration checkpointing can function correctly. The valid range is:

- Minimum—0
- Maximum—4294967295

11. **cfg-sync-comp-time**—Enter the number of milliseconds that the standby Oracle Communications Session Border Controller waits before checkpointing to obtain configuration transaction information after the initial checkpointing process is complete.

We recommend you leave this field at its default value, **1000**, so that configuration checkpointing can function correctly. The valid range is:

- Minimum—0
- Maximum—4294967295

12. Save your work and exit the redundancy configuration.

```
ACMEPACKET (redundancy) # done
ACMEPACKET (redundancy) # exit
ACMEPACKET (system) #
```

## Manually Checking Configuration Synchronization

You can check that the current and active configurations are synchronized across the HA node. The current configuration is the one with which you are currently working, and the active configuration is the one active on the system.

To confirm that the systems in the HA node have synchronized configurations:

1. On the active Oracle Communications Session Border Controller in the Superuser menu, enter the following ALCI commands and press Enter. Note the configuration version numbers for comparison with those on the standby Oracle Communications Session Border Controller.

- **display-current-cfg-version**—Shows the version number of the configuration you are currently viewing (for editing, updating, etc.).

```
ACMEPACKET# display-current-cfg-version
Current configuration version is 30
```

- **display-running-cfg-version**—Shows the version number of the active configuration running on the Oracle Communications Session Border Controller.

```
ACMEPACKET# display-running-cfg-version
Running configuration version is 30
```

2. On the standby Oracle Communications Session Border Controller, enter the following ALCI commands and press Enter. Note the configuration version numbers for comparison with those on the active Oracle Communications Session Border Controller.

```
ACMEPACKET# display-current-cfg-version
Current configuration version is 30
ACMEPACKET# display-running-cfg-version
Running configuration version is 30
```

3. Compare the configuration numbers. If the version numbers on the active Oracle Communications Session Border Controller match those on the standby Oracle Communications Session Border Controller, then the systems are synchronized.

If the version numbers do not match, you need to synchronize the Oracle Communications Session Border Controllers. You can do so using the ACLI **acquire-config** command.

## Media Interface Link Detection and Gateway Polling

You can use media interface link detection and gateway polling globally on the Oracle Communications Session Border Controller, or you can override those global parameters on a per-network-interface basis.

- Use the Oracle Communications Session Border Controller HA node (redundancy) configuration to establish global parameters. When configured globally, they will appear like this in the ACLI:

```
gateway-heartbeat-interval 0
gateway-heartbeat-retry 0
gateway-heartbeat-timeout 1
gateway-heartbeat-health 0
```

- Use the network interface's gateway heartbeat configuration to override global parameters on a per-network-interface basis. When configured for the network interface, these parameters will appear like this in the ACLI:

```
gw-heartbeat
state enabled
heartbeat 0
retry-count 0
retry-timeout 1
health-score 0
```

## Media Interface Link Detection and Gateway Polling Configuration

To configure global media interface link detection and gateway polling:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # system
```

3. Type **redundancy** and press Enter.

```
ACMEPACKET (system) # redundancy
```

From here, you can configure gateway heartbeat parameters. To view all gateway heartbeat parameters, enter a **?** at the system prompt.

4. **gateway-heartbeat-interval**—Enter the number of seconds between heartbeats for the media interface gateway. Heartbeats are sent at this interval as long as the media interface is viable. The default value is **0**. The valid range is:
  - Minimum—0
  - Maximum—65535
5. **gateway-heartbeat-retry**—Enter the number of heartbeat retries (subsequent ARP requests) to send to the media interface gateway before it is considered unreachable. The default value is **0**. The valid range is:
  - Minimum—0
  - Maximum—65535
6. **gateway-heartbeat-timeout**—Enter the heartbeat retry time-out value in seconds. The default value is **1**. The valid range is:
  - Minimum—0
  - Maximum—65535

This parameter sets the amount of time between Oracle Communications Session Border Controller ARP requests to establish media interface gateway communication after a media interface gateway failure.

7. **gateway-heartbeat-health**—Enter the amount to subtract from the Oracle Communications Session Border Controller's health score if a media interface gateway heartbeat fails. If the value you set in the gateway

time-out retry field is exceeded, this amount will be subtracted from the system's overall health score. The default value is **0**. The valid range is:

- Minimum—0
- Maximum—100

## Media Interface Link Detection and Gateway Polling Configuration 2

To configure media interface link detection and gateway polling on a per-network-interface basis in the CLI:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # system
```

3. Type **network-interface** and press Enter.

```
ACMEPACKET (system) # network-interface
```

4. Type **gw-heartbeat** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (network-interface) # gw-heartbeat  
ACMEPACKET (gw-heartbeat) #
```

From here, you can configure gateway heartbeat parameters for the network interface. To view all gateway heartbeat parameters, enter a **?** at the system prompt.

5. **state**—Enable or disable the gateway heartbeat feature. The default value is **enabled**. The valid values are:

- enabled | disabled

6. **heartbeat**—Enter the number of seconds between heartbeats for the media interface gateway. Heartbeats are sent at this interval as long as the media interface is viable. The default value is zero (**0**). The valid range is:

- Minimum—0
- Maximum—65535

The value you configure in this field overrides any globally applicable value set in the gateway heartbeat interval parameter in the Oracle Communications Session Border Controller HA node (redundancy) configuration.

7. **retry-count**—Enter the number of heartbeat retries that you want sent to the media interface gateway before it is considered unreachable. The default value is zero (**0**). The valid range is:

- Minimum—0
- Maximum—65535

8. **retry-timeout**—Enter the heartbeat retry time-out value in seconds. The default value is **1**. The valid range is:

- Minimum—1
- Maximum—65535

This parameter sets the amount of time between system ARP requests to establish media interface gateway communication after a media interface gateway failure.

9. **health-score**—Enter the amount to subtract from the system's health score if a media interface gateway heartbeat fails; this parameter defaults to 0. If the value you set in the retry-time-out field is exceeded, this amount will be subtracted from the system's overall health score. The default value is zero (**0**). The valid range is:

- Minimum—0
- Maximum—100


## Signaling Checkpointing

You can configure your HA node to checkpoint signaling for SIP.

### SIP Signaling Checkpointing

In the SIP configuration, you can set parameters that enable SIP signaling checkpointing across an HA node.

When configured, these parameters will appear in the ACLI as they do in example below.

 **Note:** This example shows the default values being used, and we recommend that you do not change these values from their defaults.

```
red-sip-port          1988
red-max-trans        10000
red-sync-start-time  5000
red-sync-comp-time   1000
```

### Signaling Checkpointing Configuration

To configure SIP signaling checkpointing across an HA node in the ACLI:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-router** and press Enter.

```
ACMEPACKET (session-router) # sip-config
```

From here, you can configure SIP parameters for HA nodes. To view all SIP configuration parameters, enter a **?** at the system prompt.

When configuring SIP for HA, you only need to set the parameters addressed in this procedure.

4. **red-sip-port**—Enter the port on which SIP signaling checkpointing messages are sent and received. The default value is **1988**. A value of **0** disables the SIP signaling checkpointing. The valid range is:
  - Minimum—0, 1024
  - Maximum—65535
5. **red-max-trans**—Enter the maximum size of the transaction list, or how many SIP transactions you want to store in memory at one time. Oldest transactions will be discarded first in the event that the limit is reached. The default value is **10000**. The valid range is:
  - Minimum—0
  - Maximum—999999999
6. **red-sync-start-time**—Enter the number of milliseconds before the Oracle Communications Session Border Controller will try to synchronize its signaling state checkpointing.

If the active Oracle Communications Session Border Controller is still adequately healthy, this timer will simply reset itself. If for any reason the active Oracle Communications Session Border Controller has become the standby, it will start to checkpoint with the newly active system when this timer expires.

We recommend that you leave this parameter set to its default, **5000**. The valid range is:

- Minimum—0
  - Maximum—999999999
7. **red-sync-comp-time**—Enter the number of milliseconds representing how frequently the standby Oracle Communications Session Border Controller checkpointing with the active Oracle Communications Session Border

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Controller to obtain the latest SIP signaling information. The first interval occurs after initial synchronizations of the systems.

We recommend that you leave this parameter set to its default, **1000**. The valid range is:

- Minimum—0
- Maximum—999999999

## Media State Checkpointing

---

By default, the Oracle Communications Session Border Controller performs media checkpointing across the HA node for all signaling protocols. You can keep the default port set for redundancy media flows.

H.323 media high availability is supported through a TCP socket keep-alive, which determines whether or not the other end of a TCP/IP network connection is still in fact connected. This type of checkpointing prevents the listening side of a connection from waiting indefinitely when a TCP connection is lost. When there is a switchover in the HA node, the system that has just become active takes over sending TCP keep-alives. Media continues to flow until the session ends or the flow guard timers expire.

This parameter will appear in the ACLI as follows:

```
red-flow-port 1985
```

## Media State Checkpointing Configuration

To configure media state checkpointing across an HA node in the ACLI:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# media-manager
```

3. Type **media-manager-config** and press Enter.

```
ACMEPACKET(media-manager)# media-manager-config
```

4. **red-flow-port**—Enter the port number for checkpointing media flows associated with the HA interface. This is the port where media flow checkpoint messages are sent and received.

Setting this field to **0** disables media state checkpointing. The default value is **1985**. The valid range is:

- Minimum—0, 1025
- Maximum—65535

## HA Media Interface Keepalive

---

In an HA node, it is possible for the two systems in the node to lose communication via the management (rear, wancom) interfaces. For example, wancom 1 and wancom 2 might become disconnected, and cause the heartbeat synchronization to fail. This type of failure causes communication errors because both systems try to assume the active role and thereby access resources reserved for the active system.

To avoid these types of conditions, you can enable an option instructing the standby system to take additional time before going to the active state. This check occurs through the system's media interfaces. Using it, the standby can determine whether or not there has been a true active failure.

In cases when the standby determines the active system has not truly failed, it will go out of service because it will have determined it no longer has up-to-date data from its active counterpart. You can restore functionality by re-establishing management (rear) interface communication between the system in the node, and then re-synchronizes the standby by rebooting it.

When you enable the media interface keepalive, the standby system in the HA node sends ARP requests to determine if the media interfaces' virtual IP address are active. There are two possible outcomes:

- If it receives responses to its ARP requests, the standby takes itself out of service—to prevent a conflict with the active.
- If it does not receive responses to its ARP requests within a timeout value you set, then standby assumes the active role in the HA node.

## Impact to Boot-Up Behavior

With the HA media interface keepalive enabled, the Oracle Communications Session Border Controller might be in the initial state longer than if the feature were disabled because it requires more information about the media (front) interfaces.

## HA Media Interface Keepalive Configuration

You turn the HA media interface keepalive on by setting a timeout value for the standby to receive responses to its ARP requests before it assumes the active role in the HA node. Keeping this parameter set to 0, its default, disables the keepalive

To enable the HA media interface keepalive:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **system** and press Enter.

```
ACMEPACKET(configure)# system
ACMEPACKET(system)#
```

3. Type **redundancy** and press Enter.

```
ACMEPACKET(session-router)# redundancy
ACMEPACKET(redundancy)#
```

If you are adding this feature to an existing configuration, then you will need to select the configuration you want to edit.

4. **media-if-peercheck-time**—Enter the amount of time in milliseconds for the standby system in an HA node to receive responses to its ARP requests via the media interface before it takes over the active role from its counterpart.

The default is 0, which turns the HA media interface keepalive off. The maximum value is 500 milliseconds.

5. Save and activate your configuration.

## RTC Notes

Starting in Release 4.1, the HA configuration is supported for real-time configuration (RTC). However, not all of the HA-related parameters are covered by RTC because of the impact on service it would cause to reconfigure these parameters dynamically.

This section sets out what parameters you should not dynamically reconfigure, or should dynamically reconfigure with care.

## HA

Changes to the following ACLI parameters will have the noted consequences when dynamically reconfigured:

- **cfg-max-trans**—Changing this value could cause the activation time to lengthen slightly

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- **init-time**, **becoming-standby-time**, and **becoming-active-time**—Changes take place only if the system is not transitioning between these states; otherwise the system waits until the transition is complete to make changes
- **percent-drift** and **advertisement-time**—Changes are communicated between nodes in the HA pair as part of regular health advertisements

In addition, the following parameters are not part of the RTC enhancement, for the reason specified in the right-hand column.

Parameter	Impact
state	Disrupts service
port	Disrupts service; leaves systems in an HA node without a means of communicating with each other
cfg-port	Disrupts service; leaves systems in an HA node without a means of communicating with each other
cfg-max-trans	Disrupts service
cfg-sync-start-time	Disrupts configuration replication
cfg-sync-comp-time	Disrupts configuration replication

## Protocol-Specific Parameters and RTC

In addition, you should not change any of the parameters related to HA that are part of protocol or media management configurations that are used for protocol/media checkpointing. These are:

- SIP configuration
  - **red-max-trans**
  - **red-sync-start-time**
  - **red-sync-comp-time**
- Media Manager configuration
  - **red-flow-port**
  - **red-max-trans**
  - **red-sync-start-time**
  - **red-sync-comp-time**



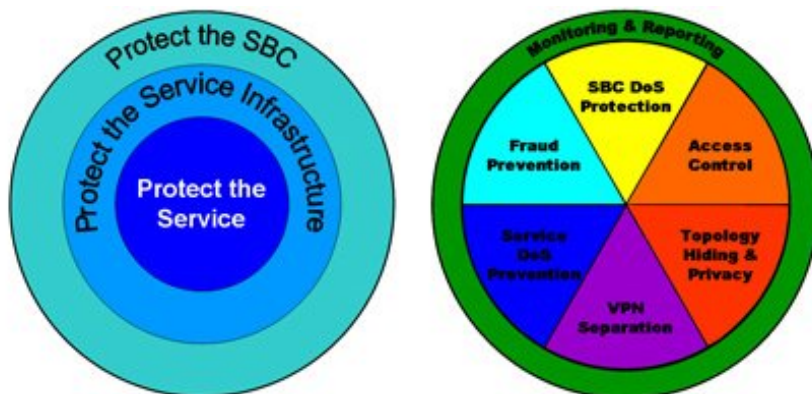
## Security

This chapter explains Oracle Communications Session Border Controller security, which is designed to provide security for administrative security, VoIP and other multimedia services. It includes Admin Security, access control, DoS attack, and overload protection, which help secure service and protect the network infrastructure (including the Oracle Communications Session Border Controller). In addition, Oracle Communications Session Border Controller security lets legitimate users still place calls during attack conditions; protecting the service itself.

### Security Overview

Oracle Communications Session Border Controller security includes the Net-SAFE framework's numerous features and architecture designs. Net-SAFE is a requirements framework for the components required to provide protection for the Session Border Controller (SBC), the service provider's infrastructure equipment (proxies, gateways, call agents, application servers, and so on), and the service itself.

The following diagrams illustrate Net-SAFE:



Each of Net-SAFE's seven functions consists of a collection of more specific features:

- Session border controller DoS protection: autonomic, SBC self-protection against malicious and non-malicious DoS attacks and overloads at Layers 2 to 4 (TCP, SYN, ICMP, fragments, and so on) and Layers 5 to 7 (SIP signaling floods, malformed messages, and so on).
- Access control: session-aware access control for signaling and media using static and dynamic permit/deny access control lists (ACLs) at layer 3 and 5.

## Security

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- Topology hiding and privacy: complete infrastructure topology hiding at all protocol layers for confidentiality and attack prevention security. Also, modification, removal or insertion of call signaling application headers and fields. Includes support for the SIP Privacy RFC.
- VPN separation: support for Virtual Private Networks (VPNs) with full inter-VPN topology hiding and separation, ability to create separate signaling and media-only VPNs, and with optional intra-VPN media hair-pinning to monitor calls within a VPN.
- Service infrastructure DoS prevention: per-device signaling and media overload control, with deep packet inspection and call rate control to prevent DoS attacks from reaching service infrastructure such as SIP servers, softswitches, application servers, media servers or media gateways.
- Fraud prevention: session-based authentication, authorization, and contract enforcement for signaling and media; and service theft protection.
- Monitoring and reporting: audit trails, event logs, access violation logs and traps, management access command recording, Call Detail Records (CDRs) with media performance monitoring, raw packet capture ability and lawful intercept capability. The monitoring method itself is also secured, through the use of SSH and SFTP, and through the ability to use a separate physical Ethernet port for management access.

## Administrative Security

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This section describes implications of installing and deleting the Admin Security license and the Admin Security ACP license on an Oracle Communications Session Border Controller (SBC).


These licenses enable the various security enhancements described in this document. In the absence of an Admin Security or Admin Security ACP license, these enhancements are not available.

As with any other license, an **activate-config** command must be executed after license installation for all changes to take effect. Certain ACLI aspects, such as login and password change prompts, change immediately after license installation.

These two licenses relate as follows:

1. Both licenses can exist together or separately on an SBC.
2. Removal of either or both licenses does not make available the protected areas of the system. This ensures that a system cannot be compromised by simply removing the Admin Security license(s).

 **Note:** The Admin Security or the Admin Security ACP feature sets are not intended for all customer use. Consult your Oracle representative to understand the ramifications of enabling these features.

 **Note:** Once the Admin Security or the Admin Security with ACP entitlement is provisioned, it can not be removed from the system in the field; your chassis must be returned to Oracle for replacement.

Admin Security Features for either license:

- telnet access is denied
- FTP access is denied
- history log access is denied
- shell access is denied
- additional password policy features are enabled

Additional Security features available with the Admin Security license:

- EMS (Element Management System) access is blocked
- ACP (Acme Control Protocol) is blocked

Additional Security features available with the Admin Security ACP license

- EMS (Element Management System) access is open
- ACP is open

The below table reflects feature availability under each license scenario.

Case	Admin Security Features	Shell Access	Telnet, FTP, and SSH Keys	History Log File in code/history	Password-policy	EMS access and ACP Ports
Only Admin Security license is present	enabled	denied	denied	denied	enabled	blocked
Only Admin Security license was deleted	disabled	denied	denied	denied	disabled	open
Only Admin Security ACP license is present	enabled	denied	denied	denied	enabled; <b>password-security-strength</b> is available	open
Only Admin Security ACP license was deleted	disabled	denied	denied	denied	disabled	open
Both are present	enabled	denied	denied	denied	enabled; <b>password-security-strength</b> is available	open
Both were present and only Admin-Security license was deleted	enabled	denied	denied	denied	enabled; <b>password-security-strength</b> is available	open
Both were present and only Admin-Security ACP license was deleted	enabled	denied	denied	denied	enabled; <b>password-security-strength</b> is not available	blocked
Both were present then both were deleted	disabled	denied	denied	denied	disabled	open

## Password Policy

Both the Admin Security and Admin Security ACP licenses support the creation of a password policy that enhances the authentication process by imposing requirements for:

- password length
- password strength
- password history and re-use
- password expiration and grace period

The Admin Security license restricts access to the ACP ports and mandates the following password length/strength requirements.

- user password must contain at least 9 characters
- admin password must contain at least 15 characters

The Admin Security and Admin Security ACP licenses both work to increase the security of the Oracle Communications Session Border Controller (SBC). If a device already has an Admin Security license installed, you can add an Admin Security ACP license later in certain high-security environments. Both licenses may co-exist on a single device, or either license may be on the device alone. An Admin Security ACP license performs the same functions as an Admin Security license, but also allows access to the ACP ports blocked by an Admin Security license.

- passwords must contain at least 2 lower case alphabetic characters
- passwords must contain at least 2 upper case alphabetic characters
- passwords must contain at least 2 numeric characters
- passwords must contain at least 2 special characters
- passwords must differ from the prior password by at least 4 characters
- passwords cannot contain, repeat, or reverse the user name
- passwords cannot contain three consecutive identical characters

The Admin Security ACP license imposes the same password length/strength requirements as above except for the minimum length requirement, and also maintains or reopens access to the ACP ports.

With the enabling of the password-strength command as part of the Admin Security ACP license, you also impose these requirements:

- passwords cannot contain two or more characters from the user ID
- passwords cannot contain a sequence of three or more characters from any password contained in the password history cache
- passwords cannot contain a sequence of two or more characters more than once
- passwords cannot contain either sequential numbers or characters, or repeated characters more than once.

In the absence of the Admin Security APC license, retain the default value (**disabled**). With the Admin Security APC license installed, use **enabled** to add the new password requirements as listed above; use **disabled** to retain only the password requirements defined by the Admin Security license.

Some specific password policy properties, specifically those regarding password lifetime and expiration procedures, are also applicable to SSH public keys used to authenticate client users.

## Configuring Password Policy Properties

The single instance **password-policy** configuration element defines the password policy.

1. From superuser mode, use the following command path to access password-policy configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# password-policy
ACMEPACKET(password-policy)#
```

The **password-policy** configuration element properties (with the introduction of the Admin Security or Admin Security ACP license) are shown below with their default values.

```
min-secure-pwd-length      8
expiry-interval            90
expiry-notify-period      30
grace-period               30
grace-logins                3
password-history-count     3
password-change-interval   24
password-policy-strength   disabled
```

2. The **min-secure-pwd-length** command is ignored when the Admin Security ACP license is installed and the **password-policy-strength** configuration element is set to **enabled**.
3. Use the **expiry-interval** command to specify the password lifetime in days. Password lifetime tracking begins when a password is changed.

Allowable values are integers within the range 1 through 65535, with a default value of 90 (days).

```
ACMEPACKET (password-policy) # expiry-interval 60
ACMEPACKET (password-policy) #
```

4. Use the **password-change-interval** command to specify the minimum password lifetime (the minimum time that must elapse between password changes.)

Allowable values are integers within the range 1 through 24, with a default value of 24 (hours).

```
ACMEPACKET (password-policy) # password-change-interval 18
ACMEPACKET (password-policy) #
```

5. Use the **expiry-notify-period** to specify the number of days prior to expiration that users begin to receive password expiration notifications.

Allowable values are integers within the range 1 through 90, with a default value of 30 (days).

During the notification period, users are reminded of impending password expiration at both Session Director login and logout.

```
ACMEPACKET (password-policy) # expiry-notify-period 10
ACMEPACKET (password-policy) #
```

6. Use the **grace-period** command in conjunction with the **grace-logins** command, to police user access after password expiration.

After password expiration, users are granted some number of logins (specified by the **grace-logins** command) for some number of days (specified by the **grace-period** command). Once the number of logins has been exceeded, or once the grace period has expired, the user is forced to change his or her password.

Allowable values for **grace-period** are integers within the range 1 through 90, with a default value of 30 (days).

Allowable values for **grace-logins** are integers within the range 1 through 10, with a default value of 3 (logins).

```
ACMEPACKET (password-policy) # grace-period 1
ACMEPACKET (password-policy) # grace-logins 1
ACMEPACKET (password-policy) #
```

7. Use the **password-history-count** command to specify the number of previously used passwords retained in encrypted format in the password history cache.

Allowable values are integers within the range 1 through 10, with a default value of 3 (retained passwords).

By default, a user's three most recently expired passwords are retained in the password history. As the user's current password is changed, that password is added to the history, replacing the oldest password entry.

New, proposed passwords are evaluated against the contents of the password cache, to prevent password re-use, and guard against minimal password changes.

```
ACMEPACKET (password-policy) # password-history-count 10
ACMEPACKET (password-policy) #
```

8. (Optional) Use the **password-policy-strength** command to enable the enhanced password strength requirements.

In the absence of the Admin Security ACP license, this command can be safely ignored.

**password-policy-strength** may be enabled when the Admin Security ACP license is enabled. This license includes all the password security features contained in the Admin Security license and also adds password strength requirements beyond those imposed by the Admin Security license. Specific new requirements are as follows:

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- passwords cannot contain two or more characters from the user ID  
For example, given a user ID of administrator, the password thispasswordistragic is not allowed because istra is a substring of administrator
- passwords cannot contain a sequence of three or more characters from any password contained in the password history cache
- passwords cannot contain a sequence of two or more characters more than once  
For example, ...w29W29... is legal; ...w29W29&&29... is not.
- passwords cannot contain either sequential numbers or characters, or repeated characters more than once  
For example, '66666', 'aaaa', 'abcd', 'fedc', '1234', '7654'.  
For example, 666, aaa abcd, fedc, 1234, and 7654 all render a password illegal.

In the absence of the Admin Security ACP license, retain the default value (**disabled**). With the Admin Security ACP license installed, use **enabled** to add the new password requirements as listed above; use **disabled** to retain only the password requirements defined by the Admin Security license.

```
ACMEPACKET(password-policy) # password-policy-strength enabled
ACMEPACKET(password-policy) #
```

9. Use **done**, **exit** and **verify-config** to complete password policy.

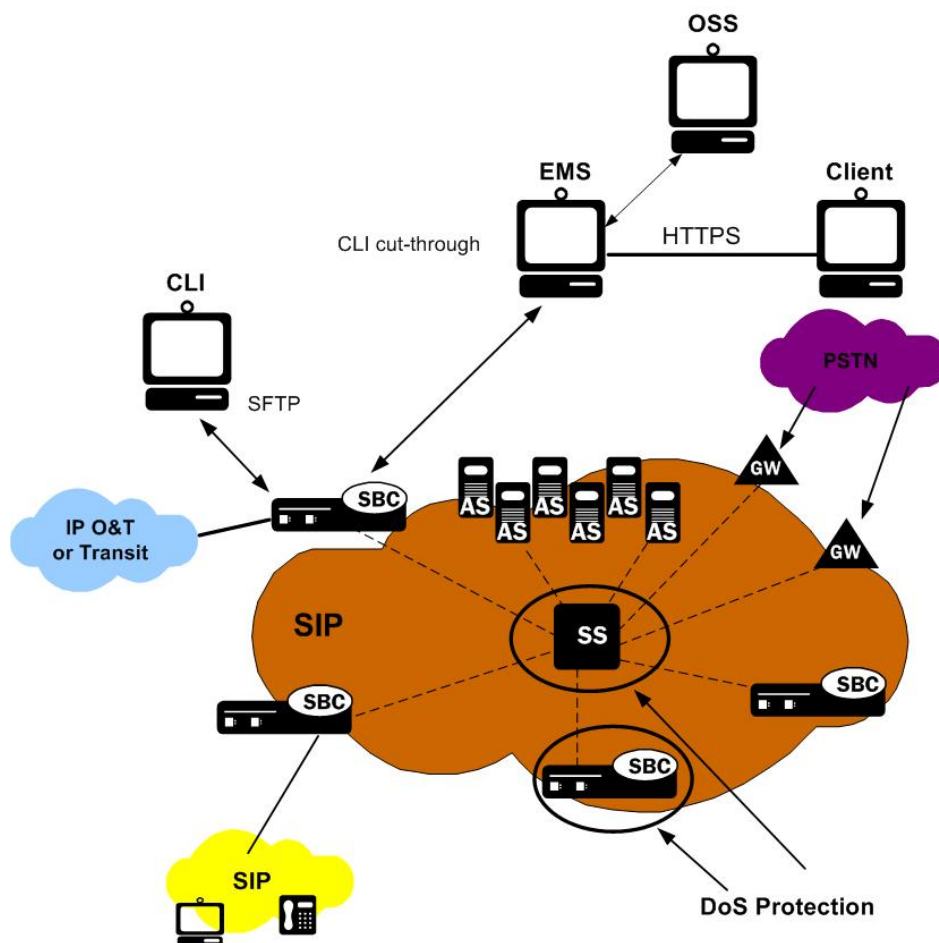
## Denial of Service Protection

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This section explains the Denial of Service (DoS) protection for the Oracle Communications Session Border Controller. The Oracle Communications Session Border Controller DoS protection functionality protects softswitches and gateways with overload protection, dynamic and static access control, and trusted device classification and separation at Layers 3-5. The Oracle Communications Session Border Controller itself is protected from signaling and media overload, but more importantly the feature allows legitimate, trusted devices to continue receiving service even during an attack. DoS protection prevents the Oracle Communications Session Border Controller host processor from being overwhelmed by a targeted DoS attack from the following:

- IP packets from an untrusted source as defined by provisioned or dynamic ACLs
- IP packets for unsupported or disabled protocols
- Nonconforming/malformed (garbage) packets to signaling ports
- Volume-based attack (flood) of valid or invalid call requests, signaling messages, and so on.
- Overload of valid or invalid call requests from legitimate, trusted sources

The following diagram illustrates DoS protection applied to the softswitch and to the Oracle Communications Session Border Controller.



## Levels of DoS Protection

The multi-level Oracle Communications Session Border Controller DoS protection consists of the following strategies:

- **Fast path filtering/access control:** access control for signaling packets destined for the Oracle Communications Session Border Controller host processor as well as media (RTP) packets. The Oracle Communications Session Border Controller performs media filtering by using the existing dynamic pinhole firewall capabilities. Fast path filtering packets destined for the host processor require the configuration and management of a trusted list and a deny list for each Oracle Communications Session Border Controller realm (although the actual devices can be dynamically trusted or denied by the Oracle Communications Session Border Controller based on configuration). You do not have to provision every endpoint/device on the Oracle Communications Session Border Controller, but instead retain the default values.
- **Host path protection:** includes flow classification, host path policing and unique signaling flow policing. Fast path filtering alone cannot protect the Oracle Communications Session Border Controller host processor from being overwhelmed by a malicious attack from a trusted source. The host path and individual signaling flows must be policed to ensure that a volume-based attack will not overwhelm the Oracle Communications Session Border Controller's normal call processing; and subsequently not overwhelm systems beyond it.

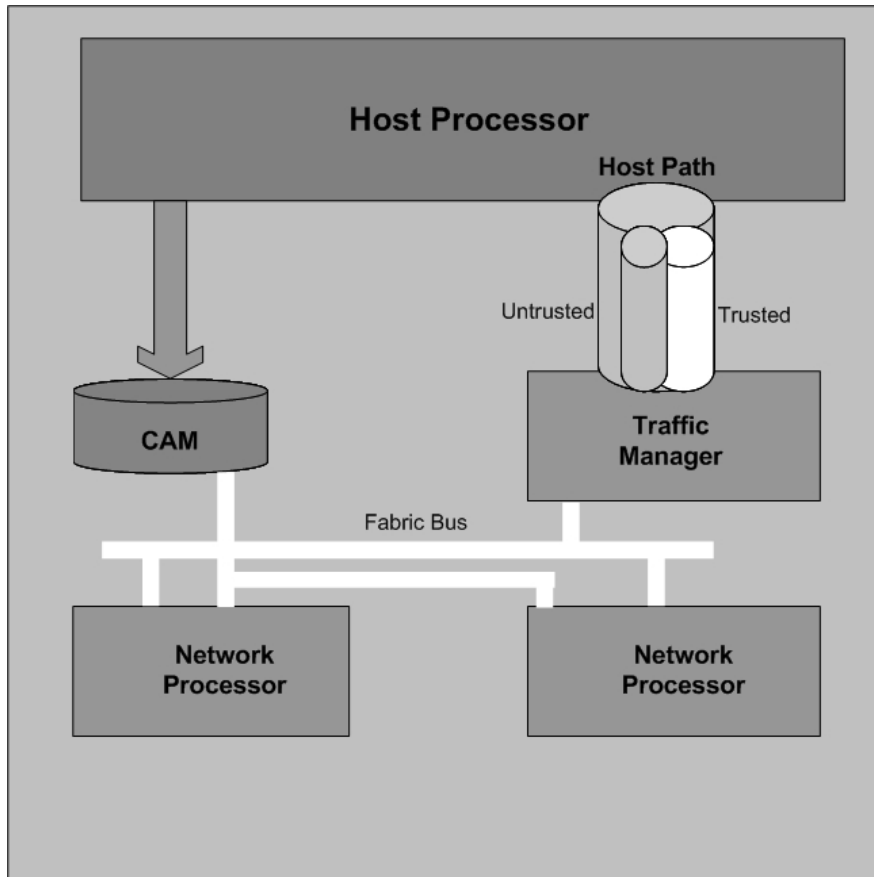
The Oracle Communications Session Border Controller must classify each source based on its ability to pass certain criteria that is signaling- and application-dependent. At first each source is considered untrusted with the possibility of being promoted to fully trusted. The Oracle Communications Session Border Controller maintains two host paths, one for each class of traffic (trusted and untrusted), with different policing characteristics to ensure that fully trusted traffic always gets precedence.

- **Host-based malicious source detection and isolation – dynamic deny list.** Malicious sources can be automatically detected in real-time and denied in the fast path to block them from reaching the host processor.

## About the Process

DoS attacks are handled in the Oracle Communications Session Border Controller's host path. The Oracle Communications Session Border Controller uses NAT table entries to filter out undesirable IP addresses; creating a deny list. After a packet from an endpoint is accepted through NAT filtering, policing is implemented in the Traffic Manager subsystem based on the sender's IP address. NAT table entries distinguish signaling packets coming in from different sources for policing purposes. The maximum number of policed calls that the Oracle Communications Session Border Controller can support is 16K (on 32K CAM / IDT CAM).

The Traffic Manager has two pipes, trusted and untrusted, for the signaling path. Each signaling packet destined for the host CPU traverses one of these two pipes.



## Trusted Path

Packets from trusted devices travel through the trusted pipe in their own individual queues. In the Trusted path, each trusted device flow has its own individual queue (or pipe). The Oracle Communications Session Border Controller can dynamically add device flows to the trusted list by promoting them from the Untrusted path based on behavior; or they can be statically provisioned.

Trusted traffic is put into its own queue and defined as a device flow based on the following:

- source IP address
- source UDP/TCP port number
- destination IP address
- destination UDP/TCP port (SIP interface to which it is sending)
- realm it belongs to, which inherits the Ethernet interface and VLAN it came in on

For example, SIP packets coming from 10.1.2.3 with UDP port 1234 to the Oracle Communications Session Border Controller SIP interface address 11.9.8.7 port 5060, on VLAN 3 of Ethernet interface 0:1, are in a separate Trusted



queue and policed independently from SIP packets coming from 10.1.2.3 with UDP port 3456 to the same Oracle Communications Session Border Controller address, port and interface.

Data in this flow is policed according to the configured parameters for the specific device flow, if statically provisioned. Alternatively, the realm to which endpoints belong have a default policing value that every device flow will use. The defaults configured in the realm mean each device flow gets its own queue using the policing values. As shown in the previous example, if both device flows are from the same realm and the realm is configured to have an average rate limit of 10K bytes per second (10KBps), each device flow will have its own 10KBps queue. They are not aggregated into a 10KBps queue.

The individual flow queues and policing lets the Oracle Communications Session Border Controller provide each trusted device its own share of the signaling, separate the device's traffic from other trusted and untrusted traffic, and police its traffic so that it can't attack or overload the Oracle Communications Session Border Controller (therefore it is trusted, but not completely).

### Address Resolution Protocol Flow

The Address Resolution Protocol (ARP) packets are given their own trusted flow with the bandwidth limitation of 8 Kbps. ARP packets are able to flow smoothly, even when a DoS attack is occurring.

## Untrusted Path

Packets (fragmented and unfragmented) that are not part of the trusted or denied list travel through the untrusted pipe. In the untrusted path, traffic from each user/device goes into one of 2048 queues with other untrusted traffic. Packets from a single device flow always use the same queue of the 2048 untrusted queues, and 1/2048th of the untrusted population also uses that same queue. To prevent one untrusted endpoint from using all the pipe's bandwidth, the 2048 flows defined within the path are scheduled in a fair-access method. As soon as the Oracle Communications Session Border Controller decides the device flow is legitimate, it will promote it to its own trusted queue.

All 2048 untrusted queues have dynamic sizing ability, which allows one untrusted queue to grow in size, as long as other untrusted queues are not being used proportionally as much. This dynamic queue sizing allows one queue to use more than average when it is available. For example, in the case where one device flow represents a PBX or some other larger volume device. If the overall amount of untrusted packets grows too large, the queue sizes rebalance, so that a flood attack or DoS attack does not create excessive delay for other untrusted devices.

In the usual attack situations, the signaling processor detects the attack and dynamically demotes the device to denied in the hardware by adding it to the deny ACL list. Even if the Oracle Communications Session Border Controller does not detect an attack, the untrusted path gets serviced by the signaling processor in a fair access mechanism. An attack by an untrusted device will only impact 1/1000th of the overall population of untrusted devices, in the worst case. Even then there's a probability of users in the same 1/1000th percentile getting in and getting promoted to trusted.

### IP Fragment Packet Flow

All fragment packets are sent through their own 1024 untrusted flows in the Traffic Manager. The first ten bits (LSB) of the source address are used to determine which fragment-flow the packet belongs to. These 1024 fragment flows share untrusted bandwidth with already existing untrusted-flows. In total, there are 2049 untrusted flows: 1024-non-fragment flows, 1024 fragment flows, and 1 control flow.

Fragmented ICMP packets are qualified as ICMP packets rather than fragment packets. Fragment and non-fragmented ICMP packets follow the trusted-ICMP-flow in the Traffic Manager, with a bandwidth limit of 8Kbs.

### Fragment Packet Loss Prevention

You can set the maximum amount of bandwidth (in the **max-untrusted-signaling** parameter) you want to use for untrusted packets. However, because untrusted and fragment packets share the same amount of bandwidth for policing, any flood of untrusted packets can cause the Oracle Communications Session Border Controller to drop fragment packets.

To prevent fragment packet loss, you can set the **fragment-msg-bandwidth**. When it is set to any value other than 0 (which disables it), the Oracle Communications Session Border Controller:

- Provides for a separate policing queue for fragment packets (separate from that used for untrusted packets)

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- Uses this new queue to prevent fragment packet loss when there is a flood from untrusted endpoints.

When you set up a queue for fragment packets, untrusted packets likewise have their own queue—meaning also that the **max-untrusted-signaling** and **min-untrusted-signaling** values are applied to the untrusted queue.

## Static and Dynamic ACL Entry Limits

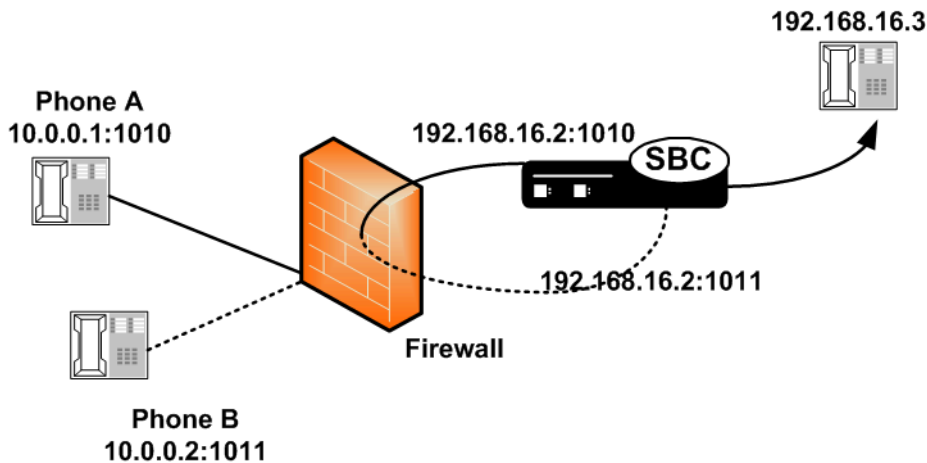
The Oracle Communications Session Border Controller can simultaneously police a maximum of 250,000 trusted device flows, while at the same time denying an additional 32,000 attackers. If list space becomes full and additional device flows need to be added, the oldest entries in the list are removed and the new device flows are added.

## Dynamic Deny for HNT

Dynamic deny for HNT has been implemented on the Oracle Communications Session Border Controller for cases when callers are behind a NAT or firewall. Without this feature, if one caller behind a NAT or firewall were denied, the Oracle Communications Session Border Controller would also deny all other users behind the same NAT or firewall. This would be true even for endpoints behind the firewall that had not crossed threshold limits you set for their realm; all endpoints behind the firewall would go out of service. In the following diagram, both Phone A and Phone B would be denied because their IP addresses would be translated by the firewall to the same IPv4 address (192.168.16.2).

However, dynamic deny for HNT allows the Oracle Communications Session Border Controller to determine, based on the UDP/TCP port, which endpoints should be denied and which should be allowed. The Oracle Communications Session Border Controller can determine that even though multiple endpoints originating behind a firewall appear with the same IPv4 address, those addresses use different ports and are unique.

As shown in the diagram below, the ports from Phone A and Phone B remain unchanged. This way, if Phone A violates the thresholds you have configured, the Oracle Communications Session Border Controller can block traffic from Phone A while still accepting traffic from Phone B.



## Host and Media Path Protection Process

The Oracle Communications Session Border Controller Network Processors (NPs) check the deny and permit lists for received packets, and classify them as trusted, untrusted or denied (discard). Only packets to signaling ports and dynamically signaled media ports are permitted. All other packets sent to Oracle Communications Session Border Controller ports are filtered. Only packets from trusted and untrusted (unknown) sources are permitted; any packet from a denied source is dropped by the NP hardware. The Traffic Manager manages bandwidth policing for trusted and untrusted traffic, as described earlier. Malicious traffic is detected in the host processor and the offending device is dynamically added to denied list, which enables early discard by the NP. Devices become trusted based on behavior detected by the Signaling Processor, and dynamically added to the trusted list. This process enables the proper classification by the NP hardware. All other traffic is untrusted (unknown).

## Session Director Access Control

You can create static trusted/untrusted/deny lists with source IP addresses or IP address prefixes, UDP/TCP port number or ranges, and based on the appropriate signaling protocols. Furthermore, the Oracle Communications Session Border Controller can dynamically promote and demote device flows based on the behavior, and thus dynamically creates trusted, untrusted, and denied list entries.

### Access Control for Hosts

ACLs are supported for all VoIP signaling protocols on the Oracle Communications Session Border Controller: SIP and H.323. The Oracle Communications Session Border Controller loads ACLs so they are applied when signaling ports are loaded. The following rules apply to static NAT entries based on your configuration:

- If there are no ACLs applied to a realm that have the same configured trust level as that realm, the Oracle Communications Session Border Controller adds a default NAT entry using the realm parameters.
- If you configure a realm with none as its trust level and you have configured ACLs, the Oracle Communications Session Border Controller only applies the ACLs.
- If you set a trust level for the ACL that is lower than the one you set for the realm, the Oracle Communications Session Border Controller will not add a separate NAT entry for the ACL.

ACLs provide access control based on destination addresses when you configure destination addresses as a way to filter traffic. You can set up a list of access control exceptions based on the source or the destination of the traffic.

For dynamic ACLs based on the promotion and demotion of endpoints, the rules of the matching ACL are applied.

## Access Control Endpoint Classification Capacity and DoS

The following capacities are for both IPv4 and IPv6 endpoints.

Hardware	Software	Denied	Trusted	Media	Untrusted	Dynamic Trusted	ARP
Net-Net 6300	7.1.2	32768	16384	160000	16384	500000	16386
Net-Net 4500	7.1.2	8192	4096	64000	4096	250000	4104

## Media Access Control

The media access control consists of media path protection and pinholes through the firewall. Only RTP and RTCP packets from ports dynamically negotiated through signaling (SIP and H.323) are allowed, which reduces the chance of RTP hijacking. Media access depends on both the destination and source RTP/RTCP UDP port numbers being correct, for both sides of the call.

## Host Path Traffic Management

The host path traffic management consists of the dual host paths discussed earlier:

- Trusted path is for traffic classified by the system as trusted. You can initially define trusted traffic by ACLs, as well as by dynamically promoting it through successful SIP registration, or a successful call establishment. You can configure specific policing parameters per ACL, as well as define default policing values for dynamically-classified flows. Traffic for each trusted device flow is limited from exceeding the configured values in hardware. Even an attack from a trusted, or spoofed trusted, device cannot impact the system.
- Untrusted path is the default for all unknown traffic that has not been statically provisioned otherwise. For example, traffic from unregistered endpoints. Pre-configured bandwidth policing for all hosts in the untrusted path occurs on a per-queue and aggregate basis.

## Traffic Promotion

Traffic is promoted from untrusted to trusted list when the following occurs:

- successful SIP registration for SIP endpoints
- successful session establishment for SIP calls

### Malicious Source Blocking

Malicious source blocking consists of monitoring the following metrics for each source:

- SIP transaction rate (messages per second)
- SIP call rate (call attempts per second)
- Nonconformance/invalid signaling packet rate

Device flows that exceed the configured invalid signaling threshold, or the configured valid signaling threshold, within the configured time period are demoted, either from trusted to untrusted, or from untrusted to denied classification.

### Blocking Actions

Blocking actions include the following:

- Dynamic deny entry added, which can be viewed through the ACLI.
- SNMP trap generated, identifying the malicious source

Dynamically added deny entries expire and are promoted back to untrusted after a configured default deny period time. You can also manually clear a dynamically added entry from the denied list using the ACLI.

### Protecting Against Session Agent Overloads

You can prevent session agent overloads with registrations by specifying the registrations per second that can be sent to a session agent.

### ARP Flood Protection Enhancements

Enhancements have been made to the way the Oracle Communications Session Border Controller provides ARP flood protection. In releases prior to Release C5.0, there is one queue for both ARP requests and responses, which the Oracle Communications Session Border Controller polices at a non-configurable limit (eight kilobytes per second). This method of ARP protection can cause problems during an ARP flood, however. For instance, gateway heartbeats the Oracle Communications Session Border Controller uses to verify (via ARP) reachability for default and secondary gateways could be throttled; the Oracle Communications Session Border Controller would then deem the router or the path to it unreachable, decrement the system's health score accordingly. Another example is when local routers send ARP requests for the Oracle Communications Session Border Controller's address are throttled in the queue; the Oracle Communications Session Border Controller never receives the request and so never responds, risking service outage.

The solution implemented to resolve this issue is to divide the ARP queue in two, resulting in one ARP queue for requests and a second for responses. This way, the gateway heartbeat is protected because ARP responses can no longer be flooded from beyond the local subnet. In addition, the Oracle Communications Session Border Controllers in HA nodes generate gateway heartbeats using their shared virtual MAC address for the virtual interface.

In addition, this solution implements a configurable ARP queue policing rate so that you are not committed to the eight kilobytes per second used as the default in prior releases. The previous default is not sufficient for some subnets, and higher settings resolve the issue with local routers sending ARP request to the Oracle Communications Session Border Controller that never reach it or receive a response.

As a security measure, in order to mitigate the effect of the ARP table reaching its capacity, configuring the media-manager option, **active-arp**, is advised. Enabling this option causes all ARP entries to get refreshed every 20 minutes.

### Dynamic Demotion for NAT Devices

In addition to the various ways the Oracle Communications Session Border Controller already allows you to promote and demote devices to protect itself and other network elements from DoS attacks, it can now block off an entire NAT device. The Oracle Communications Session Border Controller can detect when a configurable number of devices behind a NAT have been blocked off, and then shut off the entire NAT's access.

This dynamic demotion of NAT devices can be enabled for an access control (ACL) configuration or for a realm configuration. When you enable the feature, the Oracle Communications Session Border Controller tracks the number of endpoints behind a single NAT that have been labeled untrusted. It shuts off the NAT's access when the number reaches the limit you set.

The demoted NAT device then remains on the untrusted list for the length of the time you set in the **deny-period**.

## Configuring DoS Security

---

This section explains how to configure the Oracle Communications Session Border Controller for DoS protection.

### Configuration Overview

Configuring Oracle Communications Session Border Controller DoS protection includes masking source IP and port parameters to include more than one match and configuring guaranteed minimum bandwidth for trusted and untrusted signaling path. You can also configure signaling path policing parameters for individual source addresses. Policing parameters are defined as peak data rate (in bytes/sec), average data rate (in bytes/sec), and maximum burst size.

You can configure deny list rules based on the following:

- ingress realm
- source IP address
- source port
- transport protocol (TCP/UDP)
- application protocol (SIP or H.323)

### Changing the Default Oracle Communications Session Border Controller Behavior

The Oracle Communications Session Border Controller automatically creates permit untrusted ACLs that let all sources (address prefix of 0.0.0.0/0) reach each configured realm's signaling interfaces, regardless of the realm's address prefix. To deny sources or classify them as trusted, you create static or dynamic ACLs, and the global permit untrusted ACL to specifically deny sources or classify them as trusted. Doing this creates a default permit-all policy with specific deny and permit ACLs based on the realm address prefix.

You can change that behavior by configuring static ACLs for realms with the same source prefix as the realm's address prefix; and with the trust level set to the same value as the realm. Doing this prevents the permit untrusted ACLs from being installed. You then have a default deny all ACL policy with specific static permit ACLs to allow packets into the system.

#### Example 1 Limiting Access to a Specific Address Prefix Range

The following example shows how to install a permit untrusted ACL of source 12.34.0.0/16 for each signalling interface/port of a realm called access. Only packets from within the source address prefix range 12.34.0.0/16, destined for the signaling interfaces/port of the realm named access, are allowed. The packets go into untrusted queues until they are dynamically demoted or promoted based on their behavior. All other packets are denied/dropped.

- Configure a realm called access and set the trust level to low and the address prefix to 12.34.0.0/16.
- Configure a static ACL with a source prefix of 12.34.0.0/16 with the trust level set to low for the realm named access.

#### Example 2 Classifying the Packets as Trusted

Building on Example 1, this example shows how to classify all packets from 12.34.0.0/16 to the realm signaling interfaces as trusted and place them in a trusted queue. All other packets from outside the prefix range destined to the realm's signaling interfaces are allowed and classified as untrusted; then promoted or demoted based on behavior.

You do this by adding a global permit untrusted ACL (source 0.0.0.0) for each signaling interface/port of the access realm. You configure a static ACL with a source prefix 12.34.0.0/16 and set the trust level to high.

Adding this ACL causes the Oracle Communications Session Border Controller to also add a permit trusted ACL with a source prefix of 12.34.0.0/16 for each signaling interface/port of the access realm. This ACL is added because the trust level of the ACL you just added is high and the realm's trust level is set to low. The trust levels must match to remove the global permit trusted ACL.

### Example 3 Installing Only Static ACLs

This example shows you how to prevent the Oracle Communications Session Border Controller from installing the global permit (0.0.0.0) untrusted ACL.

- Configure a realm with a trust level of none.
- Configure static ACLs for that realm with the same source address prefix as the realm's address prefix, and set the trust level to any value.

The system installs only the static ACLs you configure.

## Access Control List Configuration

To configure access control lists:

1. Access the access-control configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# access-control
ACMEPACKET(access-control)#
```

2. Type select to choose and configure an existing object.

```
ACMEPACKET(access-control)# select
<src-ip>:
1: src 0.0.0.0; 0.0.0.0; realm01; ; ALL
```

3. **realm-id**—Enter the ID of the host's ingress realm.

4. **source-address**—Enter the source IPv4 address and port number for the host in the following format:

```
<IP address>[/number of address bits][:<port>][/<port bits>]
```

For example:

```
10.0.0.1/24:5000/14
10.0.0.1/16
10.0.0.1/24:5000
10.0.0.1:5000
```

You do not need to specify the number of address bits if you want all 32 bits of the address to be matched. You also do not need to specify the port bits if you want the exact port number matched. If you do not set the port mask value or if you set it to 0, the exact port number will be used for matching. The default value is 0 . 0 . 0 . 0 .

5. **destination-address**—(Is ignored if you configure an application protocol in step 7.) Enter the destination IPv4 address and port for the destination in the following format:

```
<IP address>[/number of address bits][:<port>[/<port bits>]]
```

You do not need to specify the number of address bits if you want all 32 bits of the address to be matched. You also do not need to specify the port bits if you want the exact port number matched. If you do not set the port mask value or if you set it to 0, the exact port number will be used for matching. The default value is 0 . 0 . 0 . 0 .

6. **application-protocol**—Enter the application protocol type for this ACL entry. The valid values are:

- SIP | H.323 | None



**Note:** If application-protocol is set to none, the destination-address and port will be used. Ensure that your destination-address is set to a non-default value (0.0.0.0.)

7. **transport-protocol**—Select the transport-layer protocol configured for this ACL entry. The default value is **ALL**. The valid values are:
  - ALL | TCP | UDP
8. **access**—Enter the access control type or trusted list based on the trust-level parameter configuration for this host. The default value is **permit**. The valid values are:
  - **permit**—Puts the entry into the untrusted list. The entry is promoted or demoted according to the trust level set for this host.
  - **deny**—Puts the entry in the deny list.
9. **average-rate-limit**—Indicate the sustained rate in bytes per second for host path traffic from a trusted source within the realm. The default value is **0**. A value of **0** means policing is disabled. The valid range is:
  - Minimum—0
  - Maximum—999999999
10. **trust-level**—Indicate the trust level for the host with the realm. The default value is **none**. The valid values are:
  - **none**—Host is always untrusted. It is never promoted to the trusted list or demoted to the deny list.
  - **low**—Host can be promoted to the trusted list or demoted to the deny list.
  - **medium**—Host can be promoted to the trusted list but is only demoted to untrusted. It is never added to the deny list.
  - **high**—Host is always trusted.
11. **invalid-signal-threshold**— Enter the number of invalid signaling messages that trigger host demotion. The value you enter here is only valid when the trust level is low or medium. Available values are:
  - Minimum—Zero (0) is disabled.
  - Maximum—999999999

If the number of invalid messages exceeds this value based on the tolerance window parameter, configured in the media manager, the host is demoted.

The tolerance window default is 30 seconds. Bear in mind, however, that the system uses the same calculation it uses for specifying "recent" statistics in show commands to determine when the number of signaling messages exceeds this threshold. This calculation specifies a consistent start time for each time period to compensate for the fact that the event time, such as a user running a show command, almost never falls on a time-period's border. This provides more consistent periods of time for measuring event counts.

The result is that this invalid signal count increments for two tolerance windows, 60 seconds by default, within which the system monitors whether or not to demote the host. The signal count for the current tolerance window is always added to the signal count of the previous tolerance window and compared against your setting.
12. **maximum-signal-threshold**—Set the maximum number of signaling messages the host can send within the tolerance window. The value you enter here is only valid when the trust level is low or medium. The default value is **0**, disabling this parameter. The valid range is:
  - Minimum—0
  - Maximum—999999999

If the number of messages received exceeds this value within the tolerance window, the host is demoted.
13. **untrusted-signal-threshold**—Set the maximum number of untrusted messages the host can send within the tolerance window. Use to configure different values for trusted and un-trusted endpoints for valid signaling message parameters. Also configurable per realm. The default value is **0**, disabling this parameter. The valid range is:
  - Minimum—0
  - Maximum—999999999

14. **deny-period**—Indicate the time period in seconds after which the entry for this host is removed from the deny list. The default value is 30. The valid range is:

- Minimum—0
- Maximum—999999999

15. **nat-trust-threshold**—Enter the number of endpoints behind a NAT that must be denied for the Oracle Communications Session Border Controller to demote the NAT device itself to denied (dynamic demotion of NAT devices). The default is 0, meaning dynamic demotion of NAT devices is disabled. The range is from 0 to 65535.

The following example shows access control configured for a host in the external realm.

```
access-control
  realm-id                external
  source-address          192.168.200.215
  destination-address     192.168.10.2:5000
  application-protocol    SIP
  transport-protocol      ALL
  access                  permit
  average-rate-limit      3343
  trust-level             low
  invalid-signal-threshold 5454
  maximum-signal-threshold 0
  untrusted-signal-threshold 0
  deny-period             0
```

The following example of how to configure a black-list entry:

```
access-control
  realm-id                external
  source-address          192.168.200.200
  destination-address     192.168.10.2:5000
  application-protocol    SIP
  transport-protocol      ALL
  access                  deny
  average-rate-limit      0
  trust-level             none
  invalid-signal-threshold 0
  maximum-signal-threshold 0
  untrusted-signal-threshold 0
  deny-period             0
```

## Host Access Policing

You can configure the Oracle Communications Session Border Controller to police the overall bandwidth of the host path.

To configure host access policing:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# media-manager
```

3. Type **media-manager** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# media-manager
ACMEPACKET(media-manager-config)#
```

4. **max-signaling-bandwidth**—Set the maximum overall bandwidth available for the host path in bytes per second, which includes signaling messages from trusted and untrusted sources. It also includes any Telnet and FTP traffic on media ports. The default value is 1000000. The valid range is:

- Minimum—71000



- Maximum—10000000
5. **max-untrusted-signaling**—Set the percentage of the maximum signaling bandwidth you want to make available for messages coming from untrusted sources. This bandwidth is only available when not being used by trusted sources. The default value is **100**. The valid range is:
    - Minimum—1
    - Maximum—100
  6. **min-untrusted-signaling**—Set the percentage of the maximum signaling bandwidth you want reserved for the untrusted sources. The rest of the bandwidth is available for trusted resources, but can also be used for untrusted sources (see max-untrusted-signaling). The default value is **30**. The valid range is:
    - Minimum—1
    - Maximum—100
  7. **fragment-msg-bandwidth**—Enter the amount of bandwidth to use for the fragment packet queue. If you leave this parameter set to 0, then the Oracle Communications Session Border Controller will use the same queue for and share bandwidth between untrusted packets and fragment packets. The default value is zero (**0**). The valid range is:
    - Minimum—0
    - Maximum—10000000
  8. **tolerance-window**—Set the size of the window used to measure host access limits. The value entered here is used to measure the invalid message rate and maximum message rate for the realm configuration. The default value is **30**. The valid range is:
    - Minimum—0
    - Maximum—999999999


The following example shows a host access policing configuration.

```
media-manager
  state                enabled
  latching             enabled
  flow-time-limit      86400
  initial-guard-timer  300
  subsq-guard-timer    300
  tcp-flow-time-limit  86400
  tcp-initial-guard-timer 300
  tcp-subsq-guard-timer 300
  tcp-number-of-ports-per-flow 2
  hnt-rtcp             disabled
  algd-log-level       WARNING
  mbcd-log-level       WARNING
  home-realm-id
  red-flow-port        1985
  red-max-trans        10000
  red-sync-start-time  5000
  red-sync-comp-time   1000
  max-signaling-bandwidth 1000000
  max-untrusted-signaling 50
  min-untrusted-signaling 30
  tolerance-window     30
  rtcp-rate-limit      0
```

## Configuring ARP Flood Protection

You do not need to configure the Oracle Communications Session Border Controller to enable the use of two separate ARP queues; that feature is enabled automatically.

If you want to configure the ARP queue policing rate, you can do so in the media manager configuration.

-  **Note:** this feature is not RTC-supported, and you must reboot your Oracle Communications Session Border Controller in order for your configuration changes to take effect.

To set the ARP queue policing rate:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager  
ACMEPACKET (media-manager) #
```

3. Enter **media-manager** and press <Enter>.

```
ACMEPACKET (media-manager) # media-manager  
ACMEPACKET (media-manager-config) #
```

4. **arp-msg-bandwidth**—Enter the rate at which you want the Oracle Communications Session Border Controller to police the ARP queue; the value you enter is the bandwidth limitation in bytes per second. The default value is **32000**. The valid range is:

- Minimum—2000
- Maximum—200000

5. Save your configuration.
6. Reboot your Oracle Communications Session Border Controller.

## Access Control for a Realm

Each host within a realm can be policed based on average rate, peak rate, and maximum burst size of signaling messages. These parameters take effect only when the host is trusted. You can also set the trust level for the host within the realm. All untrusted hosts share the bandwidth defined for the media manager: maximum untrusted bandwidth and minimum untrusted bandwidth.

To configure access control for a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config  
ACMEPACKET (realm-config) #
```

4. **addr-prefix**—Set the IP address prefix used to determine if an IP address is associated with the realm. This value is then associated with the ACLs you create to determine packet access. The default value is **0.0.0.0**.
5. **average-rate-limit**—Set the sustained rate for host path traffic from a trusted source within the realm in bytes per second. The default value is zero (**0**), disabling this parameter. The valid range is:

- Minimum—0
- Maximum—4294967295

6. **access-control-trust-level**—Set the trust level for the host within the realm. The default value is **none**. The valid values are:

- **none**—Host is always untrusted. It is never promoted to the trusted list or demoted to the deny list.
- **low**—Host can be promoted to the trusted list or demoted to the deny list.
- **medium**—Host can be promoted to the trusted list but is only demoted to untrusted. It is never added to the deny list.
- **high**—Host is always trusted.

7. **invalid-signal-threshold**— Enter the number of invalid signaling messages that trigger host demotion. The value you enter here is only valid when the trust level is low or medium. Available values are:

- Minimum—Zero (0) is disabled.
- Maximum—999999999

If the number of invalid messages exceeds this value based on the tolerance window parameter, configured in the media manager, the host is demoted.

The tolerance window default is 30 seconds. Bear in mind, however, that the system uses the same calculation it uses for specifying "recent" statistics in show commands to determine when the number of signaling messages exceeds this threshold. This calculation specifies a consistent start time for each time period to compensate for the fact that the event time, such as a user running a show command, almost never falls on a time-period's border. This provides more consistent periods of time for measuring event counts.

The result is that this invalid signal count increments for two tolerance windows, 60 seconds by default, within which the system monitors whether or not to demote the host. The signal count for the current tolerance window is always added to the signal count of the previous tolerance window and compared against your setting.

8. **maximum-signal-threshold**—Set the maximum number of signaling messages one host can send within the window of tolerance. The host is demoted if the number of messages received by the Oracle Communications Session Border Controller exceeds the number set here. Valid only when the trust level is set to low or medium. The default value is zero (0), disabling this parameter. The valid range is:

- Minimum—0
- Maximum—4294967295

9. **untrusted-signal-threshold**—Set the maximum number of untrusted messages the host can send within the tolerance window. Use to configure different values for trusted and un-trusted endpoints for valid signaling message parameters. Also configurable per realm. The default value is zero (0), disabling the parameter. The valid range is:

- Minimum—0
- Maximum—4294967295

10. **deny-period**—Set the length of time an entry is posted on the deny list. The host is deleted from the deny list after this time period. The default value is 30. A value of 0 disables the parameter. The valid range is:

- Minimum—0
- Maximum—4294967295

11. **nat-trust-threshold**—Enter the number of endpoints behind a NAT that must be denied for the Oracle Communications Session Border Controller to demote the NAT device itself to denied (dynamic demotion of NAT devices). The default is 0, meaning dynamic demotion of NAT devices is disabled. The range is from 0 to 65535.

The following example shows a host access policing configuration.

```
realm-config
  identifier                private
  addr-prefix               192.168.200.0/24
  network-interfaces

  mm-in-realm               private:0
  mm-in-network             disabled
  msm-release               enabled
  qos-enable                disabled
  max-bandwidth             0
  ext-policy-svr            0
  max-latency               0
  max-jitter                0
  max-packet-loss           0
  observ-window-size       0
  parent-realm
  dns-realm
```

```

media-policy
in-translationid
out-translationid
class-profile
average-rate-limit           8000
access-control-trust-level   medium
invalid-signal-threshold     200
maximum-signal-threshold     0
untrusted-signal-threshold   500
deny-period                  30
symmetric-latching          disabled
pai-strip                    disabled
trunk-context

```

## Configuring Overload Protection for Session Agents

The Oracle Communications Session Border Controller offers two methods to control SIP registrations to smooth the registration flow.

You can limit the:

- number of new register requests sent to a session agent (using the **max-register-sustain-rate** parameter)
- burstiness which can be associated with SIP registrations

The first method guards against the Oracle Communications Session Border Controller's becoming overwhelmed with register requests, while the second method guards against a transient registration that can require more than available registration resources.

SIP registration burst rate control allows you to configure two new parameters per SIP session agent—one that controls the registration burst rate to limit the number of new registration requests, and a second to set the time window for that burst rate. When the registration rate exceeds the burst rate you set, the Oracle Communications Session Border Controller responds to new registration requests with 503 Service Unavailable messages.

Note that this constraint is not applied to re-registers resulting from a 401 Unauthorized challenge request.

To configure overload protection for session agents:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # session-agent
ACMEPACKET (session-agent) #
```

4. **constraints**—Enable this parameter to set the sustained rate window constraint you configure in the next step. The default value is **disabled**. The valid values are:
  - enabled | disabled
5. **sustain-rate-window**—Enter a number to set the sustained window period (in milliseconds) that is used to measure the sustained rate. The default value is zero (0). The valid range is:
  - Minimum—10
  - Maximum—4294967295

The value you set here must be higher than or equal to the value you set for the burst rate window.



**Note:** If you are going to use this parameter, you must set it to a minimum value of 10.

6. **max-register-sustain-rate**—Enter a number to set the maximum number of registrations per second you want sent to the session agent. The default value is zero (0), disabling the parameter. The valid range is:
  - Minimum—0
  - Maximum—4294967295
7. **register-burst-window**—Define the window size in seconds for the maximum number of allowable SIP registrations. 0 is the minimum and default value for this parameter; the maximum value is 999999999.
8. **max-register-burst-rate**—Enter the maximum number of new registrations you want this session agent to accept within the registration burst rate window. If this threshold is exceeded, the Oracle Communications Session Border Controller will respond to new registration requests with 503 Service Unavailable messages. 0 is the minimum and default value for this parameter; the maximum value is 999999999.
9. Save and activate your configuration.

## DDoS Protection from Devices Behind a NAT

A DDoS attack could be crafted such that multiple devices from behind a single NAT could overwhelm the Oracle Communications Session Border Controller. The Oracle Communications Session Border Controller would not detect this as a DDoS attack because each endpoint would have the same source IP but multiple source ports. Because the Oracle Communications Session Border Controller allocates a different CAM entry for each source IP:Port combination, this attack will not be detected. This feature remedies such a possibility.

### Restricting the Number of Endpoints behind a NAT

Each new source IP address and source IP port combination now counts as an endpoint for a particular NAT device. After the configured value of a single NAT's endpoints is reached, subsequent messages from behind that NAT are dropped and the NAT is demoted. This is set with the `max-endpoints-per-nat` parameter located in both the `access-control` and `realm-config` configuration elements.

### Counting Invalid Messages from Endpoints behind a NAT

The Oracle Communications Session Border Controller also counts the number of invalid messages sent from endpoints behind the NAT. Once a threshold is reached, that NAT is demoted. Numerous conditions are counted as Errors/Invalid Messages from an endpoint. The aggregate of all messages from endpoints behind the NAT are counted against the NAT device, in addition to the existing count against the endpoint. This threshold is set with the `nat-invalid-message-threshold` parameter located in both the `access-control` and `realm-config` configuration elements.

As a unique case, the absence of a REGISTER message following a 401 response is counted as an invalid message from the end point. And if that endpoint is behind a NAT, this scenario will be counted as invalid message from that NAT device as well. You set a timeout period in which the REGISTER message must arrive at the Oracle Communications Session Border Controller. This period is set with the `wait-time-for-invalid-register` parameter located in the `realm config`.

### DDoS Protection Configuration realm-config

To set the DDoS Protection from devices behind NATs in the `realm-config`:

1. Access the **realm-config** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

2. Select the **realm-config** object to edit.

```
ACMEPACKET(realm-config)# select
identifier:
1: realm01 left-left:0 0.0.0.0

selection: 1
ACMEPACKET(realm-config)#
```

3. **max-endpoints-per-nat**— Set the maximum number of endpoints that can exist behind a NAT before demoting the NAT device. Valid values are 0-65535, with 0 disabling this feature. This parameter is also found in the access control configuration element.
4. **nat-invalid-message-threshold**—Set the maximum number of invalid messages that may originate behind a NAT before demoting the NAT device. Valid values are 0-65535, with 0 disabling this feature. This parameter is also found in the access control configuration element.
5. **wait-time-for-invalid-register**—Set the period (in seconds) that the Oracle Communications Session Border Controller counts before considering the absence of the REGISTER message as an invalid message.
6. Type **done** to save your configuration.

### DDoS Protection Configuration access-control

To set the DDoS Protection from devices behind NATs in the access-control configuration element:

1. Access the access-control configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# access-control
ACMEPACKET(access-control)#
```

2. Type select to choose and configure an existing object.

```
ACMEPACKET(access-control)# select
<src-ip>:
1: src 0.0.0.0; 0.0.0.0; realm01; ; ALL
selection:1
```

3. **max-endpoints-per-nat**— Set the maximum number of endpoints that can exist behind a NAT before demoting the NAT device. Valid values are 0-65535, with 0 disabling this feature. This parameter is also found in the access control configuration element.
4. **nat-invalid-message-threshold**—Set the maximum number of invalid messages that may originate behind a NAT before demoting the NAT device. Valid values are 0-65535, with 0 disabling this feature. This parameter is also found in the access control configuration element.
5. Type **done** to save your work and continue.

### SNMP Trap support

The following trap is sent by the Oracle Communications Session Border Controller whenever a NAT device is blacklisted due to the triggers listed in this feature. Reasons are not included in the trap, but are available from the syslogs.

```
apSysMgmtExpDOSTrap      NOTIFICATION-TYPE
  OBJECTS                 { apSysMgmtDOSIpAddress,  apSysMgmtDOSRealmID ,
                          apSysMgmtDOSFromUri }
  STATUS                  deprecated
  DESCRIPTION
    "This trap is generated when an IP is placed on a deny list due
    to denial-of-service attempts, and provides the ip address that
    has been demoted, the realm-id of that IP, and (if available)
    the URI portion of the SIP From header of the message that
    caused the demotion."
  ::= { apSysMgmtDOSNotifications 2 }
```

Ensure that the `enable-snmp-monitor-traps` parameter in the `system-config` configuration element is enabled for the Oracle Communications Session Border Controller to send out this trap.

### Syslog Support

Set the `syslog-on-demote-to-deny` parameter in the `media-manager-config` to enabled to generate syslog on endpoint demotion from untrusted to deny. NAT device demotion will also generate a unique syslog message with accompanying text explaining that it is the NAT device demotion event.

## Debugging

The show sip acl command now includes counts of Blocked NAT devices.

```
ACMEPACKET# show sipd acl
13:57:28-71
SIP ACL Status          -- Period -- ----- Lifetime -----
                        Active   High   Total   Total PerMax   High
Total Entries           0      0      0      0      0      0
Trusted                 0      0      0      0      0      0
Blocked                 0      0      0      0      0      0
Blocked NATs            0      0      0      0      0      0
ACL Operations          ---- Lifetime ----
                        Recent   Total   PerMax
ACL Requests            0      0      0
Bad Messages            0      0      0
Promotions              0      0      0
Demotions               0      0      0
Trust->Untrust          0      0      0
Untrust->Deny           0      0      0
```

## Media Policing

Media policing controls the throughput of individual media flows in the Oracle Communications Session Border Controller, which in turn provides security and bandwidth management functionality. The media policing feature works for SIP, H.323 and SIP-H.323 protocols. The media policing feature also lets you police static flows and RTCP flows.

The term media policing refers to flows that go through the Oracle Communications Session Border Controller. Flows that are directed to the host application are not affected by media policing.

You can use media policing to protect against two potential security threats that can be directed against your Oracle Communications Session Border Controller:

- **Media DoS**—Once media flows are established through the Oracle Communications Session Border Controller, network resources are open to RTP media flooding. You can eliminate the threat of a media DoS attack by constraining media flows to absolute bandwidth thresholds.
- **Bandwidth Piracy**—Bandwidth policing ensures that sessions consume no more bandwidth than what is signaled for.

## Policing Methods

The Oracle Communications Session Border Controller polices real-time traffic by using Constant Bit Rate (CBR) media policing. CBR policing is used when a media flow requires a static amount of bandwidth to be available during its lifetime. CBR policing best supports real-time applications that have tightly constrained delay variation. For example, voice and video streaming are prime candidates for CBR policing.

### Session Media Flow Policing

Session media encompasses RTP and RTCP flows. In order to select policing constraints for these flows, the Oracle Communications Session Border Controller watches for the codec specified in an SDP or H.245 message. When a match is made between the codec listed in an incoming session request and a configured **media-profile** configuration element, the Oracle Communications Session Border Controller applies that **media-profile's** bandwidth policing constraint to the media flow about to start.

If multiple codecs are listed in the SDP message, the Oracle Communications Session Border Controller will use the **media-profile** with the most permissive media policing constraints for all of the flows associated with the session. If a codec in the H.245/SDP message is not found in any configured **media-profile**, the Oracle Communications Session Border Controller uses the **media-profile** with the most permissive media policing constraints configured. If no **media-profiles** are configured, there will be no session media flow policing.

If a mid-call change occurs, bandwidth policing is renegotiated.

### Static Flow Policing

Static flows can also be policed in the same way as media flows are policed. A static flow configuration redirects flows entering the Oracle Communications Session Border Controller on a media interface. The redirection is based on realm, source, destination, and protocol. When a flow matches the configured static flow criteria, besides being redirected toward a specified destination, its rate can also be controlled based on a static flow policing parameter found in the **static-flow** element. Static flow policing operates obliviously to the data contained within the flow.

## Configuration Notes

Review the following information before configuring your Oracle Communications Session Border Controller to perform media policing.

### Session Media Flow Policing

Session media flow policing applies to both RTP and RTCP flows. Setting either of the parameters listed below to 0 disables media policing, letting RTP or RTCP flows pass through the system unrestricted.

- RTP Policing
  - Set in the **media-profile** configuration element's **average-rate-limit** parameter to police RTP traffic with the CBR policing method.
  - **average-rate-limit**—Establishes the maximum speed for a flow in bytes per second.
- RTCP Policing
  - Set in the **media-manager-config** configuration element's **rtcp-rate-limit** parameter to police RTCP traffic with the CBR policing method.
  - **rtcp-rate-limit**—Establishes the maximum speed for an RTCP flow in bytes per second.

### Static Flow Policing

Static flow policing is configured with one parameter found in the **static-flow** configuration element. To configure CBR, you have to set the **average-rate-limit** parameter to a non-zero value. Setting the parameter listed below to 0 disables static flow policing, effectively letting the flow pass through the Oracle Communications Session Border Controller unrestricted.

In a CBR configuration, the **average-rate-limit** parameter determines the maximum bandwidth available to the flow.

- **average-rate-limit**—Establishes the maximum speed for a static flow in bytes per second.



**Note:** Static flow policing is not necessarily tied to any type of media traffic, it can affect flows of any traffic type.

## Media Policing Configuration for RTP Flows

You can configure media policing in the **media-profile** configuration element using the ACLI. In the following example, you will configure media policing for the G723 media profile.

To configure media policing for RTP flows:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the **session-router** path.

```
ACMEPACKET (configure) # session-router
```

3. Type **media-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # media-profile
```



4. Select an existing media profile to which you will add policing constraints.

```
ACMEPACKET(media-profile)# select
<name>:
1: audio 4=G723 RTP/AVP 16 0 0 0
selection:1
ACMEPACKET(media-profile)#
```

From this point, you can configure media policing parameters. To view all **media-profile** parameters, enter a **?** at the system prompt

5. **average-rate-limit**—Enter the maximum rate in bytes per second for any flows that this **media-profile** polices. The default value is zero (0), disabling media policing. The valid range is:

- Minimum—0
- Maximum—125000000

Average rate limit values for common codecs:

- PCMU—80000 Bps
- G729—26000 Bps

The following example shows a **media-profile** configuration element configured for media policing.

```
media-profile
  name G723
  media-type audio
  payload-type 4
  transport RTP/AVP
  req-bandwidth 16
  frames-per-packet 0
  parameters
  average-rate-limit 15000
```

### Media Policing Configuration for RTCP Flows

You can configure media policing for RTCP flows by using the ACLI.

To configure media policing for RTCP flows:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the **media-manager** path.

```
ACMEPACKET(configure)# media-manager
```

3. Type **media-manager** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# media-manager
ACMEPACKET(media-manager-config)#
```

4. **rtcp-rate-limit**—Enter the RTCP policing constraint in bytes per second. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—125000000

### Media Policing Configuration for Static Flows

You can configure media policing for static flows using the ACLI.

To configure media policing for static flows:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

## Security

2. Type **media-manager** and press Enter to access the **media-manager** path.

```
ACMEPACKET (configure) # media-manager
```

3. Type **static-flow** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # static-flow  
ACMEPACKET (static-flow) #
```

4. Select an existing static flow to which you will add policing constraints.

```
ACMEPACKET (static-flow) # select  
<in-dest-ip>:  
1: dest 0.0.0.0; src 192.168.2.1/24; static-flow-in-realm; UDP  
selection:1
```

From this point, you can configure media policing parameters for static flows. To view all **static-flow** parameters, enter a **?** at the system prompt

5. **average-rate-limit**—Enter the maximum rate in bytes per second for any flows that this **static-flow** polices. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—125000000

The following example shows a **static-flow** configuration element configured for media policing.

```
static-flow  
  in-realm-id          static-flow-in-realm  
  in-source            192.168.2.1/24  
  in-destination       0.0.0.0  
  out-realm-id         static-flow-out-realm  
  out-source           192.168.128.1/24  
  out-destination     0.0.0.0  
protocol              UDP  
  average-rate-limit   15000
```

## RTP Payload Type Mapping

The Oracle Communications Session Border Controller maintains a default list of RTP payload types mapped to textual encoding names as defined in RFC 3551.

The following table defines the preconfigured payload type for standard encodings.

Payload Type	Encoding Name	Audio (A) / Video (V)	Clock Rate
0	PCMU	A	8000
4	G723	A	8000
8	PCMA	A	8000
9	G722	A	8000
15	G728	A	8000
18	G729	A	8000

If you configure any payload type to encoding name mappings, the default mappings will be ignored. You must then manually enter all payload type mappings you use in the **media-profile** configuration element.

## ITU-T to IANA Codec Mapping

The Oracle Communications Session Border Controller maintains a list of ITU-T (H.245) codecs that map to IANA RTP codecs. An ITU codec is directly mapped to an IANA Encoding Name for media profile lookups. All codecs are

normalized to IANA codec names before any matches are made. New ITU-T codecs can not be added to the media profiles list.

The following table defines the ITU-T to IANA codec mappings.

ITU-T	IANA
g711Ulaw64k	PCMU
g711Alaw64k	PCMA
g726	G726
G7231	G723
g728	G728
g729wAnnexB	G729
g729	G729 fmp:18 annexb=no
H261VideoCapability	H261
H263VideoCapability	H263
t38Fax	T38

## SDP Anonymization

In order to provide an added measure of security, the Oracle Communications Session Border Controller's topology-hiding capabilities include SDP anonymization. Enabling this feature gives the Oracle Communications Session Border Controller the ability to change or modify certain values in the SDP so that malicious parties will be unable to learn information about your network topology.

To do this, the Oracle Communications Session Border Controller hides the product-specific information that can appear in SDP `o=` lines and `s=` lines. This information can include usernames, session names, and version fields. To resolve this issues, the Oracle Communications Session Border Controller makes the following changes when you enable SDP anonymization:

- Sets the session name (or the `s=` line in the SDP) to `s=-`
- Sets the username in the origin field to `-SBC`
- Sets the session ID in the origin field to an integer of incrementing value

Note that for mid-call media changes, the session identifier is not incremented.

To enable this feature, you set a parameter in the media manager configuration.

## SDP Anonymization Configuration

To enable SDP anonymization:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `media-manager` and press Enter.

```
ACMEPACKET(configure)# media-manager
```

3. Type `media-manager` again to access the media manager configuration, and press Enter.

```
ACMEPACKET(media-manager)# media-manager
ACMEPACKET(media-manager-config)#
```

4. `anonymous-sdp`—Set this parameter to `enabled` to use the SDP anonymization feature. When you leave this parameter empty the feature is turned off. The default value is `disabled`. The valid values are:

- `enabled` | `disabled`

5. Save and activate your configuration.

### Unique SDP Session ID

Codec negotiation can be enabled by updating the SDP session ID and version number. The `media-manager` option, `unique-sdp-id` enables this feature.

With this option enabled, the Oracle Communications Session Border Controller will hash the session ID and IP address of the incoming SDP with the current date/time of the Oracle Communications Session Border Controller in order to generate a unique session ID.

### Unique SDP Session ID Configuration

To enable unique SDP session ID in `media-manager`:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type `media-manager` and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. **options**—Set the options parameter by typing `options`, a Space, the option name `unique-sdp-id` with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(media-manager)# options +unique-sdp-id
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the realm configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

4. Save and activate your configuration.

## TCP Synchronize Attack Prevention

---

This section explains how the Oracle Communications Session Border Controller protects itself from a Transmission Control Protocol (TCP) synchronize (SYN) packet flooding attack sourced from a remote hostile entity.

SIP and H.323 signaling can be configured on the Oracle Communications Session Border Controller to be TCP protocol-based. In this configuration, the Oracle Communications Session Border Controller can be a target of a TCP SYN attack. The Oracle Communications Session Border Controller C is able to service new call requests throughout the duration of an attack

### About SYN

SYN is used by TCP when initiating a new connection to synchronize the sequence numbers on two connecting computers. The SYN is acknowledged by a SYN-ACK by the responding computer. After the SYN-ACK, the client finishes establishing the connection by responding with an ACK message. The connection between the client and the server is then open, and the service-specific data can be exchanged between the client and the server.

A SYN flood is a series of SYN packets from forged IP addresses. The IP addresses are chosen randomly and do not provide any hint of the attacker's location. The SYN flood keeps the server's SYN queue full. Normally this would force the server to drop connections. A server that uses SYN cookies, however, will continue operating normally. The biggest effect of the SYN flood is to disable large windows.

#### Server Vulnerability

Vulnerability to attack occurs when the server has sent a SYN-ACK back to client, but has not yet received the ACK message; which is considered a half-open connection. The server has a data structure describing all pending

connections built in its system memory. This data structure is of finite size, and it can be made to overflow by intentionally creating too many partially-open connections.

The attacking system sends SYN messages to the server that appear to be legitimate, but in fact reference a client that is unable to respond to the SYN-ACK messages. The final ACK message is never sent to the server.

The half-open connections data structure on the server fills and no new incoming connections are accepted until the table is emptied out. Typically there is a timeout associated with a pending connection (the half-open connections will eventually expire and the server will recover). But the attacking system can continue sending IP-spoofed packets requesting new connections faster than the server can expire the pending connections. The server has difficulty in accepting any new incoming network connections.

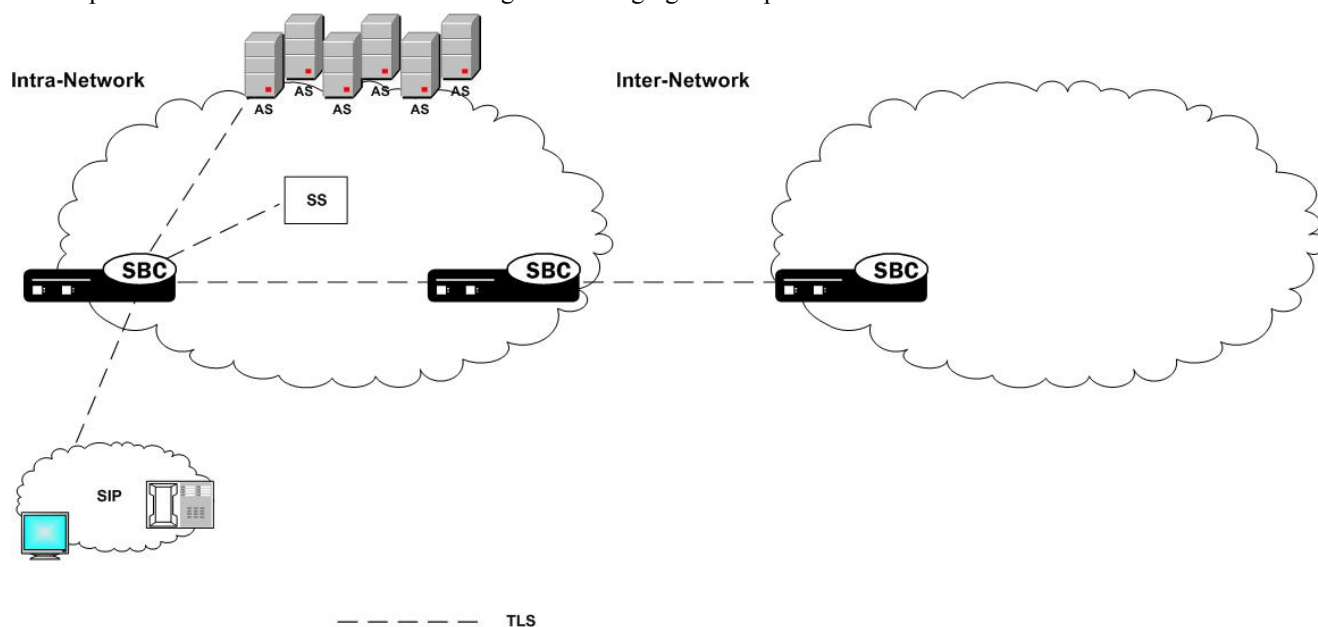
## Configuring TCP SYN Attack Prevention

No configuration is necessary to enable TCP SYN attack prevention. Internal TCP protocol changes were made to provide protection.

## Transport Layer Security

The Oracle Communications Session Border Controller provides support for Transport Layer Security (TLS) for SIP, which can be used to protect user and network privacy by providing authentication and guaranteeing the integrity for communications between the Oracle Communications Session Border Controller and the following:

- Another device in your network infrastructure (intra-network)
- Another Oracle Communications Session Border Controller when you are using a peering application (inter-network) for interior network signaling security
- An endpoint for authentication before allowing SIP messaging to take place
- 



## The Oracle Communications Session Border Controller and TLS

The Oracle Communications Session Border Controller's TLS functionality depends on the presence of the Security Service Module (SSM) for hardware acceleration of encryption and decryption and random media generation. The SSM is a plug-on module that can be added to your Oracle Communications Session Border Controller chassis given the installation of the necessary bootloader and minimum hardware revision levels.

With the requisite hardware revision levels, the plug-on unit can be added to your Oracle Communications Session Border Controller in the field by qualified personnel. This provision makes upgrades fast, forgoing the need for you to return your Oracle Communications Session Border Controller to Oracle manufacturing for hardware upgrade. When

your Oracle Communications Session Border Controller is upgraded with the SSM card that supports TLS, a new CLEI code will be added to your chassis; the code will also appear on the SSM card (also referred to as the plug-on unit) and visible if the system's chassis cover is opened. New Oracle Communications Session Border Controller s outfitted with the SSM card will have the code labels already affixed in all required locations.

TLS support will not behave in the manner described here if you do not have the SSM component installed on your Oracle Communications Session Border Controller , because it is the presence of this hardware that enables the TLS software support.

The accelerator card performs:

- RSA
- Diffie-Hellman
- DES
- 3DES
- 40/128 bit ARCFOUR
- AES256
- Random number generation

## TLS Features

The Oracle Communications Session Border Controller supports the following TLS features:

- TLSv1/SSLv3
- RFC 3261 specific SIPS and TLS support in SIP
- Importing X509v3 certificates in PKCS-7/X509v3 PEM/Base64 format
- Generating a private key and a certificate request in PKCS-10 PEM/Base64 format
- Displaying imported certificates in text format
- Configuration verification, including verification that all dependencies are resolved
- Connection reuse draft (draft-ietf-sip-connect-reuse-03.txt)
- S-CX6.3.0 allows a maximum of 60000 TLS and 60000 TCP connections where each TLS connection also consumes one TCP connection.
- As of S-CX6.3.0M3, these capacities are raised to 100000 TLS connections and 120000 TCP connections.
- HA for TLS—When the active system in an HA node fails, the standby has the same TLS-related configuration, which is accomplished through configuration checkpointing as described in the HA Nodes chapter.

Existing active calls are not affected by a failover—Enduser experiences no interruption or disturbance in service. SIP signaling messages sent over the connection following failover do not impact the active call.

New calls, new TLS connections are made


The Oracle Communications Session Border Controller does not support certificate revocation listing handling.

## Domestic and International Versions

There are two versions of the Acme Packet OS that support TLS: a U.S. version and an international version. Two versions exist because of the laws governing the strength of algorithms that can be shipped domestically and internationally. If you require further information, consult with your ACMEPACKET sales representative directly.

## Supported Encryption

The Oracle Communications Session Border Controller provides support for TLSv1 and SSLv3 encryption.

 **Note:** We do not support RC4 ciphers on the Net-Net 3800 or the Net-Net 4500. We do continue to support RC4 ciphers on the Net-Net 4250 for backwards-compatibility purposes.

## TLS Ciphers

The Oracle Communications Session Border Controller (SBC) supports TLS v1, TLSv1.1, TLSv1.2, and SSLv3 ciphers.


For encryption, the SBC supports: AES-128, AES-256, 3DES, DES and ARC4 (40 and 128 bit) algorithms. It also supports:

- TLS\_DHE\_RSA\_WITH\_AES\_256\_GCM\_SHA384
- TLS\_DHE\_RSA\_WITH\_AES\_256\_CBC\_SHA256
- TLS\_DHE\_RSA\_WITH\_AES\_256\_CBC\_SHA
- TLS\_RSA\_WITH\_AES\_256\_GCM\_SHA384
- TLS\_RSA\_WITH\_AES\_256\_CBC\_SHA256
- TLS\_RSA\_WITH\_AES\_256\_CBC\_SHA
- TLS\_DHE\_RSA\_WITH\_AES\_128\_GCM\_SHA256
- TLS\_DHE\_RSA\_WITH\_AES\_128\_CBC\_SHA256
- TLS\_DHE\_RSA\_WITH\_AES\_128\_CBC\_SHA
- TLS\_RSA\_WITH\_AES\_128\_GCM\_SHA256
- TLS\_RSA\_WITH\_AES\_128\_CBC\_SHA256
- TLS\_RSA\_WITH\_AES\_128\_CBC\_SHA
- TLS\_DHE\_RSA\_WITH\_3DES\_EDE\_CBC\_SHA
- TLS\_DHE\_RSA\_WITH\_DES\_CBC\_SHA
- TLS\_RSA\_WITH\_3DES\_EDE\_CBC\_SHA
- TLS\_RSA\_WITH\_DES\_CBC\_SHA
- TLS\_RSA\_EXPORT1024\_WITH\_DES\_CBC\_SHA
- TLS\_RSA\_WITH\_NULL\_SHA256
- TLS\_RSA\_WITH\_NULL\_SHA
- TLS\_RSA\_WITH\_NULL\_MD5
- TLS\_ECDHE\_ECDSA\_WITH\_AES\_128\_GCM\_SHA256
- TLS\_ECDHE\_ECDSA\_WITH\_AES\_128\_GCM\_SHA256
- ALL [default]
- NONE

### Mapping SSL3 to TLSv1 Ciphers

The following table shows the mapping of SSL3 ciphers to TLSv1 ciphers:

SSL3	TLSv1
SSL_RSA_WITH_NULL_MD5	TLS_RSA_WITH_NULL_MD5
SSL_RSA_WITH_NULL_SHA	TLS_RSA_WITH_NULL_SHA
SSL_RSA_WITH_RC4_128_MD5	TLS_RSA_WITH_RC4_128_MD5
SSL_RSA_WITH_RC4_128_SHA	TLS_RSA_WITH_RC4_128_SHA
SSL_RSA_WITH_DES_CBC_SHA	TLS_RSA_WITH_DES_CBC_SHA
SSL_RSA_WITH_3DES_EDE_CBC_SHA	TLS_RSA_WITH_3DES_EDE_CBC_SHA
SSL_DHE_RSA_WITH_DES_CBC_SHA	TLS_DHE_RSA_WITH_DES_CBC_SHA
SSL_DHE_RSA_WITH_3DES_EDE_CBC_SHA	TLS_DHE_RSA_WITH_3DES_EDE_CBC_SHA

 **Note:** The Oracle Communications Session Border Controller supports TLS\_RSA\_WITH\_NULL\_MD5 and TLS\_RSA\_WITH\_NULL\_SHA although neither does any encryption. These ciphers might be used for debugging purposes, and should not be deployed.

### TLS 1.1 and 1.2 Support

The Oracle Communications Session Border Controller supports TLS 1.1 and 1.2. In order to support TLS version 1.2, additional cipher suites have been added.

## Security

New values have been added to the **tls-version** parameter in the **tls-profile** configuration element to enable TLS v1.1 or 1.2. These values are:

- **tls11**—TLS Version 1.1
- **tls12**—TLS Version 1.2

Additionally, setting **tls-version** to **compatibility** now encompasses all supported versions, i.e. TLS 1.0, 1.1, 1.2, and SSLv3.

The following list contains new ciphers that have been added to the **cipher list** parameter to support TLS version 1.2:

- TLS\_RSA\_WITH\_NULL\_SHA256
- TLS\_RSA\_WITH\_AES\_128\_CBC\_SHA256
- TLS\_RSA\_WITH\_AES\_256\_CBC\_SHA256
- TLS\_DHE\_RSA\_WITH\_AES\_128\_CBC\_SHA256
- TLS\_DHE\_RSA\_WITH\_AES\_256\_CBC\_SHA256
- TLS\_RSA\_WITH\_AES\_128\_GCM\_SHA256
- TLS\_RSA\_WITH\_AES\_256\_GCM\_SHA384
- TLS\_DHE\_RSA\_WITH\_AES\_128\_GCM\_SHA256
- TLS\_DHE\_RSA\_WITH\_AES\_256\_GCM\_SHA384

### TLS 1.1 and 1.2 Hardware and License requirements

Refer to the following table for hardware requirements for TLS 1.1 and 1.2.

Platform	Hardware and License Prerequisites
Acme Packet 3820	Software TLS License, TLS License
Acme Packet 4500	SSMv2, TLS License
Acme Packet 4600	SSMv3, TLS License
Acme Packet 6100	SSMv3, TLS License
Acme Packet 6300	SSMv3, TLS License
Server Edition (Session Router on COTS)	Software TLS License, TLS License

You can use the **show security ssm-accelerator** command to display the SSM hardware present in your system.

### 4096 bit RSA Key Support

The Oracle Communications Session Border Controller supports 4096 bit RSA keys for SIP TLS on AP platforms with SSMv3 hardware. Systems supporting the SSMv3 include the AP6100 and AP6300.

To accommodate 4096 bit RSA keys, the **security > certificate record > key-size** parameter now includes the 4096 value.

```
ACMEPACKET(certificate-record)# key-size <TAB>
1024          2048          4096          512
```

Systems that support SSMv2 hardware (AP3820 and AP4500) will not display the option to use 4096 bit RSA keys in the **security > certificate record > key-size** parameter.

### Signaling Support

The Oracle Communications Session Border Controller's TLS functionality supports SIP and SIPS. In addition, the Oracle Communications Session Border Controller can accommodate a mixture of TLS and non-TLS sessions within a realm as because a request for TLS is controlled by the endpoint (TLS UA).



## DoS Protection

The Oracle Communications Session Border Controller provides the following forms of DoS protection from:

- Too many simultaneous TLS connections being requested by a single IP address.

The Oracle Communications Session Border Controller limits the number of TLS connections from a single IP address; you can set a maximum simultaneous number of TCP/TLS connections a SIP interface will allow from a single IP address.

- Too many simultaneous TLS connections being requested by limiting the maximum number of connections for a SIP interface.

In other words, the maximum simultaneous TCP/TLS connections a SIP interface will allow in aggregate from all IP addresses served by that signaling interface.

- Endpoints establishing TCP/TLS connections that never send any messages (application layer messages; once the TLS handshake completes).

This protection is triggered by inactivity, measured by lack of any message from a peer. The value specified for this timer is in seconds.

- Endpoints requesting an initial registration that never send messages thereafter.



**Note:** It is expected that whenever an endpoint establishes a TCP/TLS connection, it will keep the connection active by sending additional messages or by using the NAT interval configuration. Whenever a connection is torn down because of inactivity, a log at the level "ERROR" is generated.

- Malformed packets by counting and limiting the maximum number of malformed packets.

Whenever the Oracle Communications Session Border Controller receives an invalid TLS message, it increments the internal invalid signalling threshold counter. When that counter reaches the configured value, the Oracle Communications Session Border Controller denies the endpoints for the configured deny period. This also requires configuration of tolerance window in media manager.

## Endpoint Authentication

The Oracle Communications Session Border Controller does not operate as a CA. Instead, the Oracle Communications Session Border Controller's TLS implementation assumes that you are using one of the standard CAs for generating certificates:

- Verisign
- Entrust
- Thawte
- free Linux-based CA (for example, openssl)



**Note:** Self-signed certificates are available only as an option for MSRP connections

The Oracle Communications Session Border Controller can generate a certificate request in PKCS10 format and to export it. It can also import CA certificates and a Oracle Communications Session Border Controller certificate in the PKCS7/X509 PEM format.

The Oracle Communications Session Border Controller generates the key pair for the certificate request internally. The private key is stored as a part of the configuration in 3DES encrypted form (with an internal generated password) and the public key is returned to the user along with other information as a part of PKCS10 certificate request.

The Oracle Communications Session Border Controller supports the option of importing CA certificates and marking them as trusted. However, the Oracle Communications Session Border Controller only authenticates client certificates that are issued by the CAs belonging to its trusted list. If you install only a specific vendor's CA certificate on the Oracle Communications Session Border Controller, it authenticates that vendor's endpoints. Whether the certificate is an individual device certificate or a site-to-site certificate does not matter because the Oracle Communications Session Border Controller authenticates the signature/public key of the certificate.

### Key Usage Control

You can configure the role of a certificate by setting key usage extensions and extended key usage extensions. Both of these are configured in the certificate record configuration.

#### Key Usage List

This section defines the values you can use (as a list) in the **key-usage-list** parameter. You can configure the parameter with more than one of the possible values.

Value	Description
digitalSignature (default with keyEncipherment)	Used when the subject public key is used with a digital signature mechanism to support security services other than non-repudiation, certificate signing, or revocation information signing. Digital signature mechanisms are often used for entity authentication and data origin authentication with integrity.
nonRepudiation	Used when the subject public key is used to verify digital signatures that provide a non-repudiation service protecting against the signing entity falsely denying some action, excluding certificate or CRL signing.
keyEncipherment (default with digitalSignature)	Used with the subject public key is used for key transport. (For example, when an RSA key is to be used for key management.)
dataEncipherment	Used with the subject public key is used for enciphering user data other than cryptographic keys.
keyAgreement	Used with the subject public key is used key agreement. (For example, when a Diffie-Hellman key is to be used for a management key.)
encipherOnly	The keyAgreement type must also be set. Used with the subject public key is used only for enciphering data while performing key agreement.
decipherOnly	The keyAgreement type must also be set. Used with the subject public key is used only for deciphering data while performing key agreement.

#### Extended Key Usage List

This section defines the values you can use in the **extended-key-usage-list** parameter.

Value	Description
serverAuth (default)	Used while the certificate is used for TLS server authentication. In Oracle Communications Session Border Controller access-side deployments, the system typically acts as a TLS server accepting TLS connections. You might use this setting while generating the end-entity-cert.
clientAuth	Used while the certificate is used for TLS client authentication. In Oracle Communications Session Border Controller core-side deployments, the system typically acts as a TLS client initiating TLS connections. You might use this setting while generating the end-entity-cert.

### Configuring TLS

This section explains how to configure your Oracle Communications Session Border Controller for TLS support.

## Process Overview

In summary, you need to take the following steps to enable your Oracle Communications Session Border Controller for TLS.

1. Make sure that your Oracle Communications Session Border Controller has the appropriate hardware installed and that you have obtained an enabled the licenses related to TLS support. (Note that the Acme Packet 4250 does not require an additional license for TLS support.)
2. Configure certificates.
3. Configure the specific parameters related to TLS.

## Configuring Certificates


Configuring certificates is a three-step process:

1. Create a certificate record configuration on the Oracle Communications Session Border Controller
2. Generate a certificate request by the Oracle Communications Session Border Controller and save the configuration
3. Import the certificate record into the Oracle Communications Session Border Controller and save the configuration

### Configuring the Certificate Record

The certificate record configuration represents either the end-entity or the CA certificate on the Oracle Communications Session Border Controller. If it is used to present an end-entity certificate, a private key should be associated with this certificate record configuration using the ACLI **generate-certificate-request** command. A certificate, provided by a CA in response to a certificate request, can be imported to a certificate record configuration using the ACLI **import-certificate** command.

No private key should be associated with the certificate record configuration if it was issued to hold a CA certificate.

 **Note:** There is no need to create a certificate record when importing a CA certificate or certificate in pkcs12 format.

The following is sample of the certificate record configuration parameters as seen in the ACLI.

```
certificate-record
name          certificate record name
country       country name
state         state name
locality      locality name
organization  organization name
unit          organization unit
common-name   common name
key-size      key size
alternate-name alternate name
trusted       certificate-record trusted or not
```

To enter a certificate record using the ACLI configuration menu:

1. Access the **certificate-record** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# certificate record
ACMEPACKET(certificate-record)#
```

2. **name**—Provide the required name of this certificate record object.
3. **country, state, locality, organization, unit, and common-name**—Use these parameters to identify the certificate's subject.

```
ACMEPACKET(certificate-record)# country us
ACMEPACKET(certificate-record)# state ca
ACMEPACKET(certificate-record)# locality "San Francisco"
ACMEPACKET(certificate-record)# organization "Office of the CTO"
```

```
ACMEPACKET(certificate-record)# unit cyzygy.com
ACMEPACKET(certificate-record)# common-name www.cyzygy.org/
emailAddress=cto@cyzygy.org
```

Based on this ACLI sequence, the subject field of the resulting CA-issued certificate reads as follows.

```
Subject: C=US, ST=California, L=San Francisco, O=Office of the CTO
OU=Cyzygy,
CN=www.cyzygy.org/emailAddress=cto@cyzygy.org
```

4. **key-size**—Enter the size of the key for the certificate. The default value is **1024**. The valid range is:

- 512 | 1024 (default) | 2048

5. **alternate-name**—Optionally provide one or more alternative names for the certificate holder. The Subject Alternative Name certificate extension is defined in section 4.2.1.6 of RFC 5280, *Internet X.509 Public Key Infrastructure Certificate and Certificate Revocation List (CRL) Profile*. This extension allows one or more identities to be bound to the subject of a certificate. As a result, each subject alternative name is considered as an alias for the certificate subject. These aliases may be included in addition to, or in place of the identity in the subject field of the certificate, meaning that a certificate can contain a subject name and one or more aliases, or no subject name and one or more aliases. A certificate cannot lack both a subject name and an alias. As specifically defined in RFC 5280, subject alternative names can take the form of an e-mail address, an IP address (IPv4 or IPv6), a DNS address, a Registered ID (RID), a Distinguished Name (DN), or a URI.

RFC also 5280 allows future support for other alternative name forms, but only if such forms are specified in an IETF RFC.

Most alternative names take one of the following three formats: IP address (which requires an **IP: prefix**), DNS (which requires a **DNS: prefix**), or email (which requires an **email: prefix**).

Example:

```
CMEPACKET(certificate-record)# alternate-name IP:192.168.12.101
ACMEPACKET(certificate-record)#

ACMEPACKET(certificate-record)# alternate-name IP:13::17
ACMEPACKET(certificate-record)#

ACMEPACKET(certificate-record)# alternate-name DNS:sgla.ba.de
ACMEPACKET(certificate-record)#

ACMEPACKET(certificate-record)# alternate-name email:my@other.address
ACMEPACKET(certificate-record)#

ACMEPACKET(certificate-record)# alternate-name copy,email:my@other.address
ACMEPACKET(certificate-record)#
```

The email form can include a special copy value that includes any email addresses contained in the certificate subject name in the extension.

For other formats, consult RFC 5280, later RFCs that define future formats, or the OpenSSL documentation available at: [http://www.openssl.org/docs/apps/x509v3\\_config.html](http://www.openssl.org/docs/apps/x509v3_config.html)

6. **trusted**—Leave this parameters set to **enabled** to make the certificate trusted. Enter **disabled** to make this certificate untrusted. The default value is **enabled**. The valid values are:

- enabled | disabled

7. **key-usage-list**—Enter the usage extensions you want to use with this certificate record. This parameter can be configured with multiple values, and it defaults to the combination of **digitalSignature** and **keyEncipherment**. For a list of possible values and their descriptions, see the section “Key Usage Control” in the *Oracle Communications Session Border Controller Configuration Guide*.

8. **extended-key-usage-list**—Enter the extended key usage extensions you want to use with this certificate record. The default is **serverAuth**. For a list of possible values and their descriptions, see the section “Key Usage Control” in the *Oracle Communications Session Border Controller Configuration Guide*.

9. Type **done** and press Enter.

```
ACMEPACKET (certificate-record) # done
```

10. Type **done** to save your configuration.

You create TLS profiles, using your certificate records, to further define the encryption behavior and create the configuration element that you can then apply to a SIP interface.

## Generating a Certificate Request

Using the ACLI **generate-certificate-request** command allows you to generate a private key and a certificate request in PKCS10 PEM format. You take this step once you have configured a certificate record.

The Oracle Communications Session Border Controller stores the private key that is generated in the certificate record configuration in 3DES encrypted form with an internally generated password. The PKCS10 request is displayed on the screen in PEM (Base64) form.

You use this command for certificate record configurations that hold end-entity certificates. If you have configured the certificate record to hold a CA certificate, then you do not need to generate a certificate request because the CA publishes its certificate in the public domain. You import a CA certificate by using the ACLI **import-certificate** command.

This command sends information to the CA to generate the certificate, but you cannot have Internet connectivity from the Oracle Communications Session Border Controller to the Internet. You can access the internet through a browser such as Internet Explorer if it is available, or you can save the certificate request to a disk and then submit it to the CA.

To run the applicable command, you must use the value you entered in the name parameter of the certificate record configuration. You run the command from main Superuser mode command line:

```
ACMEPACKET# generate-certificate-request acmepacket
Generating Certificate Signing Request. This can take several minutes...
-----BEGIN CERTIFICATE REQUEST-----
MIIDHzCCAoigAwIBAgIIAhMCUACEAHEwDQYJKoZIhvcNAQEFBQAwcDELMAkGA1UE
BhMCVVMxEzARBgNVBAgTCkNhbkG1mb3JuaWEwETAPBgNVBACTCFNhbiBKb3N1MQ4w
DAYDVQQKEwVzaXBpdDEpMCCGA1UECXMgU21waXQgVGVzZCBZJ0aWZpY2F0ZSBB
dXR0b3JpdHkwHhcnMDUwNDEzMjEzNzQzWhcNMDgwNDEyMjEzNzQzWjBUMQswCQYD
VQQGEwJVUzELMAkGA1UECBMCTUEwEzARBgNVBACTCkJK1cmxpbmd0b24xFDASBgNV
BAoTC0VuZ21uZWVyaW5nMQ0wCwYDVQQDEwRhY211MIGfMA0GCSqGSIb3DQEBAQUA
A4GNADCBiQKBgQCXjIeOyFKAUB3rKkKK/+59LT+r1GuW7Lgc1V6+hFTSr0co+ZsQ
bHFUWAA15qxUUBTLJG13QN5VfG96f7gGAbWayfOS9Uymold3JPCUDoGgb2E7m8iu
vtq7gwjSeKNXAw/y7yWy/c04FmUD2U0pZX0CNIR3Mns5OAxQmq0bNYDhawIDAQAB
o4HdMIHaMBEGA1UdEQQKMAiCBnBrdW1hcjAJBgNVHRMEAjAAMB0GA1UdDgQWBBTG
tpodxa6Kmmn04L3Kg62t8BZJHTCBmgYDVR0jBIGSMIGPgBRrRhcu6pR2JYBUbhNU
2qhJvBShtqF0pHIwcDELMAkGA1UEBhMCVVMxEzARBgNVBAgTCkNhbkG1mb3JuaWEw
ETAPBgNVBACTCFNhbiBKb3N1MQ4wDAYDVQQKEwVzaXBpdDEpMCCGA1UECXMgU21w
aXQgVGVzZCBZJ0aWZpY2F0ZSBBdXR0b3JpdHhMCAQAwDQYJKoZIhvcNAQEFBQAD
gYEAbEs8nUCi+cA2hc/1M49Sith8QmpL81KONApsoC4Em24L+DZwz3uInoWjbjJ
QhefcUfteNYkbuMH7LAK0hnDPvW+St4rQGvK6LJhZj7/yeLXmYWIPUY3Ux4OGVrd
2UgV/B2SOqH9Nf+FQ+mNZOL7EuF4IxSz9/69LuYlXqKsG4=
-----END CERTIFICATE REQUEST-----;
WARNING: Configuration changed, run save-config command.
ACMEPACKET# save-config
Save-config received, processing.
waiting 1200 for request to finish
Request to 'SAVE-CONFIG' has Finished,
Save complete
Currently active and saved configurations do not match!
To sync & activate, run 'activate-config' or 'reboot-activate'
ACMEPACKET# activate-config
Activate-Config received, processing.
waiting 12000 for request to finish
Add LI flows
LiSysClientMgr::handleNotifyReq
```

```
H323 Active Stack Cnt: 0
Request to 'ACTIVATE-CONFIG' has finished
Activate Complete
ACMEPACKET#
```

## Importing a Certificate Using the CLI

For an end-entity certificate, once a certificate is generated using the CLI `generate-certificate-request` command, that request should be submitted to a CA for generation of a certificate in PKCS7 or X509v3 format. When the certificate has been generated, it can be imported into the Oracle Communications Session Border Controller using the `import-certificate` command.

The syntax is:

```
ACMEPACKET # import-certificate [try-all|pkcs7|x509] [certificate-record file-name]
```

To import a certificate:

1. When you use the `import-certificate` command, you can specify whether you want to use PKCS7 or X509v3 format, or try all. In the command line, you enter the command, the format specification, and the name of the certificate record.

```
ACMEPACKET# import-certificate try-all acme
```

The following will appear:

```
Please enter the certificate in the PEM format.
Terminate the certificate with ";" to exit.....
-----BEGIN CERTIFICATE-----
MIIDHzCCAoigAwIBAgIIAhMCUACEAHEwDQYJKoZIhvcNAQEFBQAwDELMAkGA1UE
BhMCMCVVMxEzARBgNVBAGTCkNhbGlm3JuaWExETAPBgNVBACTCFNhbiBkb3NlMQ4w
DAYDVQQKEwVzaXBpdDEpMCCGA1UECXMgU2lwaXQgVGVzdCBDZlJ0aWZpY2F0ZSBB
dXR0b3JpdHkwHhcNMDUwNDEzNzEzNzQzWhcNMDgwNDEzNzEzNzQzWjBUMQswCQYD
VQQGEwJVUzELMAkGA1UECBMCTUEEzARBgNVBACTCkJK1cmxpbmd0b24xZDASBgNV
BAoTC0VuZ2luZWVyaW5nMQ0wCwYDVQQDEwRhY21lMIGfMA0GCSqGSIb3DQEBAQUA
A4GNADCBiQKBgQCXjIeOyFKAUB3rKkKK/+59LT+rlGuW7Lgc1V6+hfTSr0co+ZsQ
bHFUWAA15qXUUBTLJG13QN5VfG96f7gGAbWayfOS9Uymold3JPCUDoGgb2E7m8iu
vtq7gwjSeKNXAw/y7yWy/c04FmUD2U0pZX0CNIR3Mns5OAxQmq0bNYDhawIDAQAB
o4HdMIHaMBEGA1UdEQQKMAiCBnBrdW1hcjAJBgNVHRMEAjAAMB0GA1UdDgQWBGTG
tpodxa6Kmmn04L3Kg62t8BZJHTCBmgYDVR0jBIGSMIGPgBRrRhcU6pR2JYBUbNU
2qHjVBShtqF0pHIwcDELMAkGA1UEBhMCMCVVMxEzARBgNVBAGTCkNhbGlm3JuaWEx
ETAPBgNVBACTCFNhbiBkb3NlMQ4wDAYDVQQKEwVzaXBpdDEpMCCGA1UECXMgU2lwa
XQgVGVzdCBDZlJ0aWZpY2F0ZSBBdXR0b3JpdHmCAQAwDQYJKoZIhvcNAQEFBQAD
gYEAbEs8nUCi+cA2hC/lM49Sith8QmpL81KONApsoC4Em24L+DZwz3uInoWjbjJ
QhefcUfteNYkbuMH7LAK0hnDPvW+St4rQGvK6LJhZj7/yeLXmYWIPIUY3Ux4OGVrd
2UgV/B2SOqH9Nf+FQ+mNZOL7EuF4IxSz9/69LuYlXqKsG4=
-----END CERTIFICATE-----;
Certificate imported successfully....
WARNING: Configuration changed, run "save-config" command.
```

2. Save your configuration.

```
ACMEPACKET# save-config
Save-Config received, processing.
waiting 1200 for request to finish
Request to 'SAVE-CONFIG' has Finished,
Save complete
Currently active and saved configurations do not match!
To sync & activate, run 'activate-config' or 'reboot activate'.
```

3. Synchronize and activate your configurations.

```
ACMEPACKET# activate-config
Activate-Config received, processing.
waiting 120000 for request to finish
Add LI Flows
```

```
LiSysClientMgr::handleNotifyReq
H323 Active Stack Cnt: 0
Request to 'ACTIVATE-CONFIG' has Finished,
Activate Complete
ACMEPACKET#
```

## Importing a Certificate Using FTP

You can also put the certificate file in the directory /ramdrv and then executing the **import-certificate** command or by pasting the certificate in the PEM/Base64 format into the ACLI. If you paste the certificate, you might have to copy and paste it a portion at a time rather than pasting in the whole thing at once.

To import the certificate using FTP:

1. FTP the certificate file on to the Oracle Communications Session Border Controller (directory /ramdrv), let us say the name of the certificate file is cert.pem.
2. Once the certificate is successfully transferred to the Oracle Communications Session Border Controller, run the **import-certificate** command.

The syntax is:

```
ACMEPACKET# import-certificate [try-all|pkcs7|x509] [certificate-record file-
name]
```

Using the command will look like this when you have used FTP.

```
ACMEPACKET# import-certificate try-all acme cert.pem
Certificate imported successfully...
WARNING: Configuration changed, run "save-config" command.
```

1. Save your configuration.

```
ACMEPACKET# save-config
Save-Config received, processing.
waiting 1200 for request to finish
Request to 'SAVE-CONFIG' has Finished,
Save complete
Currently active and saved configurations do not match!
To sync & activate, run 'activate-config' or 'reboot activate'.
```

2. Synchronize and activate your configurations.

```
ACMEPACKET# activate-config
Activate-Config received, processing.
waiting 120000 for request to finish
Add LI Flows
LiSysClientMgr::handleNotifyReq
H323 Active Stack Cnt: 0
Request to 'ACTIVATE-CONFIG' has Finished,
Activate Complete
ACMEPACKET#
```

## Configuring a TLS Profile

The TLS profile configuration has been added to the security section of the ACLI's configure terminal menu. This configuration holds the information required to run SIP over TLS.

In the ALCI menu for this configuration, the parameters appear as follows:

```
tls-profile
name                tls profile name
end-entity-certificate  end entity certificate for the TLS connection
trusted-ca-certificates  list of trusted certificate records
cipher-list          list of ciphers
verify-depth         maximum length of the certificate chain
mutual-authenticate  mutually authenticate
```

To configure a TLS profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **security** and press Enter to access the **session-router** path.

```
ACMEPACKET (configure) # security
```

3. Type **tls-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (security) # tls-profile  
ACMEPACKET (tls-profile) #
```

4. **name**—Enter the name of the TLS profile. This parameter is required; you cannot leave it empty.
5. **end-entity-certificate**—Enter the name of the entity certification record.
6. **trusted-ca-certificates**—Enter the names of the trusted CA certificate records.
7. **cipher-list**—Either use the default **ALL**, or enter a list of ciphers you want to support.
8. **verify-depth**—Specify the maximum depth of the certificate chain that will be verified. The default value is **10**. The valid range is:
  - Minimum—0
  - Maximum—10
9. **mutual-authenticate**—Define whether or not you want the Oracle Communications Session Border Controller to mutually authenticate the client. The default value is **disabled**. The valid values are:
  - enabled | disabled
10. **tls-version**—Enter the TLS version you want to use with this TLS profile. Valid values are **TLSv1**, **SSLv3**, and **compatibility** (default).
11. Save your work.
12. Exit out to the configuration terminal menu to apply the TLS profile.

```
ACMEPACKET (tls-profile) # exit  
ACMEPACKET (security) # exit  
ACMEPACKET (configure) #
```

## Applying a TLS Profile

To apply the TLS profile, you need to specify it for the SIP interface with which it will be used. You must take this step from within the SIP interface configuration.

1. Type **session-router** and press Enter to access the **session-router** path.

```
ACMEPACKET (configure) # session-router
```

2. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-interface  
ACMEPACKET (sip-interface) #
```

3. Select the existing SIP interface to which you want to apply the TLS profile. If you do not know the name of the profile, press Enter again after you use the select command to see a list of all SIP interfaces. Type in the number corresponding to the SIP interface you want to select, and press Enter. You will then be modifying that SIP interface.

```
ACMEPACKET (sip-interface) # select
```

4. Type **sip-ports** and Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-interface) # sip-ports  
ACMEPACKET (sip-port) #
```



5. **transport-protocol**—Change the transport protocol to **TLS**.

```
ACMEPACKET(sip-interface) # transport-protocol tls
```

6. **tls-profile**—Enter the name of the TLS profile you want applied. This is the same value you enter for the name parameter in the TLS profile configuration. This profile will be applied when the transport protocol is TLS.

```
ACMEPACKET(sip-interface) # tls-profile acmepacket
```

7. Save your updated SIP interface configuration.

## Reusing a TLS Connection

The Oracle Communications Session Border Controller supports TLS connection reuse if and when an alias is included in the Via header by the originator of the TLS connection. When this is the case, the Oracle Communications Session Border Controller reuses the same connection for any outgoing request from the Oracle Communications Session Border Controller.

## Keeping Pinholes Open at the Endpoint

The Oracle Communications Session Border Controller provides configurable TCP NAT interval on a per-realm basis. You need to configure a NAT interval for the applicable realm to support either all conforming or all non-conforming endpoints.

- Conforming endpoints use the draft-jennings sipping-outbound-01. It describes how to keep the endpoint keeps the connection alive.



**Note:** Currently the endpoint uses REGISTER.

- Non-conforming endpoints have short NAT interval, where the HNT application with the TCP connection for TLS operates as it does for regular TCP. We give the UA a shorter expires time so that it refreshes frequently, implicitly forcing the UA to keep the TVP socket open and reuse it for further requests (in-dialog or out-of-dialog). Regular requests using TLS sent from the Oracle Communications Session Border Controller to the UA reuse the same TCP connection so that further TLS certificate exchanges are not required.

## Viewing Certificates

You can view either a brief version or detailed information about the certificates.

### Brief Version

Obtaining the brief version uses this syntax, and will appear like the following example:

```
ACMEPACKET# show security certificates brief acmepacket
certificate-record:acmepacket
Certificate:
  Data:
    Version: 3 (0x2)
    Serial Number:
      02:13:02:50:00:84:00:71
    Issuer:
      C=US
      ST=California
      L=San Jose
      O=sipit
      OU=Sipit Test Certificate Authority
    Subject:
      C=US
      ST=MA
      L=Burlington
      O=Engineering
      CN=acme
ACMEPACKET#
```

### Detailed Version

Obtaining the detailed version uses this syntax, and will appear like the following example:

```
ACMEPACKET# show security certificates detail acmepacket
certificate-record:acmepacket
Certificate:
  Data:
    Version: 3 (0x2)
    Serial Number:
      02:13:02:50:00:84:00:71
    Signature Algorithm: sha1WithRSAEncryption
    Issuer:
      C=US
      ST=California
      L=San Jose
      O=sipit
      OU=Sipit Test Certificate Authority
    Validity
      Not Before: Apr 13 21:37:43 2005 GMT
      Not After : Apr 12 21:37:43 2008 GMT
    Subject:
      C=US
      ST=MA
      L=Burlington
      O=Engineering
      CN=acme
    X509v3 extensions:
      X509v3 Subject Alternative Name:
        DNS:pkumar
      X509v3 Basic Constraints:
        CA:FALSE
ACMEPACKET#
```

## Host Certificate Retrieval via SNMP

---

When a security certificate is installed locally on the Oracle Communications Session Border Controller, you can poll the expiration of the certificate using the `apSecurityCertificateTable`.

You can configure the Oracle Communications Session Border Controller to generate the `apSecurityCertExpiredNotification` trap once a certificate has expired. The number of minutes between notifications sent is configured in the security-config parameter `local-cert-trap-int`.

To send a warning of expiration, you can set the security-config parameter `local-cert-exp-warn-period` to the number of days before the locally installed certificate expires in which you would like a warning.

### Host Certificate Retrieval Configuration

To configure the Oracle Communications Session Border Controller to generate traps when a certificate has or is about to expire:

1. Navigate to the security-config configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security#
ACMEPACKET(security)# security-config
ACMEPACKET(security-config)#
```

2. Set the `local-cert-exp-warn-period` parameter to the number of days before the locally installed certificate expires in order to receive a warning. A value of 0 disables the trap.

```
ACMEPACKET(security-config)# local-cert-exp-warn-period 3
ACMEPACKET(security-config)#
```

3. Set the **local-cert-trap-int** parameter for the number of minutes between notifications sent once a certificate has expired. A value of 0 disables the warning trap.

```
ACMEPACKET(security-config)# local-cert-exp-trap-int 15
ACMEPACKET(security-config)#
```

4. Use **done**, **exit**, and **verify-config** to complete required configuration.

## Denial of Service for TLS

This section explains the DoS for TLS feature. With this feature, the Oracle Communications Session Border Controller can provide protection from TCP/TLS message flood by limiting the number of connections from an end point and by limiting the number of simultaneous TCP/TLS connections to a SIP interface.

The Oracle Communications Session Border Controller protects against a flood of invalid TLS messages and against end points establishing TCP/TLS connections or doing an initial registration without then sending any messages. The Oracle Communications Session Border Controller protects against:

- Too many simultaneous TLS connections being requested by a single IP address by limiting the number of TLS connections from a single IP address. There is a maximum simultaneous number of TCP/TLS connections a SIP interface will allow from a single IP address.
- Too many simultaneous TLS connections being requested by limiting the maximum number of connections for a SIP interface. There is a maximum number of simultaneous TCP/TLS connections a SIP interface will allow in aggregate from all IP addresses served by that signaling interface.
- End points establishing TCP/TLS connections without then sending any messages (application layer messages post TLS handshake complete). Triggered by inactivity as measured by lack of any message from this peer.
- End points doing an initial registration without then sending any messages.

This timer could be used by the administrator to detect errors with the SIP configuration. It is expected that whenever an end point establishes a TCP/TLS connection, the end point will keep the connection active by sending messages with REGISTER or by using the NAT interval configuration. Whenever a connection is torn down because of inactivity, a log at the level ERROR is generated.)

- Malformed packets by counting and limiting the maximum number of malformed packets. Whenever an invalid TLS message is received, the internal counter corresponding to invalid-signal-threshold is incremented. When the invalid signal threshold reaches the configured value, the end point will be denied for the configured deny period. (Also requires configuration of the tolerance window in media manager.)
- The max-incoming-conns parameter is well under the maximum number of TLS connections supported by the system. You can set this parameter to it's maximum value of 20000. If you need more than 20000 TLS connections available on this SIP interface, you must set max-incoming-conns to 0 which allows up to the system maximum number of TLS connections, taken on a first come first served basis, on this SIP interface.

## DoS for TLS Configuration

You configure the SIP interface and the realm to support DoS for TLS.

### DoS protection for TLS Connections on the SIP Interface Configuration

To configure the DoS protection for TCP/TLS connections on a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

From this point, you can configure SIP interface parameters. To view all sip-interface parameters, enter a ? at the system prompt.

4. **max-incoming-conns**—Enter the maximum number of simultaneous TCP/TLS connections for this SIP interface. The default value is zero (0) which disables any limit to the number of simultaneous TCP/TLS connections on this SIP interface. The valid range is:

- Minimum—0
- Maximum—20000

5. **per-src-ip-max-incoming-conns**—Enter the maximum number of connections allowed from an end point. The default value is zero (0). The default disables the parameter. The valid range is:

- Minimum—0
- Maximum—20000



**Note:** To make this parameter effective, you need to set the realm's access-control-trust-level to low or medium.

6. **inactive-conn-timeout**—Enter the time in seconds you want a connection from an endpoint discontinued. This provides protection from end points doing an initial registration without sending any messages. The default value is zero (0). The default disables the parameter. The valid range is:

- Minimum—0
- Maximum—999999999

7. Save and activate your configuration.

### Configuring the SIP Configuration

To configure the SIP configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-config** and press Enter. The system prompt changes.

```
ACMEPACKET (session-router) # sip-config  
ACMEPACKET (sip-config) #
```

From this point, you can configure SIP configuration parameters. To view all sip-config parameters, enter a ? at the system prompt.

4. **inactive-dynamic-conn**—Enter the time in seconds after which the Oracle Communications Session Border Controller tears down inactive dynamic TCP connections. Inactive is defined as not transporting any traffic. This protects against endpoints establishing TCP/TLS connections and then not sending messages. The default value is 32. The valid range is:

- Minimum—0
- Maximum—999999999



**Note:** Setting this parameter to 0 disables this parameter.

Because the Oracle Communications Session Border Controller first establishes a TCP connection, then the TLS connection it waits twice the value entered here after the initiation of a TLS connection before tearing down the connection.

After an endpoint establishes a TCP/TLS connection, it is supposed to keep the connection active by sending messages or by using the NAT interval configuration. Whenever a connection is torn down because of inactivity, a log at the level ERROR is generated.

## Configuring the Realm

To configure the realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET(configure)# media-manager
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

From this point, you can configure realm parameters. To view all realm configuration parameters, enter a ? at the system prompt.

4. **deny-period**—Indicate the time period in seconds after which the entry for this host is removed from the deny list. The default value is **30**. The valid range is:

- Minimum—0
- Maximum—4294967295

5. **invalid-signal-threshold**—Enter the number of invalid TLS signaling messages that trigger host demotion. The value you enter here is only valid when the trust level is low or medium. Available values are:

- Minimum—Zero (0) is disabled.
- Maximum—999999999

If the number of invalid messages exceeds this value based on the tolerance window parameter, configured in the media manager, the host is demoted.

The tolerance window default is 30 seconds. Bear in mind, however, that the system uses the same calculation it uses for specifying "recent" statistics in show commands to determine when the number of signaling messages exceeds this threshold. This calculation specifies a consistent start time for each time period to compensate for the fact that the event time, such as a user running a show command, almost never falls on a time-period's border. This provides more consistent periods of time for measuring event counts.

The result is that this invalid signal count increments for two tolerance windows, 60 seconds by default, within which the system monitors whether or not to demote the host. The signal count for the current tolerance window is always added to the signal count of the previous tolerance window and compared against your setting.

6. **access-control-trust-level**—Set the trust level for the host within the realm. The default value is **none**. The valid values are:

- **none**—Host is always untrusted. It is never promoted to the trusted list or demoted to the deny list.
- **low**—Host can be promoted to the trusted list or demoted to the deny list.
- **medium**—Host can be promoted to the trusted list but is only demoted to untrusted. It is never added to the deny list.
- **high**—Host is always trusted.

7. Save and activate your configuration.

## TLS Endpoint Certificate Data Caching

To provide a higher level of security for unified messaging (UM), the Oracle Communications Session Border Controller allows you configure enforcement profiles to cache data from TLS certificates. During the authentication process, the system caches the data so it can use that data in subsequent SIP message processing. Thus the Oracle Communications Session Border Controller can:

- Add custom SIP header populated with information from TLS certificates—When the Oracle Communications Session Border Controller receives an INVITE from a GW, it can write proprietary headers into the SIP message. It uses the certificate information the GW provided during the TLS authentication process with the Oracle Communications Session Border Controller to do so.
- Compare the host of the Request-URI with information from TLS certificates—When an INVITE is destined for the unified messaging server, the Oracle Communications Session Border Controller checks the domain of the Request-URI it has generated prior to HMR application. It does so to verify that the Request-URI matches the domain information the UM server provided during the TLS authentication process with the Oracle Communications Session Border Controller.

TLS endpoint certificate data caching can only apply to call-creating SIP INVITES. The Oracle Communications Session Border Controller looks to the following configurations, in order, to apply an enforcement profile: session agent, realm, and SIP interface associated with the INVITE. As a final step, it checks the SIP profile for enforcement profile association.

### Inserting Customized SIP Headers in an Outgoing INVITE

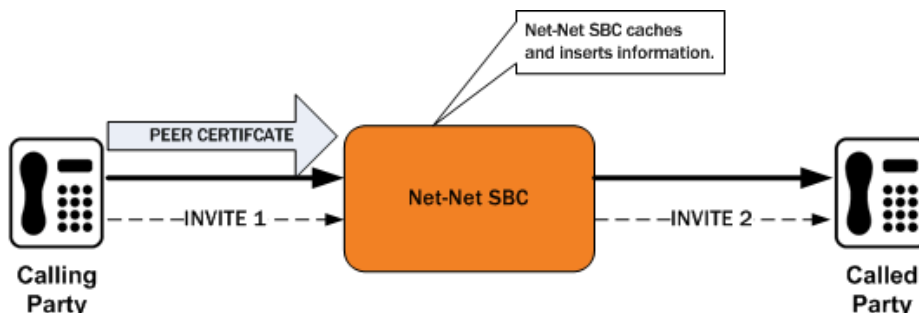
When the Oracle Communications Session Border Controller establishes a new TLS connection, it caches the following peer certificate attributes:

- Certificate Subject Name
- Certificate Subject Alternative Name (only DNS)

The Oracle Communications Session Border Controller constructs a customized P-Certificate-Subject-Common-Name SIP header and inserts the header into the outgoing INVITE with the Certificate Subject Name. The Oracle Communications Session Border Controller also constructs and inserts in the outgoing INVITE one or more P-Certificate-Subject-Alternative-Name SIP headers.

If you enable this capability and the incoming INVITE already has P-Certificate-Subject-Common-Name and P-Certificate-Subject-Alternative-Name headers, the Oracle Communications Session Border Controller strips them before inserting the new customized ones. It does so to avoid the risk of any attempt to spoof the headers and thereby gain unauthorized access to the UM server.

The following diagram shows a scenario where the calling party establishes a TLS connection with the Oracle Communications Session Border Controller. Because mutual authentication is enabled, the Oracle Communications Session Border Controller receives the peer certificate and caches required information from it. This information is inserted in the outgoing INVITE.



The peer certificate from the calling party during the TLS handshake with the Oracle Communications Session Border Controller looks like the following example.

```
Certificate:
  Data:
    Version: 3 (0x2)
    Serial Number: 9 (0x9)
    Signature Algorithm: sha1WithRSAEncryption
    Issuer: C=US, ST=MA, L=Woburn, O=Smith Securities, OU=Certificate
    Authority Dept, CN=Smith Certificate Authority/emailAddress=Smith@CA.com
    Validity
      Not Before: Dec 10 21:14:56 2009 GMT
```

```

Not After : Jul 11 21:14:56 2019 GMT
Subject: C=US, ST=MA, L=Burlington, O=Acme Packet, OU=Certificate
Authority Dept, CN=*.acme.com/emailAddress=ph1Client@acme.com
Subject Public Key Info:
  Public Key Algorithm: rsaEncryption
  RSA Public Key: (2048 bit)
X509v3 extensions:
  X509v3 Basic Constraints:
    CA:FALSE
  X509v3 Issuer Alternative Name:
    email:Smith@CA.com
  X509v3 Subject Alternative Name:
DNS:gw1.acme.com, DNS:gw3.ano.com, DNS:gw2.some.com
  X509v3 Key Usage: critical
  Digital Signature, Key Encipherment
Signature Algorithm: sha1WithRSAEncryption

```

The outgoing SIP INVITE (INVITE 2 in the diagram) looks like the following sample. Bold text shows where the Oracle Communications Session Border Controller uses information from the certificate.

```

INVITE sip:222222@acme.com:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.27.113:5060;branch=z9hG4bK4jmg29cmm8l0cg7smmrn85o4q7
From: 111111 <sip:111111@acme.com>;tag=_ph1_tag
To: 222222 <sip:222222@acme.com>
Call-ID: _1-2_call_id-10147@acme.com-1-
CSeq: 1 INVITE
Contact: <sip:111111@172.16.27.113:5060;transport=udp>
P-Certificate-Subject-Common-Name: *.acme.com
P-Certificate-Subject-Alternative-Name: gw1.acme.com
P-Certificate-Subject-Alternative-Name: gw3.ano.com
P-Certificate-Subject-Alternative-Name: gw2.some.com
Max-Forwards: 69
Subject: TBD
Content-Type: application/sdp
Content-Length: 138
Route: <sip:222222@172.16.27.188:5060;lr>
v=0
o=user1 53655765 2353687637 IN IP4 172.16.27.113
s=-
c=IN IP4 172.16.27.113
t=0 0
m=audio 20000 RTP/AVP 0
a=rtpmap:0 PCMU/8000

```

## Validating the Request-URI Based on Certificate Information

When you configure the Oracle Communications Session Border Controller to cache TLS certificate information to validate Request-URIs, it stores the Certificate Subject Name and Certificate Subject Alternative Name (only DNS) it learns from peer certificate attributes. It then takes these actions:

- Extracts the host from the Request-URI of the outgoing INVITE
- Compares this host (exact or wildcard match) with the Certificate Common Name or Certificate Subject Alternative name of the certificate it has received
- Sends out an INVITE if the Certificate Common Name or Certificate Subject Alternative name match; Sends a 403 Forbidden rejection to the endpoint from it received the INVITE if there is no match

Wildcard matching applies only to the prefix of the Request-URI:

```

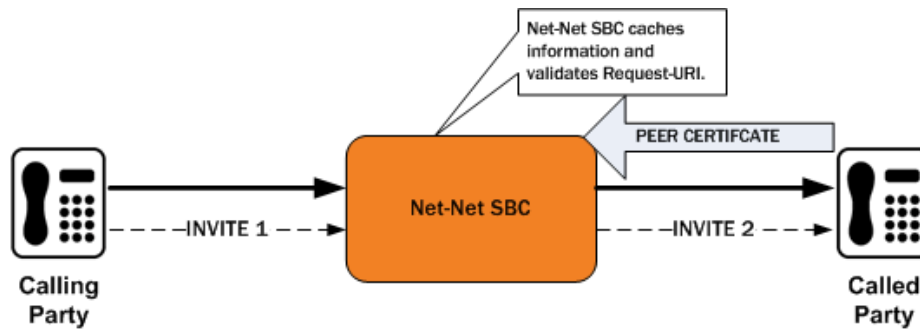
*.acme.com
*.*.acmepacket.com

```

This diagram shows a peering scenario where the Oracle Communications Session Border Controller receives an INVITE from the calling party, which it processes and prepares to send out INVITE 2. After establishing a TLS connection with the called party and caching the required information, the Oracle Communications Session Border

## Security

Controller validates the Request-URI. Once validation occurs, the Oracle Communications Session Border Controller sends INVITE 2.



The peer certificate from the called party during the TLS handshake with the Oracle Communications Session Border Controller would look like this. Relevant information in the sample appears in **bold** font.

```
Certificate:
  Data:
    Version: 3 (0x2)
    Serial Number: 9 (0x9)
    Signature Algorithm: sha1WithRSAEncryption
    Issuer: C=US, ST=MA, L=Woburn, O=Smith Securities, OU=Certificate
    Authority Dept, CN=Smith Certificate Authority/emailAddress=amith@CA.com
    Validity
      Not Before: Dec 10 21:14:56 2009 GMT
      Not After : Jul 11 21:14:56 2019 GMT
    Subject: C=US, ST=MA, L=Woburn, O=Acme Packet, OU=Certificate
    Authority Dept, CN=*.acme.com/emailAddress=ph2Server@acme.com
    Subject Public Key Info:
      Public Key Algorithm: rsaEncryption
      RSA Public Key: (2048 bit)
    X509v3 extensions:
      X509v3 Basic Constraints:
        CA:FALSE
      X509v3 Issuer Alternative Name:
        email:Smith@CA.com
      X509v3 Subject Alternative Name:
        DNS:gw1.acme.com, DNS:*.ano.com, DNS:*.some.com
      X509v3 Key Usage: critical
        Digital Signature, Key Encipherment
    Signature Algorithm: sha1WithRSAEncryption
```

The outgoing SIP INVITE (INVITE 2 in the diagram) would then look like the sample below. The INVITE is sent because **smith.acme.com** matches the common name **\*.acme.com**. Other valid SIP Request-URIs would be:

```
222222@gw1.acme.com
222222@smith.ano.com
222222@amith.some.com
```

You can see where the system uses information from the certificate; the text is **bold**.

```
INVITE sip:222222@smith.acme.com:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.27.113:5060;branch=z9hG4bK4jmg29cmm8l0cg7smmrn85o4q7
From: 111111 <sip:111111@acme.com>;tag=_ph1_tag
To: 222222 <sip:222222@acme.com>
Call-ID: _1-2_call_id-10147@acme.com-1-
CSeq: 1 INVITE
Contact: <sip:111111@172.16.27.113:5060;transport=udp>
Max-Forwards: 69
Subject: TBD
Content-Type: application/sdp
Content-Length: 138
Route: <sip:222222@172.16.27.188:5060;lr>
```



```
v=0
o=user1 53655765 2353687637 IN IP4 172.16.27.113
s=-
c=IN IP4 172.16.27.113
t=0 0
m=audio 20000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

## TLS Endpoint Certificate Data Caching Configuration

To configure SIP endpoint certificate data caching for an enforcement profile:

1. Access the **enforcement-profile** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# enforcement-profile
ACMEPACKET(enforcement-profile)#
```

2. Select the **enforcement-profile** object to edit.

```
ACMEPACKET(enforcement-profile)# select
<name>:

ACMEPACKET(enforcement-profile)#
```

3. **add-certificate-info**—Enter a list of one or more certificate attribute names to enable TLS certificate information caching and insertion of cached certificate information into a customized SIP INVITES. This parameter is empty by default.

If you want to list more than one value, enclose the value in quotation marks (“ ”) and separate the values with Spaces.

```
ACMEPACKET(enforcement-profile)# add-certificate-info "sub-common-name sub-alt-name-DNS"
```

4. **certificate-ruri-check**—Change this parameter from **disabled**, its default, to **enabled** if you want your Oracle Communications Session Border Controller to cache TLS certificate information and use it to validate Request-URIs. Enabling this parameter also means the Oracle Communications Session Border Controller will use the cached TLS certificate information in a customized SIP INVITE.
5. Type **done** to save your configuration.

## Untrusted Connection Timeout for TCP and TLS

You can configure the Oracle Communications Session Border Controller for protection against starvation attacks for socket-based transport (TCP or TLS) for SIP access applications. During such an occurrence, the attacker would open a large number of TCP/TLS connections on the Oracle Communications Session Border Controller and then keep those connections open using SIP messages sent periodically. These SIP messages act as keepalives, and they keep sockets open and consume valuable resources.

Using its ability to promote endpoints to a trusted status, the Oracle Communications Session Border Controller now closes TCP/TLS connections for endpoints that do not enter the trusted state within the period of time set for the untrusted connection timeout. The attacking client is thus no longer able to keep connections alive by sending invalid messages.

This feature works by setting a value for the connection timeout, which the Oracle Communications Session Border Controller checks whenever a new SIP service socket for TCP or TLS is requested. If the timer's value is greater than zero, then the Oracle Communications Session Border Controller starts it. If the timer expires, then the Oracle Communications Session Border Controller closes the connection. However, if the endpoint is promoted to the trusted state, then the Oracle Communications Session Border Controller will cancel the timer.

### Caveats

This connection timeout is intended for access applications only, where one socket is opened per-endpoint. This means that the timeout is not intended for using in peering applications; if this feature were enabled for peering, a single malicious SIP endpoint might cause the connection to be torn down unpredictably for all calls.

### Untrusted Connection Timeout Configuration for TCP and TLS

The untrusted connection timer for TCP and TLS is set per SIP interface.

To set the untrusted connection timer for TCP and TLS:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the signaling-level configuration elements.

```
ACMEPACKET (configure)# session-router  
ACMEPACKET (session-router)#
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router)# sip-interface  
ACMEPACKET (sip-interface)#
```

If you are adding support for this feature to a pre-existing SIP configuration, then you must select (using the ACLI **select** command) the configuration that you want to edit.

4. **untrusted-conn-timeout**—Enter the time in seconds that you want the Oracle Communications Session Border Controller to keep TCP and TLS connections open for untrusted endpoints. The default value is **0**, which will not start the timer. The valid range is:
  - Minimum—0
  - Maximum—999999999
5. Save and activate your configuration.

### Online Certificate Status Protocol

---

The Online Certificate Status Protocol (OCSP) is defined in RFC 2560, X.509 Internet Public Key Infrastructure Online Certificate Status Protocol - OCSP. The protocol enables users to determine the revocation state of a specific certificate, and may provide a more efficient source of revocation information than is possible with Certificate Revocation Lists (CRL).

The protocol specifies the data exchanged between an OCSP client (for example, the Oracle Communications Session Border Controller) and an OCSP responder, the Certification Authority (CA), or its delegate, that issued the target certificate. An OCSP client issues a request to an OCSP responder and suspends acceptance of the certificate in question until the responder replies with a certificate status.

Certificate status is reported as

- good
- revoked
- unknown

good indicates a positive response to the status inquiry. At a minimum, this positive response indicates that the certificate is not revoked, but does not necessarily mean that the certificate was ever issued or that the time at which the response was produced is within the certificate's validity interval.

revoked indicates that the certificate has been revoked, either permanently or temporarily.

unknown indicates that the responder cannot identify the certificate.

## Caveats

OCSP is currently supported only on TLS interfaces; it is not currently supported for use with IKEv1 and IKEv2.

## Online Certificate Status Protocol Configuration

OCSP configuration consists of

1. Configuring one or more certificate status profiles; each profile contains information needed to contact a specific OCSP responder.
2. Enabling certificate revocation checking by assigning a certificate status profile to a previously configured TLS profile.

To create a certificate status profile:

3. From superuser mode, use the following command sequence to access cert-status-profile configuration mode. While in this mode, you provide the information required to access one or more OCSP responders.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# cert-status-profile
ACMEPACKET(cert-status-profile)#
```

4. Use the required **name** parameter to identify this cert-status-profile instance — each profile instance provides configuration data for a specific OCSP responder. **name** is used to distinguish between multiple profile instances.
5. Use the required **ip-address** parameter to specify the IPv4 address of the OCSP responder.
6. Use the optional **port** parameter to specify the destination port.

In the absence of an explicitly configured value, the default port number of 80 is used.

7. Use the optional **realm-id** parameter to specify the realm used to transmit OCSP requests.

In the absence of an explicitly configured value, the default specifies service across the wancom0 interface.

8. Use the optional **requester-cert** parameter only if OCSP requests are signed; ignore this parameter if requests are not signed.

RFC 2560 does not require signed requests; however, local or CA policies can mandate digital signature..

9. Use the required **responder-cert** parameter to identify the certificate used to validate OCSP responses — a public key of the OCSP responder.

RFC 2560 requires that all OCSP responders digitally sign OCSP responses, and that OCSP clients validate incoming signatures.

Provide the name of the certificate configuration element that contains the certificate used to validate the signed OCSP response.

10. Use the optional **retry-count** parameter to specify the maximum number of times to retry an OCSP responder in the event of connection failure.

If the retry counter specified by this parameter is exceeded, the OCSP requester either contacts another responder (if multiple responders have been configured within this cert-status-profile) and quarantine the unavailable responder for a period defined the **dead-time** parameter.

In the absence of an explicitly configured value (an integer within the range 0 through 10), the default of 1 is used.

```
ACMEPACKET(cert-status-profile)# retry-count 2
ACMEPACKET(cert-status-profile)#
```

11. Use the optional **dead-time** parameter to specify the quarantine period imposed on an unavailable OCSP responder.

In the absence of an explicitly configured value (an integer within the range 0 through 3600 seconds), the default value (0) is used.

Customer implementations utilizing a single OCSP responder are encouraged to retain the default value, or to specify a brief quarantine period to prevent lengthy service outages.

12. Retain default values for the **type** and **trans-protocol** parameter to specify OCSP over an HTTP transport protocol.
13. Use **done**, **exit**, and **verify-config** to complete configuration of this cert-status-profile instance.
14. Repeat Steps 1 through 11 to configure additional certificate status profiles.

To enable certificate status checking:

15. Move to **tls-profile** configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# tls-profile
ACMEPACKET(tls-profile)#
```

16. Use the required **cert-status-check** parameter to enable OCSP in conjunction with an existing TLS profile.
17. Use the required **cert-status-profile-list** parameter to assign one or more cert-status-profiles to the current TLS profile.

Each assigned cert-status-profile provides the information needed to access a single OCSP responder.

Use quotation marks to assign multiple OCSP responders. The following sequence assigns three cert-status-profiles, VerisignClass3Designate, Verisign-1, and Thawte-1 to the TLS-1 profile.

18. Use **done**, **exit**, and **verify-config** to complete configuration.

### Sample Configuration:

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# cert-status-profile
ACMEPACKET(cert-status-profile)# name VerisignClass3Designate
ACMEPACKET(cert-status-profile)# ip-address 192.168.7.100
ACMEPACKET(cert-status-profile)# responder-cert VerisignClass3ValidateOCSP
ACMEPACKET(cert-status-profile)# done
ACMEPACKET(cert-status-profile)# exit
...
...
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# tls-profile
ACMEPACKET(tls-profile)# select
<name>:
1. TLS-1
2. TLS-2
3. TLS-3
selection: 1
ACMEPACKET(tls-profile)# cert-status-check enabled
ACMEPACKET(cert-status-profile)# cert-status-profile-list
VerisignClass3Designate Verisign-1 Thawte-1
ACMEPACKET(cert-status-profile)# done
ACMEPACKET(cert-status-profile)# exit
```

## Unreachable OCSR

With OCSP enabled, the client implementation running on the Oracle Communications Session Border Controller supports message exchange between the Oracle Communications Session Border Controller and an OCSR as specified in RFC 2560, *X.509 Internet Public Key Infrastructure Online Certificate Status Protocol - OCSP*. The Oracle Communications Session Border Controller contacts the OCSR whenever a remote client attempts to establish an SSL/TLS connection with the Oracle Communications Session Border Controller. The Oracle Communications Session Border Controller sends a request to the OCSR to check the current status of the certificate presented by the remote client. The Oracle Communications Session Border Controller suspends processing of the SSL/TLS connection request pending receipt of the OCSR response. In previous releases (prior to Version S-CX6.3.0), a good OCSR response resulted in the establishment of a secure SSL/TLS connection. A revoked or unknown OCSR response, or the failure to reach an OCSR, resulted in the rejection of the connection attempt.

This behavior, which adheres to the requirements of RFC 2560, conflicts with the requirements of Section 5.4.6.2.1.6.4.a.i of UCR 2008 which requires an OCSRP client to attempt authentication of remote clients in the event of an unreachable OCSR.

Release S-CX6.3F1 adds a new attribute (**ignore-dead-responder**) to the TLS profile configuration element to provide compliance with DISA/DoD requirements specifying OCSRP client operations when faced with unreachable OCSRs. By default, the attribute is disabled meaning that all client connections will be disallowed in the event of unreachable OCSRs.

In DISA/DoD environments **ignore-dead-responder** should be enabled, allowing local certificate-based authentication by the Oracle Communications Session Border Controller in the event of unreachable OCSRs. Successful authentication is achieved if the certificate presented by the remote client was signed by a Certificate Authority (CA) referenced by the **trusted-ca-certificates** attribute. If the local authentication succeeds, the secure TLS/SSL connection is established; otherwise the connection is rejected.

## Unreachable OCSR Configuration

The following sample configuration implements DISA/DoD-compliant client behavior in the event of an unreachable OCSR.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security#
ACMEPACKET(security)# tls-profile
ACMEPACKET(security)# show
tls-profile
  name                DoD
  end-entity-certificate  sylarCert-2048
  trusted-ca-certificates  dod1 dod2 disaA disaB IBM1
  cipher-list           all
  verify-depth          10
  mutual-authenticate    disabled
  tls-version            tlsv1
  cert-status-check      enabled
  cert-status-profile-list  DoD
  ignore-dead-responder  enabled
  ...
  ...
ACMEPACKET(tls-profile)#
```

## OCSR Status Monitoring

OCSR monitoring is provided to track the reachability of individual OCSRs, and, in topologies containing multiple OCSRs, the overall availability of OCSR service.

If monitoring is enabled for individual OCSRs, reachability is monitored by observing responder transactions.

Initially, all OCSRs are considered reachable. If a previously reachable OCSR fails to respond to a certificate status request, the Oracle Communications Session Border Controller marks the OCSR as unreachable, and generates an SNMP trap and log entry indicating that status. If a previously unreachable OCSR respond to a certificate status request, the Oracle Communications Session Border Controller returns the OCSR to the reachable status, and generates an SNMP trap and log entry indicating that status change.

Use the following procedure to enable monitoring of individual OCSRs.

1. Navigate to the new security-config configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security#
ACMEPACKET(security)# security-config
ACMEPACKET(security-config)#
```

2. Enable monitoring of individual OCSRs by setting the **ocsr-monitoring-traps** attribute to **enabled**; this attribute is disabled by default.

```
ACMEPACKET(security-config)# ocsr-monitoring-traps enabled
ACMEPACKET(security-config)#
```

### 3. Use **done**, **exit**, and **verify-config** to complete required configuration.

Reachability status of individual OCSRs is aggregated to monitor the overall availability of OCSR service. Using the procedure explained above, the Oracle Communications Session Border Controller maintains a count of all OCSRs, and of all reachable OCSRs.

- If all OCSRs are reachable, the Oracle Communications Session Border Controller generates a trap and log entry noting this optimal state.
- If all OCSRs are unreachable, the Oracle Communications Session Border Controller generates a trap and log entry noting this erroneous state.
- When the Oracle Communications Session Border Controller transitions from either of the two states described above (in the optimal state, when an OCSR becomes unreachable; in the erroneous state, when an unreachable OCSR becomes reachable), the Oracle Communications Session Border Controller generates a trap and log entry indicating that an unspecified number of OCSRs are reachable.

Monitoring of OCSR service availability is a by-product of enabling SNMP; no further configuration is required.

## OCSR Access via FQDN

Prior software releases supported OCSR access only via IPv4 addresses and port numbers. In response to a DISA/DoD request, Release S-CX6.3F1 adds support for OCSR access via FQDNs. Since multiple public key infrastructure (PKI) elements capable of supporting OCSP requests can exist within a DISA/DoD environment, the Domain Name Service (DNS) lookup that resolves the FQDN can result in more than one OCSR IP address being returned to the Oracle Communications Session Border Controller in its role of OCSP client. When processing a lookup that contains more than one IP address, the Oracle Communications Session Border Controller uses a round-robin algorithm to select from the list of OCSR addresses.

OCSR Access via FQDN is available on all media interfaces and on the wancom0 administrative interface. Note that support for FQDN-based access is requires the configuration of DNS support.

If the **realm** attribute is configured in the certificate-status-profile configuration element, the required DNS query is issued on the corresponding network interface. This model requires configuration of the **dns-ip-primary** attribute, and optionally the **dns-ip-backup1** and **dns-ip-backup2** attributes for the realm's network interface.

If the **realm** attribute is not configured in the certificate-status-profile, the required DNS query is issued on the wancom0 interface. This model requires configuration of the **dns-ip-primary** attribute, and optionally the **dns-ip-backup1** and **dns-ip-backup2** attributes for the wancom0 interface.

Access via an FQDN is supported by a new attribute (**hostname**) in the cert-status-profile configuration element.

The Oracle Communications Session Border Controller allows configuration of both an OCSR IP address and port number (using the **ip-address** and **port** attributes) and an OCSR domain (using the **hostname** attribute).

In such cases the **verify-config** command issues a warning and notes that IP address-based access will be used.

### OCSR Access Configuration via IP Address

The following sample configuration accesses an OCSR at 192.168.7.100:8080.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security#
ACMEPACKET(security)# cert-status-profile#
ACMEPACKET(cert-status-profile)# show
cert-status-profile
  name                defaultOCSP
  ip-address          192.168.7.100
```

```

hostname
port 8080
type OSCP
trans-proto HTTP
requestor-cert ocspsVerisignClient
responder-cert VerisignCA2
trusted-cas
realm-id admin
retry-count 1
dead-time 0
last-modified-by
last-modified-date
ACMEPACKET(cert-status-profile) #

```

### OCSR Access Configuration via FQDN

The following sample configuration accesses one or more OCSRs at example.disa.mil.

Note that in the absence of a specified domain, the wancom0 interface must be DNS-enabled.

```

ACMEPACKET# configure terminal
ACMEPACKET(configure)# security#
ACMEPACKET(cert-status-profile)# show
cert-status-profile
  name DISAdomain2
  ip-address
  hostname example.disa.mil
  port
  type OSCP
  trans-proto HTTP
  requestor-cert
  responder-cert
  trusted-cas dod1 dod2 disaA disaB IBM1
  realm-id
  retry-count 1
  dead-time 0
  last-modified-by
  last-modified-date
ACMEPACKET(cert-status-profile) #

```

## Direct and Delegated Trust Models

RFC 2560 specifies that an OCSR must digitally sign OSCP responses, and that an OSCP client must validate the received signature. In prior releases, successful validation of the signed response served to authenticate the responder. Such an authentication method is referred to as a direct trust model in that it does not require confirmation from a trusted Certificate Authority (CA). Rather it requires that the OSCP client be in possession of the public counterpart of the private key used by the OCSR to sign the response. This certificate is identified by the **responder-cert** attribute in the cert-status-profile configuration element. Prior to Release S-CX6.3F1, authentication via signature validation was the only authentication method provided by the OSCP client implementation.

Release S-CX6.3F1 continues support for the direct trust model, while also supporting an alternative delegated trust model as described in Section 5.4.6.2.1.6.1.e.3.c of UCR 2010. The delegated trust model requires that OCSR be authenticated by a trusted CA. Within the DISA/DoD delegated trust model, an OCSR certificate is appended to every response, thus eliminating the need for a pre-provisioned responder certificate. The appended certificate is a signing certificate issued and signed by a DoD-approved CA that issued the certificate that is being validated. These OCSR certificates have a short lifespan and are reissued regularly.

### Direct Trust Model Configuration

The direct trust model is used in virtually all commercial/enterprise environments. Configuration of the direct trust model is unchanged from that contained in the latest version of your hardware or the Oracle Communications Session Border Controller *Configuration Guide*.

### Delegated Trust Model Configuration

The delegated trust model is used exclusively in some strict DISA/DoD environments; other DISA/DoD environments may support both the direct and delegated trust models.

Use the following procedure to configure OCSF for DISA/DoD environments.

1. From superuser mode, use the following command sequence to access cert-status-profile configuration mode. While in this mode, you configure a cert-status-profile configuration element, a container for the information required to access a single, specific OCSR.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# cert-status-profile
ACMEPACKET(cert-status-profile)#
```

2. The **name** attribute differentiates cert-status-profile configuration elements one from another. Each cert-status-profile provides configuration information for a single, specific OCSR responder.
3. The **type** attribute selects the certificate revocation check methodology, the only currently supported methodology is OCSR.
4. Retain the default value (http) for **trans-protocol** attribute, which identifies the transport method used to access the OCSR.
5. The **ip-address** attribute works in conjunction with the **port** attribute to provide the IP address of the OCSR.

**ip-address** identifies the OCSR by its IP address. **port** identifies the port monitored by the HTTP server for incoming OCSR requests.

The **port** attribute can be safely ignored if the OCSR is specified as a FQDN by the **host-name** attribute, but is required if the OCSR is identified by the **ip-address** attribute.

Allowable **port** values are integers within the range 1025 through 65535. In the absence of an explicitly configured value, the system provides a default value of 80, the well-known HTTP port.

6. Alternatively, use the **host-name** attribute to identify the OCSR.

**host-name** identifies the OCSR by a FQDN.

If you provide both an IPv4 address/port number and a FQDN, the Oracle Communications Session Border Controller uses the IP address/port number and ignores the FQDN.

If values are provided for both attributes, the Security Gateway uses the IP address and ignores the **host-name** value.

7. The **realm-id** attribute specifies the realm used to access the OCSR.

In the absence of an explicitly configured value, the Oracle Communications Session Border Controller provides a default value of wancom0, specifying OCSR transmissions across the wancom0 management interface.

If the OCSR identified by a FQDN, the realm identified by **realm-id** must be DNS-enabled.

8. The **requester-cert** attribute is meaningful only if OCSR requests are signed; ignore this attribute if requests are not signed.

RFC 2560 does not require the digital signature of OCSR requests. OCSRs, however, can impose signature requirements.

If a signed request is required by the OCSR, provide the name of the certificate configuration element that contains the certificate used to sign OCSR requests.

9. The **responder-cert** attribute identifies the certificate used to validate signed OCSR response — a public key of the OCSR.

In DISA/DoD environments that support the direct trust model, optionally provide the name of the certificate configuration element that contains the certificate used to validate the signed OCSR response.



If a **responder-cert** is provided, it is only used if the OCSP response has no appended certificates, in which case the OCSP client attempts to validate the response signature. Depending on the validation failure or success, the response is rejected or accepted.

If the OCSP response has an appended certificate or certificate chain, the **responder-cert** is ignored, and the trusted-cas list is used to validate the appended certificate(s).

10. The **trusted-cas** attribute (a list of certificate configuration objects) identifies the approved DoD-approved CAs that sign OCSR certificates.

In DISA/DoD environments that support the delegated trust model, you must provide a list of CAs used to validate the received certificate.

If a certificate or a certificate chain is appended to the OCSP response, the OCSP client verifies that the first certificate signed the response, and that the CA is trusted by the Oracle Communications Session Border Controller (that is, the CA certificate is contained in the **trusted-cas** list. The client then walks through each additional certificate (if any exist) ensuring that each certificate is also trusted. If all certificates are trusted, the OCSP response is accepted; otherwise, it is rejected.

11. The **retry-count** attribute specifies the maximum number of times to retry an OCSP responder in the event of connection failure.

If the retry counter specified by this attribute is exceeded, the OCSP requester contacts another responder (if multiple responders have been configured) and quarantines the unavailable responder for a period defined the **dead-time** attribute.

In the absence of an explicitly configured value (an integer within the range 0 through 10), the Oracle Communications Session Border Controller provides a default value of 1 (connection retries).

12. The **dead-time** attribute specifies the quarantine period imposed on an unavailable OCSR.

In the absence of an explicitly configured value (an integer within the range 0 through 3600 seconds), the Oracle Communications Session Border Controller provides a default value of 0 (no quarantine period).

Customer implementations utilizing a single OCSP responder are encouraged to retain the default value, or to specify a brief quarantine period to prevent lengthy service outages.

13. Use **done**, **exit**, and **verify-config** to complete configuration of this cert-status-profile instance.

14. Repeat Steps 1 through 13 to configure additional cert-status-profile configuration elements.

## IPv6-IPv4 Internetworking

The Oracle Communications Session Border Controller supports the following internetworking environments:


SIP-TLS IPv6 endpoints with SIP-TLS IPv4 endpoints

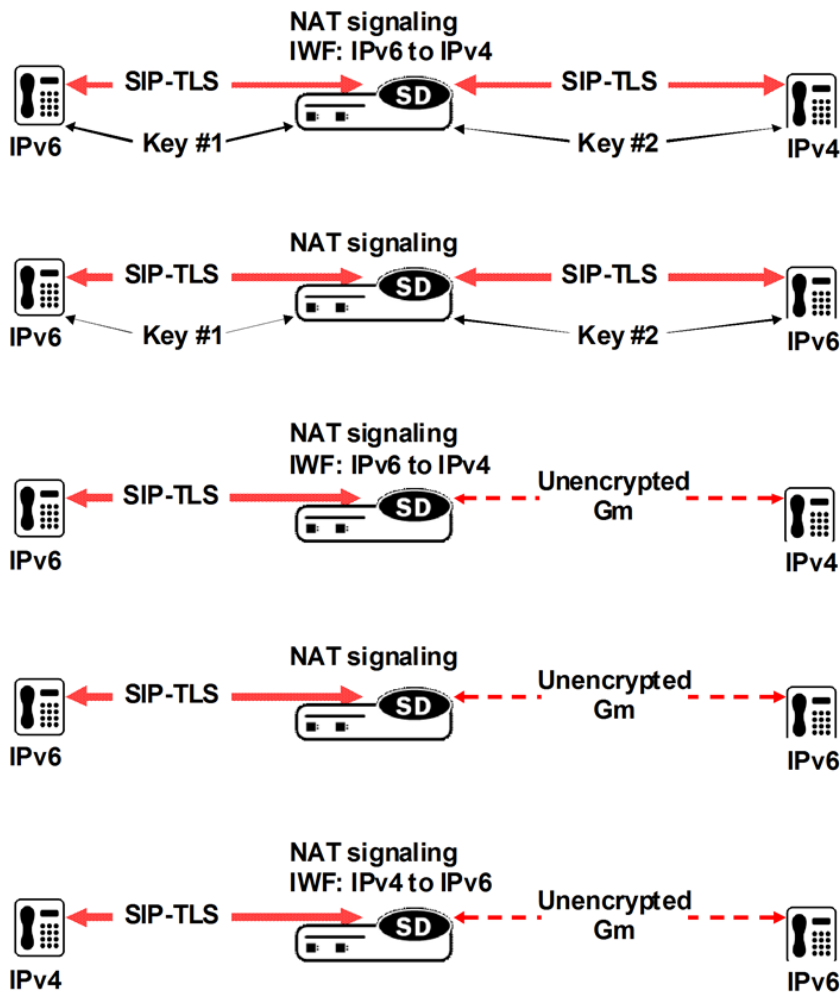
SIP-TLS IPv6 endpoints with SIP-TLS IPv6 endpoints

SIP-TLS IPv6 endpoints with non-SIP-TLS (unencrypted) IPv4 endpoints

SIP-TLS IPv6 endpoints with non-SIP-TLS (unencrypted) IPv6 endpoints

SIP-TLS IPv4 endpoints with non-SIP-TLS (unencrypted) IPv6 endpoints

 **Note:** Previously delivered TLS functionality, for example, support for certificate extensions, and support for certificate chain processing, is not affected by IPv6-IPv4 internetworking.



### TLS SRTP Decryption

The Oracle Communications Session Border Controller enables the unencrypted replication of incoming and outgoing TLS/SRTP packets and delivery of such unencrypted packets to a CRS.

Decrypted packets are forwarded to the CSR as shown below.

Original encrypted packet:

Encrypted Payload	SSL Header	TCP Header	IP Header
-------------------	------------	------------	-----------

Replicated unencrypted packet:

Decrypted Payload	UDP Header (original src/dest ports)	IP Header (original src/dest IP addresses)	IP Header
-------------------	-----------------------------------------	-----------------------------------------------	-----------

### TLS SRTP Decryption Configuration

The `decrypt-tls` attribute enables decryption of TLS/SRTP packets. `decrypt-tls` is disabled by default, meaning that TLS/SRTP forwarded to a CRS are encrypted.

Use the following procedure to enable delivery of unencrypted TLS/SRTP packets to a CRS.

1. Navigate to the new call-recording-server configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router#
ACMEPACKET(session-router)# call-recording-server
ACMEPACKET(call-recording-server)#
```

2. Enable TLS/SRTP decryption by setting the **decrypt-tls** attribute to **enabled**.

```
...
...
...
ACMEPACKET(call-recording-server)# decrypt-tls enabled
ACMEPACKET(call-recording-server)#
...
...
...
ACMEPACKET(call-recording-server)#
```

3. Use **done**, **exit**, and **verify-config** to complete required configuration.

## SRTP MIKEY IPv6 Support

The Oracle Communications Session Border Controller supports IPv6 addressing for SRTP media streams with Multimedia Internet Keying (MIKEY) key exchange for SRTP media streams. MIKEY is defined in RFC 3830, *MIKEY: Multimedia Internet KEYing*.

### Supported Topologies

The following internetworking topologies are supported:

- SRTP IPv6 endpoints with SRTP IPv4 endpoints
- SRTP IPv6 endpoints with SRTP IPv6 endpoints
- SRTP IPv6 endpoints with RTP (unencrypted) IPv4 endpoints
- SRTP IPv6 endpoints with RTP (unencrypted) IPv6 endpoints
- SRTP IPv4 endpoints with RTP (unencrypted) IPv6 endpoints

IPv6 support must be globally enabled to support SRTP internetworking as listed above.

If IPv6 has not been globally enabled, use the following procedure to do so. If IPv6 is currently enabled, no additional configuration is required.

1. Navigate to the system-config configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# system
ACMEPACKET(system)# system-config
ACMEPACKET(system-config)#
```

2. Enable IPv6 by setting the **ipv6-support** attribute to **enabled**; this attribute is disabled by default.

```
ACMEPACKET(system-config)# ipv6-support enabled
ACMEPACKET(system-config)#
```

3. Use **done**, **exit**, and **verify-config** to complete required configuration.

4. Reboot the Oracle Communications Session Border Controller.

## SRTP IPv4 IPv6 Internetworking

Internetworking IPv4 and IPv6 media while using SDES as the key exchange protocol is supported. Previous versions supported SRTP IPv4/IPv6 internetworking using the MIKEY key exchange protocol.

SRTP is defined in RFC 3711, *The Secure Real-time Transport Protocol (SRTP)*. It provides confidentiality, message authentication, and replay protection for RTP media and control traffic. SDES is defined in RFC 4568, *Session*

*Description Protocol (SDP) Security Descriptions for Media Streams.* This RFC describes a new SDP cryptographic attribute that provides a secure method to provide security for unicast media streams.

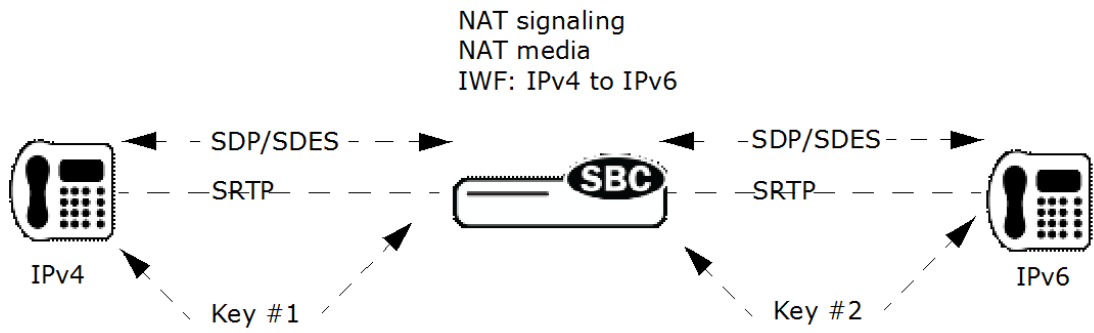
### Hardware Requirements

SRTP IPv4 and IPv6 internetworking requires the Enhanced Traffic Controller (ETC) NIU.

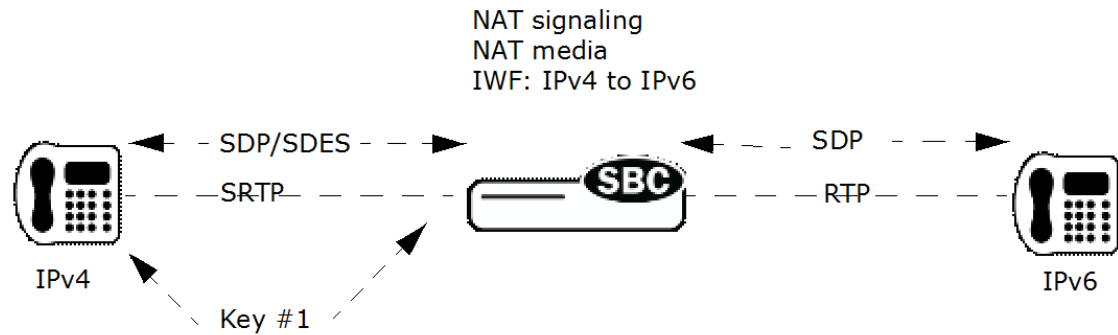
### Supported Topologies

The following internetworking topologies are supported and illustrated below

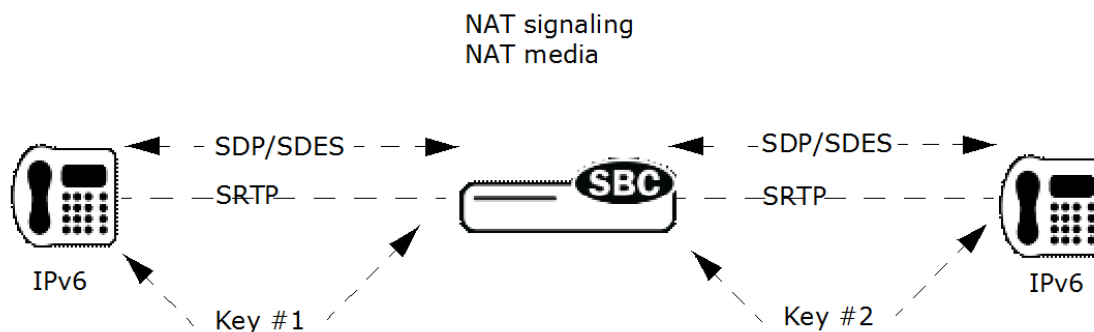
- SRTP IPv4 endpoints with SRTP IPv6 endpoints
- SRTP IPv4 endpoints with RTP (unencrypted) IPv6 endpoints
- SRTP IPv6 endpoints with SRTP IPv6 endpoints
- SRTP IPv6 endpoints with RTP (unencrypted) IPv4 endpoints
- SRTP IPv6 endpoints with RTP (unencrypted) IPv6 endpoints



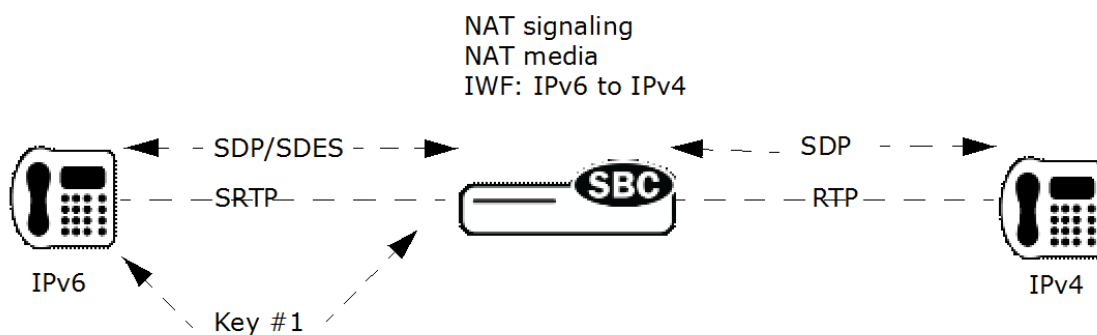
SRTP/SDES IPv4 to SRTP/SDES IPv6 Internetworking



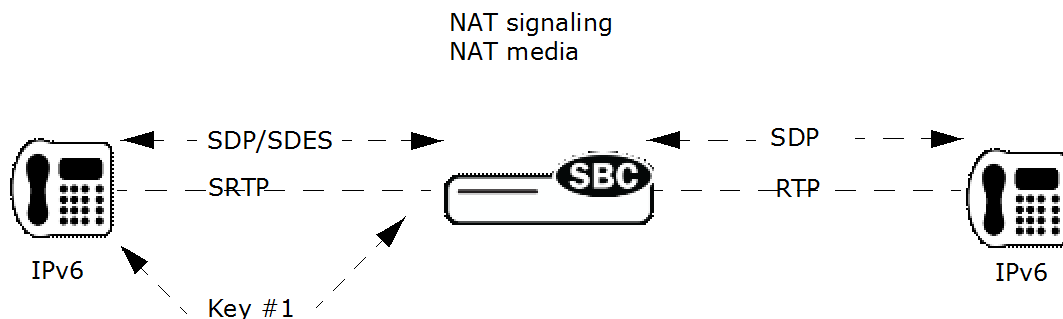
SRTP/SDES IPv4 to RTP IPv6 Internetworking



SRTP/SDES IPv6 to SRTP/SDES IPv6 Internetworking



SRTP/SDES IPv6 to RTP IPv4 Internetworking



SRTP/SDES IPv6 to RTP IPv6 Internetworking

## Configuration

IPv6 support must be globally enabled to support SRTP internetworking as described above. If IPv6 is currently enabled, no additional configuration is required.

## Key Exchange Protocols

Key exchange protocols enable secure communications over an untrusted network by deriving and distribution shared keys between two or more parties. The Internet Key Exchange (IKEv1) Protocol, originally defined in RFC 2409, provides a method for creating keys used by IPsec tunnels. MIKEY defined in RFC 3880, and Session Description Protocol Security Descriptions for Media Streams (SDES), defined in RFC 4568, provide alternative methods for creating keys used to encrypt Real-time Transport Protocol (RTP) and Real-time Transport Control Protocol (RTCP) transactions.

Each of these protocols is described in the following sections.

### IKEv1 Protocol

IKEv1 is specified by a series of RFCs, specifically RFCs 2401 through 2412. The most relevant are:

- RFC 2407, *The Internet IP Security Domain of Interpretation for ISAKMP*
- RFC 2408, *Internet Security Association and Key Management Protocol (ISAKMP)*
- RFC 2409, *The Internet Key Exchange (IKE)*
- RFC 2412, *Oakley Key Determination Protocol*

IKEv1 combines features of the Internet Security Association and Key Management Protocol (ISAKMP) and Oakley Key Determination Protocol in order to negotiate Security Associations (SA) for two communicating peers. IKEv1 also provides for key agreement using Diffie-Hellman.

IKEv1 uses two phases. Phase 1 is used to establish an ISAKMP Security Association for IKEv1 itself. Phase 1 negotiates the authentication method and symmetric encryption algorithm to be used. Phase 1 requires either six messages (main mode) or three messages (aggressive mode).

Phase 2 negotiates the SA for two IPsec peers and is accomplished with three messages.

The initial IKEv1 implementation supports RFC 2409, *Internet Key Exchange*, and RFC 3706, *A Traffic-Based Method of Detecting Dead Internet Key Exchange (IKE) Peers*.

### IKEv1 Configuration

IKEv1 configuration consists of five steps.

1. Configure IKEv1 global parameters.
2. Optionally, enable and configure Dead Peer Detection (DPD) Protocol.
3. Configure IKEv1 interfaces.
4. Configure IKEv1 Security Associations (SA).
5. Assign the IKEv1 SA to an IPsec Security Policy.

#### IKEv1 Global Configuration

**To configure global IKEv1 parameters:**

1. From superuser mode, use the following command sequence to access ike-config configuration mode. While in this mode, you configure global IKEv1 configuration parameters.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# ike
ACMEPACKET(ike)# ike-config
ACMEPACKET(ike-config)#
```

2. Use the **ike-version** parameter to specify IKEv1.

Use 1 to specify IKEv1 operations.

3. Use the **log-level** parameter to specify the contents of the IKEv1 log.

Events are listed below in descending order of criticality.

- emergency (most critical)
- critical
- major
- minor
- warning
- notice
- info (least critical — the default)
- trace (test/debug, not used in production environments)

- debug (test/debug, not used in production environments)
- detail (test/debug, not used in production environments)

In the absence of an explicitly configured value, the default value of info is used.

4. Use the optional **udp-port** parameter to specify the port monitored for IKEv1 protocol traffic.

In the absence of an explicitly configured value, the default port number of 500 is used.

5. Use the optional **negotiation-timeout** parameter to specify the maximum interval (in seconds) between Diffie-Hellman message exchanges.

In the absence of an explicitly configured value, the default specifies a 15 second timeout value.

6. Use the optional **event-timeout** parameter to specify the maximum time (in seconds) allowed for the duration of an IKEv1 event, defined as the successful establishment of an IKE or IPsec Security Association (SA).

In the absence of an explicitly configured value, the default specifies a 60 second time span.

7. Use the optional **phase1-mode** parameter to specify the IKE Phase 1 exchange mode.

During Phase 1 the IKE initiator and responder establish the IKE SA, using one of two available methods.

main mode — (the default) is more verbose, but provides greater security in that it does not reveal the identity of the IKE peers. Main mode requires six messages (3 requests and corresponding responses) to (1) negotiate the IKE SA, (2) perform a Diffie-Hellman exchange of cryptographic material, and (3) authenticate the remote peer.

aggressive mode — is less verbose (requiring only three messages), but less secure in providing no identity protection, and less flexible in IKE SA negotiation.

In the absence of an explicitly configured value, the default (main mode) is used.

8. Use the optional **phase1-dh-mode** parameter to specify the Diffie-Hellman Group used during IKE Phase 1 negotiation.

dh-group1 — as initiator, propose Diffie-Hellman group 1 (768-bit primes, less secure)

dh-group2 — as initiator, propose Diffie-Hellman group 2 (1024-bit primes, more secure)

first-supported — (the default) as responder, use the first supported Diffie-Hellman group proposed by initiator

9. If functioning as the IKE initiator, use the optional **phase1-life-seconds** parameter to specify the proposed lifetime (in seconds) for the IKE SA established during IKE Phase 1 negotiations.

Allowable values are within the range 1 through 999999999 (seconds) with a default of 3600 (1 hour).

This parameter can safely be ignored if functioning as a IKE responder.

10. If functioning as the IKE responder, use the optional **phase1-life-seconds-max** parameter to specify the maximum time (in seconds) accepted for IKE SA lifetime during IKE Phase 1 negotiations.

Allowable values are within the range 1 through 999999999 (seconds) with a default of 86400 (1 day).

This parameter can safely be ignored if functioning as a IKE initiator.

11. If functioning as the IKE initiator, use the optional **phase2-life-seconds** parameter to specify the proposed lifetime (in seconds) for an IPsec SA established during IKE Phase 2 negotiations.

Allowable values are within the range 1 through 999999999 (seconds) with a default of 28800 (8 hours).

This parameter can safely be ignored if functioning as a IKE responder.

12. If functioning as the IKE responder, use the optional **phase2-life-seconds-max** parameter to specify the maximum time (in seconds) accepted for IPsec SA lifetime during IKE Phase 2 negotiations.

Allowable values are within the range 1 through 999999999 (seconds) with a default of 86400 (1 day).


This parameter can safely be ignored if functioning as a IKE initiator.

13. Use the optional **phase2-exchange-mode** parameter to specify the Diffie-Hellman group used in Phase 2 negotiations.

dh-group1 — use Diffie-Hellman group 1 (768-bit primes, less secure)

`dh-group2` — use Diffie-Hellman group 2 (1024-bit primes, more secure)

`no-forward-secrecy` — use the same key as used during Phase 1 negotiation

 **Note:** Forward security indicates that compromise of a single key permits access only to data encrypted with that specific key. Failure to generate a new key for IKE Phase 2 potentially compromises additional data.

`phase1-group` — (the default) use the same Diffie-Hellman group as used during Phase 1 negotiation

14. Use the **`shared-password`** parameter to specify the PSK (pre-shared key) used during authentication with the remote IKE peer.

The PSK is a string of ACSII printable characters no longer than 255 characters (not displayed by the ACLI).

This global PSK can be over-ridden by an interface-specific PSK.

15. Use the optional **`dpd-time-interval`** parameter to specify the maximum period of inactivity before the DPD protocol is initiated on a specific endpoint.

Allowable values are within the range 1 through 999999999 (seconds) with a default of 0.

The default value, 0, disables the DPD protocol; setting this parameter to a non-zero value globally enables the protocol and sets the inactivity timer.

16. Use **`done`**, **`exit`**, and **`verify-config`** to complete configuration of IKEv1 global parameters instance.

### DPD Protocol Configuration

If you enabled the DPD protocol with the **`dpd-time-interval`** parameter, use the following procedure to create a DPD template, an operational set of DPD parameters, that you subsequently assign to one or more IKEv1 interfaces.

#### To configure DPD parameters:

1. From superuser mode, use the following command sequence to access `dpd-params` configuration mode. While in this mode, you configure DPD templates.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# ike
ACMEPACKET(ike)# dpd-params
ACMEPACKET(dpd-params)#
```

2. Use the required **`name`** parameter to provide a unique identifier for this `dpd-params` instance.

**`name`** enables the creation of multiple `dpd-params` instances.

3. Use the **`max-loop`** parameter to specify the maximum number DPD peers examined every `dpd-interval`, whose value is established during IKv1 global configuration.

If CPU workload surpasses the threshold set by **`max-cpu-limit`**, this value is over-ridden by **`load-max-loop`**.

Allowable values are within the range 1 through 999999999 (endpoints) with a default of 100.

4. Use the **`max-endpoints`** parameter to specify the maximum number of simultaneous DPD protocol negotiations supported when the CPU is not under load (as specified by the **`max-cpu-limit`** property).

If CPU workload surpasses the threshold set by **`max-cpu-limit`**, this value is over-ridden by **`load-max-endpoints`**.

Allowable values are within the range 1 through 999999999 (endpoints) with a default of 25.

5. Use the **`max-cpu-limit`** parameter to specify a threshold value (expressed as a percentage of CPU capacity) at which DPD protocol operations are minimized to conserve CPU resources.

Allowable values are within the range 0, which effectively disables DPD operations, through 100 (percent) with a default of 60.



6. Use the **load-max-loop** parameter to specify the maximum number of endpoints examined every **dpd-time-interval** when the CPU is under load, as specified by the **max-cpu-limit** parameter.  
Allowable values are within the range 1 through 999999999 (endpoints) with a default of 40. Ensure that the configured value is less than the value assigned to **max-loop**.
7. Use the **load-max-endpoints** parameter to specify the maximum number of simultaneous DPD Protocol negotiations supported when the CPU is under load, as specified by the **max-cpu-limit** property.  
Allowable values are within the range 1 through 999999999 (endpoints) with a default of 5. Ensure that the configured value is less than the value assigned to **max-endpoints**.
8. Use **done**, **exit**, and **verify-config** to complete configuration of the DPD template instance.
9. Repeat Steps 1 through 8 to configure additional DPD templates.

## IKEv1 Interface Configuration

To configure IKEv1 interface parameters:

1. From superuser mode, use the following command sequence to access ike-config configuration mode. While in this mode, you configure IKEv1 interface parameters.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# ike
ACMEPACKET(ike)# ike-interface
ACMEPACKET(ike-interface)#
```

2. Use the **address** parameter to specify the IPv4 address of the interface.
3. Use the **realm-id** parameter to specify the realm that contains the IP address assigned to this IKEv1 interface.
4. Use the **ike-mode** parameter to specify the operational mode, either responder (the default) or initiator.
5. If DPD has been enabled at the global level, use the **dpd-params-name** parameter to assign a DPD template, an operational set of DPD parameters, to the current IKEv1 interface.

If DPD has not been enabled, this parameter can be safely ignored.

6. Use the optional **shared-password** parameter to assign an interface PSK.

This IKEv1-interface-specific value over-rides the global default value set at the IKE configuration level.

7. Use **done**, **exit**, and **verify-config** to complete configuration of IKEv1 interface.
8. Repeat Steps 1 through 7 to configure additional IKEv1 interfaces.

## IKEv1 Security Association Configuration

An IKEv1 SA identifies cryptographic material available for IPsec tunnel establishment.

To configure IKEv1 SA parameters:

1. From superuser mode, use the following command sequence to access ike-sainfo configuration mode. While in this mode, you configure global IKEv1 SAs.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# ike
ACMEPACKET(ike)# ike-sainfo
ACMEPACKET(ike-sainfo)#
```

2. Use the required **name** parameter to provide a unique identifier for this ike-sainfo instance.

**name** enables the creation of multiple ike-sainfo instances.

3. Use the **security-protocol** parameter to specify the IPsec security (authentication and encryption) protocols supported by this SA.

The following security protocols are available.

Authentication Header (AH) — the default value — as defined by RFC 4302, *IP Authentication Header*, which provides authentication integrity to include the mutual identification of remote peers, non-repudiation of received traffic, detection of data that has been altered in transit, and detection of data that has been replayed, that is copied and then re-injected into the data stream at a later time. Authentication services utilize the authentication algorithm specified by the **auth-algo** property.

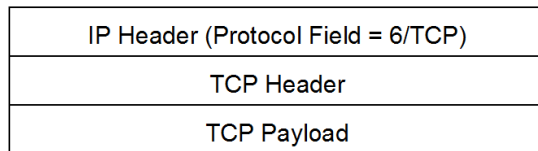
Encapsulating Security Payload (ESP) as defined by RFC 4303, *IP Encapsulating Security Payload*, which provides both authentication and privacy services. Privacy services utilize the encryption algorithm specified by the **encryption-algo** property.

ESP-AUTH (also RFC 4303-based), which supports ESP's optional authentication.

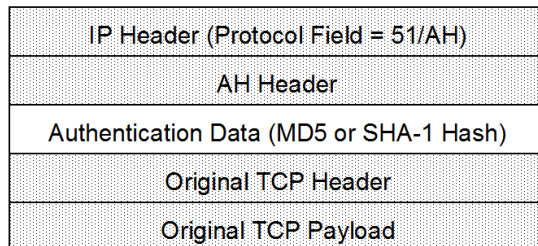
ESP-NUL (also RFC 4303-based) which proves NULL encryption as described in RFC 2410, *The NULL Encryption Algorithm and Its Use With IPsec*. This option provides no privacy services, and is not recommended for production environments.

Refer to the following figures for additional details.

### Original IP Datagram



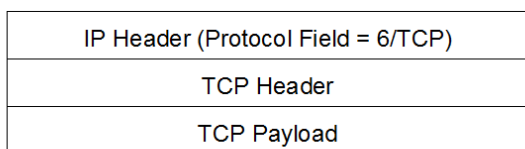
### AH Encapsulated Datagram



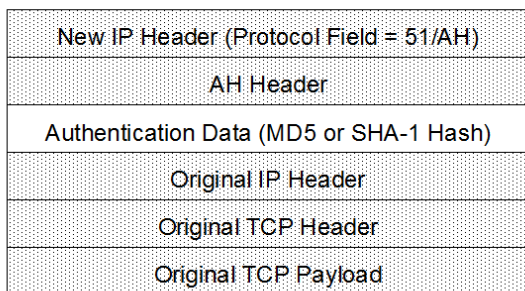
Authenticated data, note that TOS, Flags, Fragmentation, TTL, and Header Checksum fields of the IP Header are not covered by the authentication calculation.

AH Transport Mode

### Original IP Datagram



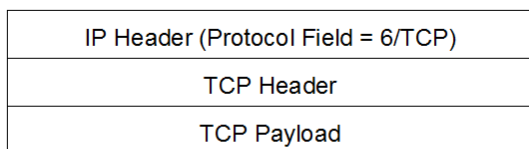
### AH Encapsulated Datagram



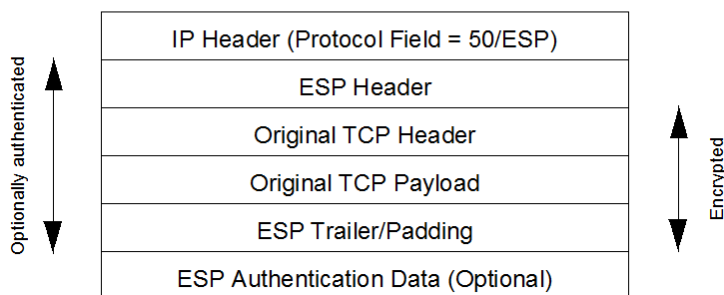
Authenticated data, note that TOS, Flags, Fragmentation, TTL, and Header Checksum fields of the IP Header are not covered by the authentication calculation.

#### AH Tunnel Mode

### Original IP Datagram

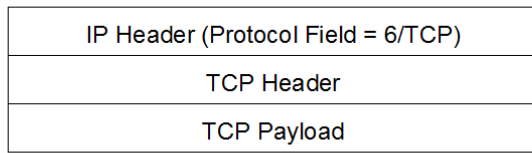


### ESP Encapsulated Datagram

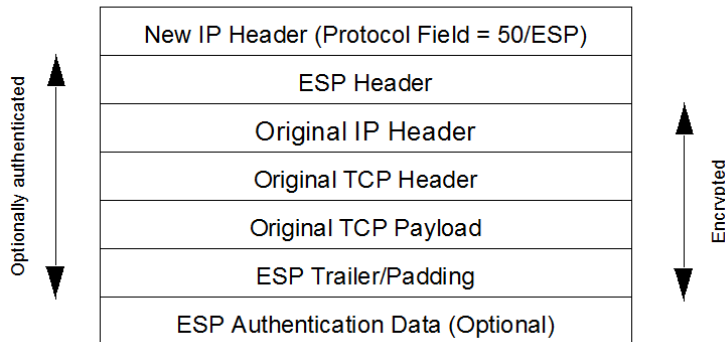


#### ESP Transport Mode

Original IP Datagram



ESP Encapsulated Datagram



ESP Tunnel Mode

```
ACMEPACKET (ike-sainfo) # security-protocol esp
ACMEPACKET (ike-sainfo) #
```

- Use the **auth-algo** parameter to specify the authentication algorithms supported by this SA.

The following authentication protocols are available

Message Digest Algorithm 5 (md5) — as defined by RFC 1321, *The MD5 Message-Digest Algorithm*.

Secure Hash Algorithm (sha) — as defined by RFC 3174, *Secure Hash Standard*.

any (the default) — supports both MD5 and SHA-1.

```
ACMEPACKET (ike-sainfo) # auth-algo md5
ACMEPACKET (ike-sainfo) #
```

- Use the **encryption-algo** parameter to specify the encryption algorithms supported by this SA.

The following encryption protocols are available

Triple DES (3des) — as defined by ANSI X.9.52 1998, *Triple Data Encryption Algorithm Modes of Operation*.

Advanced Encryption Standard (aes) — as defined by RFC 3565, *Advanced Encryption Standard*.

NULL Encryption (null) — as described in RFC 2410, *The NULL Encryption Algorithm and Its Use With IPsec*.

This option provides no privacy services, and is not recommended for production environments.

any (the default) — supports all listed encryption protocols.

```
ACMEPACKET (ike-sainfo) # encryption-algo aes
ACMEPACKET (ike-sainfo) #
```

- Use the **ipsec-mode** parameter to specify the IPsec operational mode.

Transport mode (the default) provides a secure end-to-end connection between two IP hosts. Transport mode encapsulates the IP payload.

Tunnel mode provides VPN service where entire IP packets are encapsulated within an outer IP envelope and delivered from source (an IP host) to destination (generally a secure gateway) across an untrusted internet.

Refer to the previous figures for encapsulation details.

```
ACMEPACKET(ike-sainfo) # ipsec-mode tunnel
ACMEPACKET(ike-sainfo) #
```

- If **ipsec-mode** is tunnel, use the required **tunnel-local-addr** parameter to specify the IP address of the local IKEv1 interface that terminates the IPsec tunnel.

This parameter can safely be ignored if **ipsec-mode** is transport.

```
ACMEPACKET(ike-sainfo) # tunnel-local-addr 192.169.204.14
ACMEPACKET(ike-sainfo) #
```

- If **ipsec-mode** is tunnel, use the **tunnel-remote-addr** parameter to specify the IP address of the remote IKEv1 peer that terminates the IPsec tunnel.

Provide the remote IP address, or use the default wild-card value (\*) to match all IP addresses.

This parameter can safely be ignored if **ipsec-mode** is transport.

```
ACMEPACKET(ike-sainfo) # tunnel-remote-addr *
ACMEPACKET(ike-sainfo) #
```

- Use **done**, **exit**, and **verify-config** to complete configuration of IKEv1 SA.

- Repeat Steps 1 through 9 to configure additional IKEv1 SAs.

### IPsec Security Policy Configuration

Use the following procedure to assign an IKEv1 SA to an existing IPsec Security Policy. Note that the network interface supported by the IPsec Security Policy must have been configured as an IKEv1 interface.

- From superuser mode, use the following command sequence to access ike-config configuration mode. While in this mode, you configure global IKEv1 configuration parameters.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) # security
ACMEPACKET(security) # ipsec
ACMEPACKET(ipsec) # security-policy
ACMEPACKET(security-policy) #
```

- Use the required **ike-sainfo-name** parameter to assign an IKv1 SA to this IPsec Security Policy.
- Use **done**, **exit**, and **verify-config** to complete configuration of IPsec Security Policy.

## SDP Session Description Protocol

The Secure Real-time Transport Protocol, as described in RFC 3711, *The Secure Real-time Transport Protocol (SRTP)*, provides a framework for the encryption and authentication of Real-time Transport Protocol (RTP) and RTP Control Protocol (RTCP) streams. Both RTP and RTCP are defined by RFC 3550, *RTP: A Transport Protocol for Real-Time Applications*.

Encryption ensures that the call content and associated signalling remains private during transmission. Authentication ensures that (1) received packets are from the purported source, (2) packets are not been tampered with during transmission, and (3) a packet has not been replayed by a malicious server.

### Protocol Overview

While the RFC 3711 framework provides encryption and authentication procedures and defines a set of default cryptographic transforms required for RFC compliance, it does not specify a key management protocol to securely derive and exchange cryptographic keys. RFC4568, *Session Description Protocol (SDP) Security Description for Media Streams*, defines such a protocol specifically designed to exchange cryptographic material using a newly defined SDP crypto attribute. Cryptographic parameters are established with only a single message or in single round-trip exchange using the offer/answer model defined in RFC 3264, *An Offer/Answer Model with the Session Description Protocol*.

Release S-C6.2.0 provides support for an initial SDP Security Descriptions (SDS) implementation that generates keys used to encrypt SRTP/SRTCP packets. Authentication of packets will be added to a subsequent release.

A sample SDP exchange is shown below:

The SDP offerer sends:

```
v=0
o=sam 2890844526 2890842807 IN IP4 10.47.16.5
s=SRTP Discussion
i=A discussion of Secure RTP
u=http://www.example.com/seminars/srtp.pdf
e=marge@example.com (Marge Simpson)
c=IN IP4 168.2.17.12
t=2873397496 2873404696
m=audio 49170 RTP/SAVP 0
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:WVnfX19zZWljdGwgKCkgewkyMjA7fQp9CnVubGVz|2^20|1:4
```

The SDP answerer replies:

```
v=0
o=sam 2890844526 2890842807 IN IP4 10.47.16.5
s=SRTP Discussion
i=A discussion of Secure RTP
u=http://www.example.com/seminars/srtp.pdf
e=marge@example.com (Marge Simpson)
c=IN IP4 168.2.17.12
t=2873397496 2873404696
m=audio 49170 RTP/SAVP 0
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:WVnfX19zZWljdGwgKCkgewkyMjA7fQp9CnVubGVz|2^20|1:4
```

The media-level SDP attribute, `crypto`, describes the cryptographic suite, key parameters, and session parameters for the preceding unicast media line. The `crypto` attribute takes the form:

```
a=crypto: tag crypto-suite key-parameter [session-parameters]
```

`tag`

The `tag` field contains a decimal number that identifies a specific attribute instance. When an offer contains multiple `crypto` attributes, the answer uses the `tag` value to identify the accepted offer.

In the sample offer the `tag` value is 1.

`crypto-suite`

The `crypto-suite` field contains the encryption and authentication algorithms, either `AES_CM_128_HMAC_SHA1_80` or `AES_CM_128_HMAC_SHA1_32`.

The `key-parameter` field contains one or more sets of keying material for the selected `crypto-suite` and it has following format.

```
"inline:" <key||salt> ["|" lifetime] ["|" MKI ":" length]
```

`inline` is a method and specifies that the `crypto` material to be used by the offerer is transmitted via the SDP.

The `key||salt` field contains a base64-encoded concatenated master key and salt.

Assuming the offer is accepted, the `key || salt` provides the `crypto` material used by the offerer to encrypt SRTP/SRTCP packets, and used by the answerer to decrypt SRTP/SRTCP packets.

Conversely the `key || salt` contained in the answer to the offer provides the `crypto` material used by the answerer to encrypt SRTP/SRTCP packets, and used by the offerer to decrypt SRTP/SRTCP packets.

The `lifetime` field optionally contains the master key lifetime (maximum number of SRTP or SRTCP packets encoded using this master key).

In the sample offer the `lifetime` value is 1,048, 576 (220) packets.

The `MKI:length` field optionally contains the Master Key Index (MKI) value and the MKI length.

The MKI is used only when the offer contains multiple keys; it provides a means to differentiate one key from another. The MKI takes the form of an integer, followed by its byte length when included in SRTP/SRTCP packets.

In the sample offer the MKI value is 1 with a length of 4 bytes.

The session-parameters field contains a set of optional parameters that may override SRTP session defaults for the SRTP and SRTCP streams.

UNENCRYPTED\_SRTP — SRTP messages are not encrypted

UNENCRYPTED\_SRTCP — SRTCP messages are not encrypted

UNAUTHENTICATED\_SRTP — SRTP messages are not authenticated

When generating an initial offer, the offerer must ensure that there is at least one crypto attribute for each media stream for which security is desired. Each crypto attribute for a given media stream must contain a unique tag. The ordering of multiple crypto attributes is significant — the most preferred crypto suite is listed first.

Upon receiving the initial offer, the answerer must either accept one of the offered crypto attributes, or reject the offer in its entirety.

When an offered crypto attribute is accepted, the crypto attribute in the answer **MUST** contain the tag and crypto-suite from the accepted crypto attribute in the offer, and the key(s) the answerer will be using for media sent to the offerer.

The crypto-suite is bidirectional and specifies encryption and authentication algorithms for both ends of the connection. The keys are unidirectional in that one key or key set encrypts and decrypts traffic originated by the offerer, while the other key or key set encrypts and decrypts traffic originated by the answerer. The use of symmetric keying, where the same key is used for both encryption and decryption, mandates the key exchange between the offerer and the answerer.

Key exchange via text-based SDP is unacceptable in that malicious network elements could easily eavesdrop and obtain the plaintext keys, thus compromising the privacy and integrity of the encrypted media stream. Consequently, the SDP exchange must be protected by a security protocol such as IPsec or TLS.

## Licensing and Hardware Requirements

SRTP/SRTCP support requires the presence of an IPsec NIU and an SSM/SSM2 (Security Service Module).

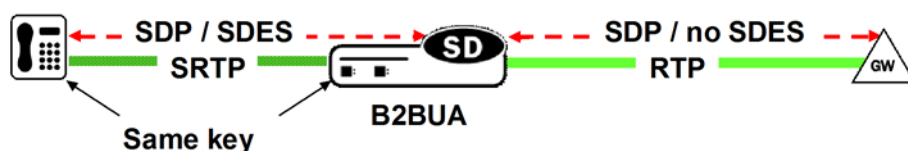
No additional licences are required.

## Operational Modes

SRTP topologies can be reduced to three basic topologies which are described in the following sections.

### Single-Ended SRTP Termination

Single-ended SRTP termination is illustrated in the following figure.



#### Single-Ended SRTP Termination

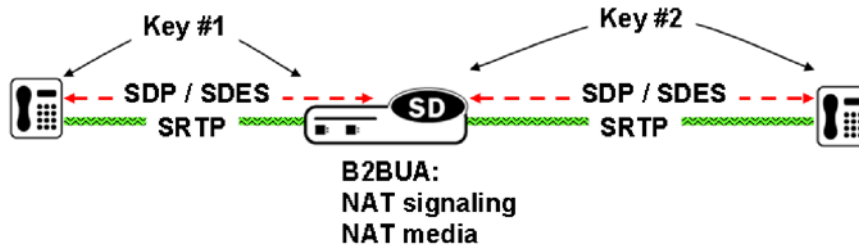
If SRTP is enabled for the inbound realm/interface, the Oracle Communications Session Border Controller handles the incoming call as specified by the Media Security Policy assigned to the inbound realm. If there is a crypto attribute contained in the offer, the Oracle Communications Session Border Controller parses the crypto attributes and optional parameters, if any. If the offer contains a crypto attribute or attributes compatible with the requirements specified by the SDES profile assigned to the Media Security policy, it selects the most preferred compatible attribute. Otherwise, the Oracle Communications Session Border Controller rejects the offer. Before the SDP is forwarded to the called party, the Oracle Communications Session Border Controller allocates resources, establishes SRTP and SRTCP

Security Associations and updates the SDP by removing the crypto attribute and inserting possibly NAT'ed media addresses and ports. At the same time, the original crypto attribute is also removed from the SDP.

Once the reply from the called party is received, the Oracle Communications Session Border Controller inserts appropriate crypto attribute(s) to form a new SDP, and forward the response back to the calling party.

### Back-to-Back SRTP Termination

Back-to-back SRTP termination is illustrated in the following figure.

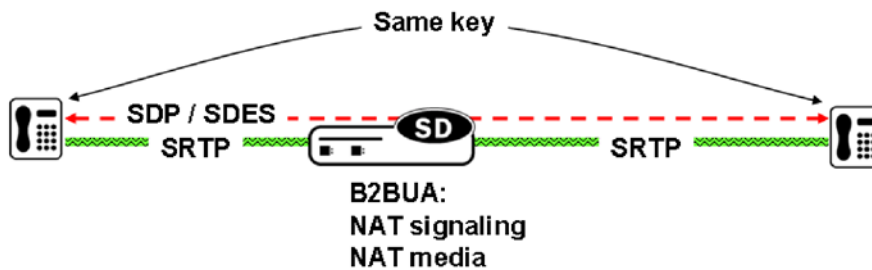


Back-to-Back SRTP Termination

Initial processing is similar to the single-ended termination described above. Before forwarding the request to the called party, the Oracle Communications Session Border Controller replaces the original crypto attribute with a new one whose crypto attribute conforms to the media security policy for the outbound realm/interface. Upon receiving the answer from the called party, the Oracle Communications Session Border Controller accepts or rejects it, again based upon conformity to the media security policy. If accepted, the Oracle Communications Session Border Controller replaces the original crypto attribute from the called party with its own to form a new SDP, which it forwards back to the calling party. At this point, SRTP media sessions are established on both sides for both calling and called parties.

### SRTP Pass-Thru

SRTP pass-thru is illustrated in the following figure.



SRTP Pass-Thru

If the media security policy specifies pass-through mode, the Oracle Communications Session Border Controller does not alter the crypto attribute exchange between the calling and the called party; the attribute is transparently passed.

## SDES Configuration

SDES configuration consists of the following steps.

1. Create one or more SDES profiles which specify parameter values negotiated during the offer/answer exchange.
2. Create one or more Media Security Policies that specify key exchange protocols and protocol-specific profiles.
3. Assign a Media Security Policy to a realm.
4. Create an interface-specific Security Policy.



## SDES Profile Configuration

An SDES profile specifies the parameter values offered or accepted during SDES negotiation.

**To configure SDES profile parameters:**


1. From superuser mode, use the following command sequence to access sdes-profile configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# media-security
ACMEPACKET(media-security)# sdes-profile
ACMEPACKET(sdes-profile)#
```

2. Use the required **name** parameter to provide a unique identifier for this sdes-profile instance.

**name** enables the creation of multiple sdes-profile instances.

3. Use the **crypto-suite** parameter to select the encryption and authentication algorithms accepted or offered by this sdes-profile.

 **Note:** SRTP authentication is not currently supported.

Allowable values are:

AES\_CM\_128\_HMAC\_SHA1\_80 (the default value)

supports AES/128 bit key for encryption and HMAC/SHA-1 80-bit digest for authentication

AES\_CM\_128\_HMAC\_SHA1\_32

supports AES/128 bit key for encryption and HMAC/SHA-1 32-bit digest for authentication

4. Because SRTP authentication is not currently supported, ignore the **srtp-auth** parameter.
5. Use the **srtp-encrypt** parameter to enable or disable the encryption of RTP packets.

With encryption enabled, the default condition, the Oracle Communications Session Border Controller offers RTP encryption, and rejects an answer that contains an UNENCRYPTED\_SRTP session parameter in the crypto attribute.

With encryption disabled, the Oracle Communications Session Border Controller does not offer RTP encryption and includes an UNENCRYPTED\_SRTP session parameter in the SDP crypto attribute; it accepts an answer that contains an UNENCRYPTED\_SRTP session parameter.

6. Use the **srtpc-encrypt** parameter to enable or disable the encryption of RTCP packets.

With encryption enabled, the default condition, the Oracle Communications Session Border Controller offers RTCP encryption, and rejects an answer that contains an UNENCRYPTED\_SRTCP session parameter in the crypto attribute.

With encryption disabled, the Oracle Communications Session Border Controller does not offer RTCP encryption and includes an UNENCRYPTED\_SRTCP session parameter in the SDP crypto attribute; it accepts an answer that contains an UNENCRYPTED\_SRTCP session parameter.

7. Use the **mki** parameter to enable or disable the inclusion of the MKI:length field in the SDP crypto attribute.

The master key identifier (MKI) is an optional field within the SDP crypto attribute that differentiates one key from another. MKI is expressed as a pair of decimal numbers in the form: |mki:mki\_length| where mki is the MKI integer value and mki\_length is the length of the MKI field in bytes.

The MKI field is necessary only in topologies that may offer multiple keys within the crypto attribute.

Allowable values are enabled and disabled (the default).

enabled – an MKI field is sent within the crypto attribute (16 bytes maximum)

disabled – no MKI field is sent

8. Use **done**, **exit**, and **verify-config** to complete configuration of this SDES profile instance.

9. Repeat Steps 1 through 8 to configure additional SDES profiles.

### Media Security Policy Configuration

Use the following procedure to create a Media Security Policy that specifies the role of the Oracle Communications Session Border Controller in the security negotiation. If the Oracle Communications Session Border Controller takes part in the negotiation, the policy specifies a key exchange protocol and SDES profile for both incoming and outgoing calls.

To configure media-security-policy parameters:

1. From superuser mode, use the following command sequence to access media-sec-policy configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# media-security
ACMEPACKET(media-security)# media-sec-policy
ACMEPACKET(media-sec-policy)#
```

2. Use the required **name** parameter to provide a unique identifier for this media-sec-policy instance.

**name** enables the creation of multiple media-sec-policy instances.

3. Use optional **pass-thru** parameter to enable or disable pass-thru mode.

With pass-thru mode enabled, the User Agent (UA) endpoints negotiate security parameters between each other; consequently, the Oracle Communications Session Border Controller simply passes SRTP traffic between the two endpoints.

With pass-thru mode disabled (the default state), the Oracle Communications Session Border Controller disallows end-to-end negotiation — rather the Oracle Communications Session Border Controller initiates and terminates SRTP tunnels with both endpoints.

4. Use the **outbound** navigation command to move to media-sec-outbound configuration mode. While in this configuration mode you specify security parameters applied to the outbound call leg, that is calls sent by the Oracle Communications Session Border Controller.
5. Use the **protocol** parameter to select the key exchange protocol.

Allowable values are mikey and sdes.

Select sdes for SDES key exchange.

6. Use the **profile** parameter to specify the name of the SDES profile applied to calls sent by the Oracle Communications Session Border Controller.

7. Use the **mode** parameter to select the real time transport protocol.

Allowable values are rtp and srtp (the default).

**mode** identifies the transport protocol (RTP or SRTP) included in an SDP offer when this media-security-policy is in effect.

8. Use the **done** and **exit** parameters to return to media-sec-policy configuration mode.
9. Use the **inbound** navigation command to move to media-sec-inbound configuration mode. While in this configuration mode you specify security parameters applied to the inbound call leg, that is calls received by the Oracle Communications Session Border Controller.

10. Use the **protocol** parameter to select the key exchange protocol.

Allowable values are mikey and sdes.

Select sdes for SDES.

11. Use the **profile** parameter to specify the name of the SDES profile applied to calls received by the Oracle Communications Session Border Controller.

12. Use the **mode** parameter to select the real time transport protocol.

Allowable values are rtp and srtp (the default).

**mode** identifies the transport protocol (RTP or SRTP) accepted in an SDP offer when this media-security-policy is in effect.

13. Use **done**, **exit**, and **verify-config** to complete configuration of this media security policy instance.
14. Repeat Steps 1 through 13 to configure additional media-security policies.

### Assign the Media Security Policy to a Realm

To assign a media-security-policy to a realm:

1. From superuser mode, use the following command sequence to access realm-config configuration mode. While in this mode, you assign an existing media-security-policy to an existing realm.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)# select
identifier:
1. access-12
...
...
selection: 1
ACMEPACKET(realm-config)#
```

2. Use the **media-sec-policy** parameter to assign the policy to the target realm.
3. Use **done**, **exit**, and **verify-config** to complete assignment of the media-security-policy to the realm.

## ACLI Example Configurations

The following section contain relevant sections of system configurations for basic operational modes.

### Single-Ended SRTP Termination Configuration

```
ACMEPACKET# show running-config
...
...
...
sdes-profile
  name                sdes1
  crypto-list         AES_CM_128_HMAC_SHA1_80
  srtp-auth           enabled
  srtp-encrypt        enabled
  srtcp-encrypt       enabled
  mki                 disabled
  key
  salt
  last-modified-by    admin@console
  last-modified-date  2009-11-16 15:37:13
media-sec-policy
  name                msp2
  pass-through        disabled
  inbound
    profile           sdes1
    mode              srtp
    protocol          sdes
  outbound
    profile           sdes1
    mode              srtp
    protocol          sdes
  last-modified-by    admin@console
  last-modified-date  2009-11-16 15:37:51
...
...
...
```

```
realm-config
  identifier                peer
  description
  addr-prefix                192.168.0.0/16
  network-interfaces        M00:0
  mm-in-realm                enabled
  mm-in-network              enabled
  mm-same-ip                 enabled
  mm-in-system               enabled
  bw-cac-non-mm              disabled
  msm-release                disabled
  qos-enable                  disabled
  generate-UDP-checksum      disabled
  max-bandwidth              0
  fallback-bandwidth         0
  max-priority-bandwidth     0
  max-latency                 0
  max-jitter                  0
  max-packet-loss            0
  observ-window-size         0
  parent-realm
  dns-realm
  media-policy
  media-sec-policy           msp2
  in-translationid
...
...
...
  last-modified-by          admin@console
  last-modified-date        2009-11-10 15:38:19
```

### Back-to-Back SRTP Termination Configuration

```
ACMEPACKET# show running-config
...
...
...
sdes-profile
  name                       sdes1
  crypto-list                 AES_CM_128_HMAC_SHA1_80
  srtp-auth                    enabled
  srtp-encrypt                 enabled
  srtcp-encrypt                enabled
  mki                          disabled
  key
  salt
  last-modified-by            admin@console
  last-modified-date          2009-11-16 15:37:13
media-sec-policy
  name                         msp2
  pass-through                 disabled
  inbound
    profile                    sdes1
    mode                        srtp
    protocol                    sdes
  outbound
    profile                    sdes1
    mode                        srtp
    protocol                    sdes
  last-modified-by            admin@console
  last-modified-date          2009-11-16 15:37:51
...
...
...
```

```

realm-config
  identifier                peer
  description
  addr-prefix                192.168.0.0/16
  network-interfaces        M00:0
  mm-in-realm                enabled
  mm-in-network              enabled
mm-same-ip                   enabled
  mm-in-system                enabled
  bw-cac-non-mm              disabled
  msm-release                 disabled
  qos-enable                  disabled
  generate-UDP-checksum      disabled
  max-bandwidth               0
  fallback-bandwidth         0
  max-priority-bandwidth     0
  max-latency                 0
  max-jitter                  0
  max-packet-loss            0
  observ-window-size         0
  parent-realm
  dns-realm
  media-policy
  media-sec-policy           msp2
...
...
...
realm-config
  identifier                core
  description
  addr-prefix                172.16.0.0/16
  network-interfaces        M10:0
  mm-in-realm                enabled
  mm-in-network              enabled
  mm-same-ip                   enabled
  mm-in-system                enabled
  bw-cac-non-mm              disabled
  msm-release                 disabled
  qos-enable                  disabled
  generate-UDP-checksum      disabled
  max-bandwidth               0
  fallback-bandwidth         0
  max-priority-bandwidth     0
  max-latency                 0
  max-jitter                  0
  max-packet-loss            0
  observ-window-size         0
  parent-realm
  dns-realm
  media-policy
  media-sec-policy           msp2
  in-translationid
...
...
...
  last-modified-by          admin@console
  last-modified-date        2009-11-10 15:38:19

```

### SRTP Pass-Thru Configuration

```

ACMEPACKET# show running-config
...
...
...

```

```

sdes-profile
  name                sdes1
  crypto-list         AES_CM_128_HMAC_SHA1_80
  srtp-auth           enabled
  srtp-encrypt        enabled
  srtcp-encrypt       enabled
  mki                 disabled
  key
  salt
  last-modified-by    admin@console
  last-modified-date  2009-11-16 15:37:13
media-sec-policy
  name                msp2
  pass-through        enabled
  inbound
    profile           sdes1
    mode              srtp
    protocol          sdes
  outbound
    profile           sdes1
    mode              srtp
    protocol          sdes
  last-modified-by    admin@console
  last-modified-date  2009-11-16 15:37:51
...
...
...
realm-config
  identifier           peer
  description
  addr-prefix         192.168.0.0/16
  network-interfaces  M00:0
  mm-in-realm         enabled
  mm-in-network       enabled
  mm-same-ip          enabled
  mm-in-system        enabled
  bw-cac-non-mm      disabled
  msm-release         disabled
  qos-enable          disabled
  generate-UDP-checksum disabled
  max-bandwidth       0
  fallback-bandwidth  0
  max-priority-bandwidth 0
  max-latency         0
  max-jitter          0
  max-packet-loss     0
  observ-window-size  0
  parent-realm
  dns-realm
  media-policy
  media-sec-policy    msp2
...
...
...
realm-config
  identifier           core
  description
  addr-prefix         172.16.0.0/16
  network-interfaces  M10:0
  mm-in-realm         enabled
  mm-in-network       enabled
  mm-same-ip          enabled
  mm-in-system        enabled
  bw-cac-non-mm      disabled

```

```

msm-release                disabled
qos-enable                 disabled
generate-UDP-checksum     disabled
max-bandwidth              0
fallback-bandwidth        0
max-priority-bandwidth    0
max-latency                0
max-jitter                 0
max-packet-loss           0
observ-window-size        0
parent-realm
dns-realm
media-policy
media-sec-policy           msp2
in-translationid
...
...
...
last-modified-by          admin@console
last-modified-date        2009-11-10 15:38:19

```

## Security Policy

A Security Policy enables the Oracle Communications Session Border Controller to identify inbound and outbound media streams that are treated as SRTP/SRTCP. The high-priority Security Policy, p1, (shown below) allows signaling traffic from source 172.16.1.3 to destination 172.16.1.10:5060. The lower-priority Security Policy, p2, (also shown below) matches media traffic with the same source and destination, but without any specific ports. Consequently, SIP signaling traffic (from local port 5060) go through, but the media stream will be handled by appropriate SRTP SA.

```

security-policy
  name                    p1
  network-interface       private:0
  priority                0
  local-ip-addr-match    172.16.1.3
  remote-ip-addr-match   172.16.1.10
  local-port-match       5060
  remote-port-match      0
  trans-protocol-match   UDP
  direction              both
  local-ip-mask           255.255.255.255
  remote-ip-mask         255.255.255.255
  action                  allow
  ike-sainfo-name
  outbound-sa-fine-grained-mask
    local-ip-mask        255.255.255.255
    remote-ip-mask       255.255.255.255
    local-port-mask      0
    remote-port-mask     0
    trans-protocol-mask  0
  valid                   enabled
  vlan-mask               0xFFF
  last-modified-by       admin@console
  last-modified-date     2009-11-09 15:01:55

security-policy
  name                    p2
  network-interface       private:0
  priority                10
  local-ip-addr-match    172.16.1.3
  remote-ip-addr-match   172.16.1.10
  local-port-match       0
  remote-port-match      0
  trans-protocol-match   UDP
  direction              both

```

```

local-ip-mask          255.255.255.255
remote-ip-mask        255.255.255.255
action                 srtp
ike-sainfo-name
outbound-sa-fine-grained-mask
  local-ip-mask       0.0.0.0
  remote-ip-mask     255.255.255.255
  local-port-mask     0
  remote-port-mask   65535
  trans-protocol-mask 255
  valid              enabled
  vlan-mask          0xFFF
last-modified-by      admin@console
last-modified-date    2009-11-09 15:38:19

```

### Modified ALCI Configuration Elements

The action parameter in security-policy configuration mode has been modified to accept additional values, srtp and srtp.

From superuser mode, use the following command sequence to access media-sec-policy configuration mode.

```

ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# ipsec
ACMEPACKET(ipsec)# security-policy
ACMEPACKET(security-policy)# action ?
<enumeration> action (default: ipsec)
ipsec, allow, discard, srtp, srtp
ACMEPACKET(security-policy)#

```

The **show security** command has been updated with an **srtp** option.

```

ACMEPACKET# show security srtp
sad
spd
statistics
SRTP Statistics
status

```

The **srtp** option is similar to the **ipsec** option save for the **sad** sub-option that provides data for only SRTP SAs.

The **show sa stats** command has been updated with an **srtp** option.

```

ACMEPACKET# show sa stats
<ENTER> Show statistics summary of all Security Associations
<ike> Show statistics for IKE Security Associations
<ims-aka> Show statistics for IMS-AKA Security Associations
<srtp> Show statistics for SRTP Security Associations
sd# show sa stats srtp
20:06:24-114
SA Statistics
                ---- Lifetime ----
                Recent  Total  PerMax
SRTP Statistics
ADD-SA Req Rcvd      0      0      0
ADD-SA Success Resp Sent 0      0      0
ADD-SA Fail Resp Sent 0      0      0
DEL-SA Req Rcvd     0      0      0
DEL-SA Success Resp Sent 0      0      0
DEL-SA Fail Resp Sent 0      0      0
SA Added             0      0      0
SA Add Failed        0      0      0
SA Deleted           0      0      0
SA Delete Failed    0      0      0

```



## Multimedia Internet KEYing Protocol

The Secure Real-Time Transport Protocol, as described in RFC 3711, *The Secure Real-time Transport Protocol (SRTP)*, provides a framework for the encryption and authentication of Real-time Transport Protocol (RTP) and RTP Control Protocol (RTCP) streams. Both RTP and RTCP are defined by RFC 3550, *RTP: A Transport Protocol for Real-Time Applications*.

Encryption ensures that the call content and associated signalling remains private during transmission. Authentication ensures that (1) received packets are from the purported source, (2) packets are not been tampered with during transmission, and (3) a packet has not been replayed by a malicious server.

A significant number of service providers require the ability to encrypt the content and signalling of their real time communications sessions. There are several approaches to meeting this need, including the use of IPsec encryption as described in 3GPP TS 33.234 I-WLAN, *3G Security: Wireless Local Area Network (WLAN) Interworking Security*. An alternative approach utilizes SRTP as a protocol to encrypt the media, and MIKEY (as defined in RFC 3830, *Multimedia Internet Keying*) to exchange the keying information.

### Protocol Overview

While the RFC 3711 framework provides encryption and authentication procedures and defines a set of default cryptographic transforms required for RFC compliance, it does not specify a key management protocol to securely derive and exchange cryptographic keys. RFC3830, *Multimedia Internet Keying*, defines such a protocol that transmits cryptographic material using an exchange of a MIKEY initiation message (referred to as an I\_MESSAGE) and a MIKEY response message (referred to as an R\_MESSAGE).

Release S-C6.2.0 provides support for an initial MIKEY implementation, based on RFC 3830, and RFC 4567, *Key Management Extensions for Session Description Protocol (SDP) and Real Time Streaming Protocol (RTSP)*.

The MIKEY I\_MESSAGE takes the following format.

HDR, T, RAND, [IDi], [IDr], {SP}, KEMAC

where

HDR (defined in section 6.1 of RFC 3830) contains the common MIKEY header.

T (defined in section 6.6 of RFC 3830) contains a timestamp.

RAND (defined in section 6.11 of RFC 3830) contains a pseudo-random string used in the generation of TEKs (traffic encryption keys).

IDi and IDr (defined in section 6.7 of RFC 3830) optionally contain the initiator and responder ID.

SP (defined in sections 6.10 and 6.10.1 of RFC 3830) contains a list of security parameters. Pertinent parameters include encryption algorithm, encryption key length, authentication algorithm, authentication key length, salt key length, pseudo random function, key derivation rate, SRTP encryption switch, SRTCP encryption switch, and SRTP authentication switch).

KEMAC (defined in sections 6.2 and 6.13 of RFC 3830) contains TGK (TEK Generation Key) material that generates TEKs.

The MIKEY R\_MESSAGE takes the following format.

HDR, T, [IDr], V

where

HDR (defined in section 6.1 of RFC 3830) contains the common MIKEY header.

T (defined in section 6.6 of RFC 3830) contains a timestamp.

[IDr] (defined in section 6.7 of RFC 3830) optionally contains the responder ID.

V (defined in section 6.9 of RFC 3830) contains verification/authentication data.

MIKEY I and R MESSAGES are conveyed in base64-encoded format via SDP using the key management extensions specified in RFC 4567, Key Management Extensions for Session Description Protocol (SDP) and Real Time Streaming Protocol (RTSP). A sample initiation/response exchange follows.

### MIKEY I\_MESSAGE

```
v=0
o=alice 2891092738 2891092738 IN IP4 w-land.example.com
s=Cool stuff
e=alice@w-land.example.com
t=0 0
c=IN IP4 w-land.example.com
a=key-mgmt:mikey AQAFgM0XflABAAAAAAAAAAAAAAAAAAsAyONQ6gAAAAAGEE
oo2pee4hp2UaDX8ZE22YwKAAAPZG9uYWxkQGRlY2suY29tAQAAAAAAAAQAk0JKpgaV
kDaawi9whVBtBt0KZl4ymNuu62+Nv3ozPLygwK/GbAV9iemnGUIZl9fWQUO
SrzkTAv9zV
m=audio 49000 RTP/SAVP 98
a=rtpmap:98 AMR/8000
m=video 52230 RTP/SAVP 31
a=rtpmap:31 H261/90000
```

The initiator proposes to establish one audio stream and one video stream running SRTP (signaled by the use of the SAVP profile). The initiator offers a session based key to encrypt both audio and video streams by locating the key management a line above the m lines that describe the audio and video streams. The initiator uses MIKEY to set up the security parameters for SRTP. The MIKEY\_I message contains the security parameters, together with the necessary key material.

Upon receiving the offer, the responder validates the MIKEY\_I message, and, assuming support for the specified crypto suite and security parameters, accepts the offer and returns a MIKEY\_R MESSAGE to the initiator (conveyed via the 200 OK response).

### MIKEY R\_MESSAGE

```
v=0
o=bob 2891092897 2891092897 IN IP4 foo.example.com
s=Cool stuff
e=bob@foo.example.com
t=0 0
c=IN IP4 foo.example.com
a=key-mgmt:mikeyAQEFgM0XflABAAAAAAAAAAAAAAAAAYAyONQ6gAAAAAJAA
AQbWlja2V5QG1vdXNlLmNvbQABn8HdGE5BMDXFiuGEga+62AgY5cc=
m=audio 49030 RTP/SAVP 98
a=rtpmap:98 AMR/8000
m=video 52230 RTP/SAVP 31
a=rtpmap:31 H261/90000
```

Key exchange via text-based SDP is unacceptable in that malicious network elements could easily eavesdrop and obtain the plaintext keys, thus compromising the privacy and integrity of the encrypted media stream. Consequently, the SDP exchange must be protected by a security protocol such as IPsec or TLS.

## Licensing and Hardware Requirements

SRTP/SRTCP support requires the presence of an IPsec NIU and an SSM/SSM2 (Security Service Module).

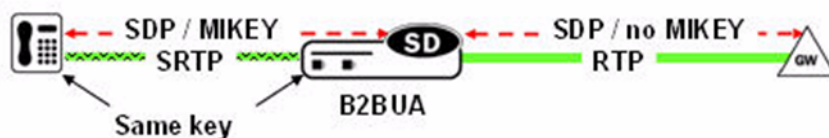
In addition, licenses for the desired SRTP capacity are also required.

## Operational Modes

SRTP topologies can be reduced to three basic topologies which are described in the following sections.

### Single-Ended SRTP Termination

Single-ended SRTP termination is illustrated in the following figure.



### Single-Ended SRTP Termination

When acting as a responder, upon receiving the SDP offer from a remote caller, the Oracle Communications Session Border Controller confirms that SRTP is enabled on the incoming realm by looking at the inbound mode in the media-sec-policy. If the SRTP is enabled, the SIP process parses the key-mgmt attributes to decrypt the MIKEY I\_MESSAGE. After successfully decrypting the message, it accepts or rejects the offer in accordance with the mikey-profile configuration element, assigned to the media-sec-policy. If the Oracle Communications Session Border Controller cannot parse or decrypt the message, the call is rejected.

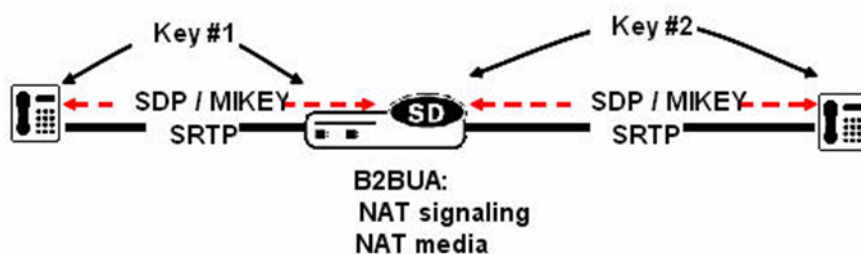
After obtaining the key and the security parameters from the I\_MESSAGE, the Oracle Communications Session Border Controller stores these parameters and sends a MIKEY R\_MESSAGE to the initiator. It also adds Security Associations to handle the SRTP traffic. If the initiator rejects the response the Oracle Communications Session Border Controller deletes the sessions along with the SRTP/SRTCP SAs.

Acting as an initiator (in the above diagram call coming from gateway to endpoint), the Oracle Communications Session Border Controller confirms that SRTP is enabled on the egress realm by looking at the outbound mode in the media-sec-policy. If SRTP is required, the Oracle Communications Session Border Controller uses MIKEY to generate cryptographic material and parameters (based on the MIKEY profile assigned to the egress realm). After generating cryptographic material, it constructs the MIKEY I\_MESSAGE, base64-encodes it, and transmits it via INVITE SDP.

On getting the R\_MESSAGE from the call recipient, the Oracle Communications Session Border Controller parses the message and determines if the offer has been accepted. If so, it forwards the answer, without the crypto parameters, back to the call originator. It also adds Security Associations to handle the SRTP traffic. If the initiator rejects the response the Oracle Communications Session Border Controller deletes the sessions along with the SRTP/SRTCP SAs.

### Back-to-Back SRTP Termination

Back-to-back SRTP termination is illustrated in the following figure.

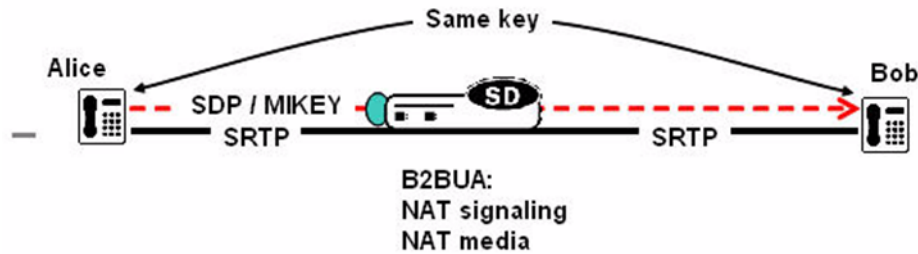


### Back-to-Back SRTP Termination

In this case the Oracle Communications Session Border Controller will act as an initiator on the one side and will act as a responder on the other side. See the description above for the initiator and responder functionality.

### SRTP Pass-Thru

SRTP pass-thru is illustrated in the following figure.



### SRTTP Pass-Through

If the media-sec-policy for the ingress and the egress realms specifies pass-thru mode, the Oracle Communications Session Border Controller does not intercept the SDES/MIKEY exchange between the caller and the callee. The SDES/MIKEY will be forwarded as it is from the caller to the callee, or vice versa.

## MIKEY Configuration

MIKEY configuration consists of the following steps.

1. The creation of one or more MIKEY profiles which specify parameter values negotiated during the MIKEY message exchange.
2. The creation of one or more Media Security Policies that specify key exchange protocols and protocol-specific profiles.
3. The assignment of a Media Security Policy to a realm.
4. The creation of an interface-specific Security Policy.

### MIKEY Profile Configuration

A MIKEY profile specifies the parameter values offered as a MIKEY initiator or accepted as a MIKEY responder.

**To configure MIKEY profile parameters:**

1. From superuser mode, use the following command sequence to access mikey-profile configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# media-security
ACMEPACKET(media-security)# mikey-profile
ACMEPACKET(mikey-profile)#
```

2. Use the required **name** parameter to provide a unique identifier for this mikey-profile instance.

**name** enables the creation of multiple media-sec-mikey-profile instances.

3. Use the **key-exchange-method** parameter to select the key exchange method that is offered when acting in the initiator role, or accepted when acting in the responder role.

pre-shared (password) is the only currently supported method.

4. Use the **encr-algorithm** parameter to select the encryption algorithm that is offered when acting in the initiator role, or accepted when acting in the responder role.

Allowable values are any, NULL, AES-CM

As a MIKEY initiator, select the encryption algorithm to be included in the I\_MESSAGE.

As a MIKEY responder, select the encryption algorithm to be accepted from the I\_MESSAGE.

The default value, any, indicates that the Oracle Communications Session Border Controller accepts the encryption algorithm specified in the MIKEY\_I\_MESSAGE, as long as it is within the valid range. For the outgoing direction (that is calls originated by the Oracle Communications Session Border Controller), the selection of the encryption algorithm for any is as follows:

- If MIKEY is used in the incoming direction the same encryption algorithm is used for the outgoing direction.
- If the incoming direction does not have SRTP, then AES-CM is used as the encryption algorithm.

When NULL is used RTP and RTCP packets are not encrypted.

5. Use the **auth-algorithm** parameter to select the encryption algorithm that is offered when acting in the initiator role, or accepted when acting in the responder role.

Allowable values are any, NULL, HMAC-SHA1-32, HMAC-SHA1-80

As a MIKEY initiator, select the authentication algorithm to be included in the I\_MESSAGE.

As a MIKEY responder, select the authentication algorithm to be accepted from the I\_MESSAGE.

The default value, any, indicates that the Oracle Communications Session Border Controller accepts the authentication algorithm specified in the MIKEY\_I\_MESSAGE, as long as it is within the valid range. For the outgoing direction (that is calls originated by the Oracle Communications Session Border Controller), the selection of the authentication algorithm for any is as follows:

- If MIKEY is used in the incoming direction the same encryption algorithm is used for the outgoing direction.
- If the incoming direction does not have SRTP, then HMAC-SHA1-80 is used as the encryption algorithm.

When NULL is used RTP and RTCP packets are not authenticated.

6. The **shared-secret** parameter contains the shared secret.

A shared secret is not required as MIKEY messages are not protected in this initial implementation.

7. Use the **mki** parameter to enable or disable the inclusion of the MKI:length field in the SDP key-management attribute.

The master key identifier (MKI) is an optional field within the SDP key-management attribute that differentiates one key from another. MKI is expressed as a pair of decimal numbers in the form: |mki:mki\_length| where mki is the MKI integer value and mki\_length is the length of the MKI field in bytes.

The MKI field is necessary only in topologies that may offer multiple keys within the key-management attribute.

Allowable values are enabled and disabled (the default).

enabled – an MKI field is sent within the crypto attribute (16 bytes maximum, dependent on hardware)

disabled – no MKI field is sent

8. Use **done**, **exit**, and **verify-config** to complete configuration of this MIKEY profile instance.
9. Repeat Steps 1 through 8 to configure additional MIKEY profiles.

## Media Security Policy Configuration

Use the following procedure to create a Media Security Policy that specifies the role of the Oracle Communications Session Border Controller in the security negotiation. If the Oracle Communications Session Border Controller takes part in the negotiation, the policy specifies a key exchange protocol and SDES profile for both incoming and outgoing calls.

**To configure media-security-policy parameters:**

1. From superuser mode, use the following command sequence to access media-sec-policy configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# media-security
ACMEPACKET(media-security)# media-sec-policy
ACMEPACKET(media-sec-policy)#
```

2. Use the required **name** parameter to provide a unique identifier for this media-sec-policy instance.

**name** enables the creation of multiple media-sec-policy instances.

3. Use optional **pass-thru** parameter to enable or disable pass-thru mode.

With pass-thru mode enabled, the User Agent (UA) endpoints negotiate security parameters between each other; consequently, the Oracle Communications Session Border Controller simply passes SRTP traffic between the two endpoints.

With pass-thru mode disabled, the Oracle Communications Session Border Controller disallows end-to-end negotiation — rather the Oracle Communications Session Border Controller initiates and terminates SRTP tunnels with both endpoints

In the absence of an explicitly configured value, the default value, disabled, is used.

4. Use the **outbound** navigation command to move to media-sec-outbound configuration mode. While in this configuration mode you specify security parameters applied to the outbound call leg, that is calls sent by the Oracle Communications Session Border Controller.

5. Use the **protocol** parameter to select the key exchange protocol.

Allowable values are mikey and sdes.

Select mikey for MIKEY key exchange.

6. Use the **profile** parameter to specify the name of the MIKEY profile associated with the outbound leg of this media-security-policy.

7. Use the **mode** parameter to select the real time transport protocol.

Allowable values are rtp and srtp (the default).

**mode** identifies the transport protocol (RTP or SRTP) included in an SDP offer when this media-security-policy is in effect.

As an initiator, the Oracle Communications Session Border Controller send a MIKEY I\_MESSAGE with media descriptions that have the transport protocol specified in this field.

As a responder, the Oracle Communications Session Border Controller accepts a MIKEY I\_MESSAGE with media descriptions that have the transport protocol specified in this field. If a media description has no transport protocol that matches the value of this field, the INVITE message is rejected with the 488 Not Acceptable Here response.

8. Use the **done** and **exit** parameters to return to media-sec-policy configuration mode.

9. Use the **inbound** navigation command to move to media-sec-inbound configuration mode. While in this configuration mode you specify security parameters applied to the inbound call leg, that is calls received by the Oracle Communications Session Border Controller.

10. Use the **protocol** parameter to select the key exchange protocol.

Allowable values are mikey and sdes.

Select mikey for MIKEY key exchange.

11. Use the **profile** parameter to specify the name of the MIKEY profile associated with the inbound leg of this media-security-policy.

12. Use the **mode** parameter to select the real time transport protocol.

Allowable values are rtp and srtp (the default).

**mode** identifies the transport protocol (RTP or SRTP) included in an SDP offer when this media-security-policy is in effect.

As an initiator, the Oracle Communications Session Border Controller sends a MIKEY I\_MESSAGE with media descriptions that have the transport protocol specified in this field.

As a responder, the Oracle Communications Session Border Controller accepts a MIKEY I\_MESSAGE with media descriptions that have the transport protocol specified in this field. If a media description has no transport protocol that matches the value of this field, the INVITE message is rejected with the 488 Not Acceptable Here response.

13. Use **done**, **exit**, and **verify-config** to complete configuration of this media security policy instance.

14. Repeat Steps 1 through 13 to configure additional media-security policies.

## Assigning the Media Security Policy to a Realm

To assign a media-security-policy to a realm:

1. From superuser mode, use the following command sequence to access realm-config configuration mode. While in this mode, you assign an existing media-security-policy to an existing realm.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)# select
identifier:
1. access-10
...
...
selection: 1
ACMEPACKET(realm-config)#
```

2. Use the **media-sec-policy** parameter to assign the policy to the current realm.
3. Use **done**, **exit**, and **verify-config** to complete assignment of the media-security-policy to the realm.

## ACLI Example Configurations

The following section contain relevant sections of system configurations for basic operational modes.

### Single-Ended SRTP Termination Configuration

```
ACMEPACKET# show running-config
...
...
...
mikey-profile
  name mikey1
  key-exchange-method pre-shared
  encr-algorithm any
  auth-algorithm any
  shared-secret
  mki disabled
  last-modified-by admin@console
  last-modified-date 2009-11-15 14:37:13
  ...
  ...
  ...
media-sec-policy
  name msp2
  pass-through disabled
  inbound
    profile mikey1
    mode srtp
    protocol mikey
  outbound
    profile sdes1
    mode srtp
    protocol mikey
  last-modified-by admin@console
  last-modified-date 2009-11-15 14:37:51
  ...
  ...
  ...
realm-config
  identifier peer
  description
  addr-prefix 192.168.0.0/16
  network-interfaces M00:0
```

```

mm-in-realm          enabled
mm-in-network        enabled
mm-same-ip           enabled
mm-in-system         enabled
bw-cac-non-mm        disabled
msm-release          disabled
qos-enable           disabled
generate-UDP-checksum disabled
max-bandwidth        0
fallback-bandwidth   0
max-priority-bandwidth 0
max-latency          0
max-jitter           0
max-packet-loss      0
observ-window-size   0
parent-realm         dns-realm
media-policy         media-sec-policy
media-sec-policy     msp2
in-translationid
...
...
...
last-modified-by    admin@console
last-modified-date  2009-11-15 14:38:19

```

### Back-to-Back SRTP Termination Configuration

```

ACMEPACKET# show running-config
...
...
...
sdes-profile
  name          sdes1
  crypto-list   AES_CM_128_HMAC_SHA1_80
  srtp-auth     enabled
  srtp-encrypt  enabled
  srtcp-encrypt enabled
  mki           disabled
  key
  salt
  last-modified-by    admin@console
  last-modified-date  2009-11-16 15:37:13
media-sec-policy
  name          msp2
  pass-through  enabled
  inbound
    profile     sdes1
    mode        srtp
    protocol    sdes
  outbound
    profile     sdes1
    mode        srtp
    protocol    sdes
  last-modified-by    admin@console
  last-modified-date  2009-11-16 15:37:51
...
...
...
realm-config
  identifier     peer
  description
  addr-prefix    192.168.0.0/16
  network-interfaces M00:0

```



```

mm-in-realm          enabled
mm-in-network        enabled
mm-same-ip           enabled
mm-in-system         enabled
bw-cac-non-mm        disabled
msm-release          disabled
qos-enable           disabled
generate-UDP-checksum disabled
max-bandwidth        0
fallback-bandwidth   0
max-priority-bandwidth 0
max-latency          0
max-jitter           0
max-packet-loss      0
observ-window-size   0
parent-realm
dns-realm
media-policy
media-sec-policy     msp2
...
...
...
realm-config
  identifier          core
  description
  addr-prefix         172.16.0.0/16
  network-interfaces M10:0
  mm-in-realm         enabled
  mm-in-network        enabled
  mm-same-ip           enabled
  mm-in-system         enabled
  bw-cac-non-mm        disabled
  msm-release          disabled
  qos-enable           disabled
  generate-UDP-checksum disabled
  max-bandwidth        0
  fallback-bandwidth   0
  max-priority-bandwidth 0
  max-latency          0
  max-jitter           0
  max-packet-loss      0
  observ-window-size   0
  parent-realm
  dns-realm
  media-policy
  media-sec-policy     msp2
  in-translationid
...
...
...
  last-modified-by    admin@console
  last-modified-date  2009-11-10 15:38:19

```

### SRTP Pass-Thru Configuration

```

ACMEPACKET# show running-config
...
...
...
sdes-profile
  name                sdes1
  crypto-list          AES_CM_128_HMAC_SHA1_80
  srtp-auth            enabled
  srtp-encrypt         enabled

```

```

    srtcp-encrypt          enabled
    mki                   disabled
    key
    salt
    last-modified-by      admin@console
    last-modified-date    2009-11-16 15:37:13
media-sec-policy
  name                   msp2
  pass-through          enabled
  inbound
    profile              sdes1
    mode                 srtcp
    protocol             sdes
  outbound
    profile              sdes1
    mode                 srtcp
    protocol             sdes
  last-modified-by      admin@console
  last-modified-date    2009-11-16 15:37:51
...
...
...
realm-config
  identifier             peer
  description
  addr-prefix            192.168.0.0/16
  network-interfaces    M00:0
  mm-in-realm           enabled
  mm-in-network         enabled
  mm-same-ip            enabled
  mm-in-system          enabled
  bw-cac-non-mm        disabled
  msm-release           disabled
  qos-enable            disabled
  generate-UDP-checksum disabled
  max-bandwidth         0
  fallback-bandwidth    0
  max-priority-bandwidth 0
  max-latency           0
  max-jitter            0
  max-packet-loss       0
  observ-window-size    0
  parent-realm
  dns-realm
  media-policy
  media-sec-policy      msp2
...
...
...
realm-config
  identifier             core
  description
  addr-prefix            172.16.0.0/16
  network-interfaces    M10:0
  mm-in-realm           enabled
  mm-in-network         enabled
  mm-same-ip            enabled
  mm-in-system          enabled
  bw-cac-non-mm        disabled
  msm-release           disabled
  qos-enable            disabled
  generate-UDP-checksum disabled
  max-bandwidth         0
  fallback-bandwidth    0

```

```

max-priority-bandwidth      0
max-latency                 0
max-jitter                  0
max-packet-loss            0
observ-window-size         0
parent-realm
dns-realm
media-policy
media-sec-policy           msp2
in-translationid
...
...
...
last-modified-by           admin@console
last-modified-date         2009-11-10 15:38:19

```

## Security Policy

A Security Policy enables the Oracle Communications Session Border Controller to identify inbound and outbound media streams that are treated as SRTP/SRTCP. The high-priority Security Policy, p1, (shown below) allows signaling traffic from source 172.16.1.3 to destination 172.16.1.10:5060. The lower-priority Security Policy, p2, (also shown below) matches media traffic with the same source and destination, but without any specific ports. Consequently, SIP signaling traffic (from local port 5060) go through, but the media stream will be handled by appropriate SRTP SA.

```

security-policy
  name                p1
  network-interface   private:0
  priority            0
  local-ip-addr-match 172.16.1.3
  remote-ip-addr-match 172.16.1.10
  local-port-match    5060
  remote-port-match   0
  trans-protocol-match UDP
  direction           both
  local-ip-mask       255.255.255.255
  remote-ip-mask      255.255.255.255
  action              allow
  ike-sainfo-name
  outbound-sa-fine-grained-mask
    local-ip-mask     255.255.255.255
    remote-ip-mask    255.255.255.255
    local-port-mask   0
    remote-port-mask  0
    trans-protocol-mask 0
    valid             enabled
    vlan-mask         0xFFF
  last-modified-by   admin@console
  last-modified-date 2009-11-09 15:01:55

security-policy
  name                p2
  network-interface   private:0
  priority            10
  local-ip-addr-match 172.16.1.3
  remote-ip-addr-match 172.16.1.10
  local-port-match    0
  remote-port-match   0
  trans-protocol-match UDP
  direction           both
  local-ip-mask       255.255.255.255
  remote-ip-mask      255.255.255.255
  action              srtp
  ike-sainfo-name
  outbound-sa-fine-grained-mask

```

local-ip-mask	0.0.0.0
remote-ip-mask	255.255.255.255
local-port-mask	0
remote-port-mask	65535
trans-protocol-mask	255
valid	enabled
vlan-mask	0xFFF
last-modified-by	admin@console
last-modified-date	2009-11-09 15:38:19

## Modified ALCI Configuration Elements

The action parameter in security-policy configuration mode has been modified to accept additional values, srtp and srtp.

From superuser mode, use the following command sequence to access media-sec-policy configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# ipsec
ACMEPACKET(ipsec)# security-policy
ACMEPACKET(security-policy)# action ?
<enumeration> action (default: ipsec)
    ipsec, allow, discard, srtp, srtp
ACMEPACKET(security-policy)#
```

The **show sa stats** command has been enhanced.

```
ACMEPACKET# show sa stats srtp
12:27:45-153
SA Statistics
----- Lifetime -----
              Recent      Total      PerMax

SRTP Statistics
ADD-SA Req Rcvd          0          4          4
ADD-SA Success Resp Sent 0          3          3
ADD-SA Fail Resp Sent   0          1          1
DEL-SA Req Rcvd          0          0          0
DEL-SA Success Resp Sent 0          0          0
DEL-SA Fail Resp Sent   0          0          0
SA Added                 0          3          3
SA Add Failed            0          1          1
SA Deleted               0          0          0
SA Delete Failed        0          0          0
ACMEPACKET# show sa stats
12:36:00-148
SA Statistics
----- Lifetime -----
              Recent      Total      PerMax

IKE Statistics
ADD-SA Req Rcvd          0          4          4
ADD-SA Success Resp Sent 0          3          3
ADD-SA Fail Resp Sent   0          1          1
DEL-SA Req Rcvd          0          0          0
DEL-SA Success Resp Sent 0          0          0
DEL-SA Fail Resp Sent   0          0          0
ACQUIRE-SA Req Sent    0          5          5
ACQUIRE-SA Success Resp 0          5          5
ACQUIRE-SA Fail Resp Rcv 0          0          0
ACQUIRE-SA Trans Timeout 0          0          0
SA Added                 0          3          3
SA Add Failed            0          1          1
SA Deleted               0          0          0
SA Delete Failed        0          0          0
IMS-AKA Statistics
ADD-SA Req Rcvd          0          0          0
ADD-SA Success Resp Sent 0          0          0
```

ADD-SA Fail Resp Sent	0	0	0
DEL-SA Req Rcvd	0	0	0
DEL-SA Success Resp Sent	0	0	0
DEL-SA Fail Resp Sent	0	0	0
SA Added	0	0	0
SA Add Failed	0	0	0
SA Deleted	0	0	0
SA Delete Failed	0	0	0
SRTP Statistics			
ADD-SA Req Rcvd	0	0	0
ADD-SA Success Resp Sent	0	0	0
ADD-SA Fail Resp Sent	0	0	0
DEL-SA Req Rcvd	0	0	0
DEL-SA Success Resp Sent	0	0	0
DEL-SA Fail Resp Sent	0	0	0
SA Added	0	0	0
SA Add Failed	0	0	0
SA Deleted	0	0	0
SA Delete Failed	0	0	0
ACMEPACKET#			

## Secure and Non-Secure Flows in the Same Realm

To simplify deployments, the Oracle Communications Session Border Controller allows secure and non-secure flows in the same realm. This broadened set of capabilities means the Oracle Communications Session Border Controller can support RTP and SRTP flows, and it can support a larger group of UAs that might have varying SRTP abilities. Prior to this release, when a cryptographic session arrived at the Oracle Communications Session Border Controller and failed to match an applicable media security profile, it was rejected.

This broadened support for secure and non-secure flows and for UAs with various SRTP abilities is established throughout the system, residing in these configurations:

- `media-sec-policy`
- `sdes-profile`
- `mikey-profile`

While configurations reside there, you should also note special considerations for the `security-policy` configuration and implications for security associations.

## Mode Settings in the Media Security Policy

The media security policy configuration's `mode` parameter offers three settings. It is the `any` mode that allows you to support secure and non-secure flows in the same realm.

### For Incoming Flows

This section describes the way all three settings behavior for incoming flows.

- `rtp`—If the incoming media security policy associated with a realm has `rtp` set as its mode, then the Oracle Communications Session Border Controller only accepts offer SDP containing RTP/AVP media lines. Otherwise, the Oracle Communications Session Border Controller rejects the session with a 488 Not Acceptable Here.
- `srtp`—If the incoming media security policy associated with a realm has `srtp` set as its mode, the Oracle Communications Session Border Controller only accepts offer SDP containing RTP/SAVP media lines. Otherwise, the Oracle Communications Session Border Controller rejects the session with a 488 Not Acceptable Here.
- `any`—If the incoming media security policy associated with a realm has `any` set as its mode, the Oracle Communications Session Border Controller accepts offer SDP that has RTP/AVP media lines, RTP/SAVP media lines, or both.

### For Outgoing Flows

This section describes the way all three settings behavior for outgoing flows.

- **rtp**—If the outgoing media security policy associated with a realm has **rtp** set as its mode, then the Oracle Communications Session Border Controller converts any RTP/SAVP media lines from incoming offer SDP to RTP/AVP for the offer SDP it sends out.

Incoming offer SDP might look like this:

```
v=0
o=MxSIP 0 1480968866 IN IP4 192.168.22.180
s=SIP Call
c=IN IP4 192.168.22.180
t=0 0
m=audio 5010 RTP/SAVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=crypto:0 AES_CM_128_HMAC_SHA1_80 inline:f0oLKTuMYwXqrKa7Ch
+MOBvLe8YnXnD6Kmnj4LQ2
```

The Oracle Communications Session Border Controller will take that and convert it to the following for outgoing traffic.

```
v=0
o=MxSIP 0 1480968866 IN IP4 172.16.22.180
s=SIP Call
c=IN IP4 172.16.22.180
t=0 0
m=audio 6000 RTP/AVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-1
```

This conversion can result in multiple media lines with RTP/AVP for the same media profile and an RTP/SAVP media line for the same media profile. To prevent duplicate lines in the SDP the Oracle Communications Session Border Controller sends, the Oracle Communications Session Border Controller inspects incoming SDP to determine if RTP/AVP and RTP/SAVP media lines exist for the same media profile. If it finds such a media profile, the Oracle Communications Session Border Controller disables the RTP/AVP (by setting the port to 0 in the outgoing offer SDP) corresponding to the RTP/AVP media line for that media profile. Doing so forces the UA answering the SDP offer to choose the media lines corresponding to the RTP/SAVP media lines in the incoming offer SDP. An SRTP-RTP session results.

The incoming offer SDP might look like this:

```
v=0
o=MxSIP 0 1480968866 IN IP4 192.168.22.180
s=SIP Call
c=IN IP4 192.168.22.180
t=0 0
m=audio 5012 RTP/AVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
```

```

a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-15
m=audio 5010 RTP/SAVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=crypto:0 AES_CM_128_HMAC_SHA1_80 inline:f0oLKTuMYwXqrKa7Ch
+MOBvLe8YnXnD6Kmnj4LQ2

```

And the outgoing offer SDP will look like this:

```

v=0
o=MxSIP 0 1480968866 IN IP4 172.16.22.180
s=SIP Call
c=IN IP4 172.16.22.180
t=0 0
m=audio 0 RTP/AVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-15
m=audio 6002 RTP/AVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-15

```

- **srtp**—If the outgoing media security policy associated with a realm has **srtp** set as its mode, the Oracle Communications Session Border Controller converts any RTP/AVP media lines from an incoming offer SDP to RTP/SAVP for the offer SDP the Oracle Communications Session Border Controller sends.

The incoming offer SDP might look like this:

```

v=0
o=MxSIP 0 1480968866 IN IP4 192.168.22.180
s=SIP Call
c=IN IP4 192.168.22.180
t=0 0
m=audio 5012 RTP/AVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-15

```

And the outgoing offer SDP will look like this:

```
v=0
o=MxSIP 0 1480968866 IN IP4 172.16.22.180
s=SIP Call
c=IN IP4 172.16.22.180
t=0 0
m=audio 6000 RTP/SAVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-1
a=crypto:0 AES_CM_128_HMAC_SHA1_80 inline:f0oLKTuMYwXqrKa7Ch
+MOBvLe8YnXnD6Kmnj4LQ2
```

This conversion might result in multiple media lines with RTP/SAVP for the same media profile if the incoming offer SDP has an RTP/AVP media line and an RTP/SAVP media for the same media profile. To prevent multiple identical media lines in the SDP it sends, the Oracle Communications Session Border Controller inspects the incoming SDP to determine whether both RTP/AVP and RTP/SAVP media lines exist for the same media profile. If it finds such a media profile, the Oracle Communications Session Border Controller disables the RTP/SAVP (by setting the port to 0 in the outgoing offer SDP) corresponding to the RTP/AVP media line for that media profile. Doing so forces the UA answering the SDP offer to choose the media lines corresponding to the RTP/SAVP media lines in the incoming offer SDP. An SRTP-SRTP session results.

The incoming offer SDP might look like this:

```
v=0
o=MxSIP 0 1480968866 IN IP4 192.168.22.180
s=SIP Call
c=IN IP4 192.168.22.180
t=0 0
m=audio 5012 RTP/AVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-15
m=audio 5010 RTP/SAVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=crypto:0 AES_CM_128_HMAC_SHA1_80 inline:f0oLKTuMYwXqrKa7Ch
+MOBvLe8YnXnD6Kmnj4LQ2
```

And the outgoing offer SDP will look like this:

```
v=0
o=MxSIP 0 1480968866 IN IP4 172.16.22.180
s=SIP Call
c=IN IP4 172.16.22.180
t=0 0
m=audio 0 RTP/SAVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
```



```

a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-15
m=audio 6002 RTP/SAVP 0 8 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - - -
a=fmtp:18 annexb=no
a=fmtp:101 0-1
a=crypto:0 AES_CM_128_HMAC_SHA1_80 inline:f0oLKTuMYwXqrKa7Ch
+MOBvLe8YnXnD6Kmnj4LQ2

```

- **any**—If the outgoing media security policy associated with a realm has **any** set as its mode, the Oracle Communications Session Border Controller creates offer SDP based on the value configured in the **egress-offer-format**, which can be set either in the **sdes-profile** or the **mikey-profile** configurations.
  - If the value is **same-as-ingress**, the Oracle Communications Session Border Controller leaves the profile of the media lines unchanged.
  - If the value is **simultaneous-best-effort**, the Oracle Communications Session Border Controller inspects the incoming offer SDP and:
    - Adds an RTP/SAVP media line for any media profile that has only the RTP/AVP media profile
    - Adds an RTP/AVP media line for any media profile that has only the RTP/SAVP media profile

Should the media profile in the incoming offer SDP already have two media lines (one for RTP/AVP and one for RTP/SAVP), the Oracle Communications Session Border Controller does not have to make these additions. It will map the media lines in the answer it receives with the media lines from the incoming offer SDP. It will also ensure that media lines in the answer SDP it sends match the media lines from the incoming offer SDP.


## Security Associations for RTP and RTCP

With RTP and SRTP supported in the same realm, you want to configure your SRTP security policies to preserve system resources and exercise the full capability number of licensed sessions. You need to do to avoid session agent interaction that can have an adverse impact on the number of sessions.

To do so, check the **local-ip-match-address** for the STRP security policy has an IP address different from the all steering pool IP addresses for realms requiring both RTP and SRTP. The Oracle Communications Session Border Controller recognizes this difference automatically and sets the connection address of media lines in SDP accordingly:

- The connection address for RTP media lines is the IP address of the applicable steering pool. The Oracle Communications Session Border Controller passes through RTP and RTCP packets sent by and received from the steering pool IP address. This operation requires no reference to session agents because the steering pool address does not match the IP address for the SRTP security policy's **local-ip-address-match** value.
- The connection address of the SRTP media lines continues to be the **local-ip-address-match** value from the applicable SRTP security policy.

Since RTP and RTCP packets are sent to and from the steering pool's IP address (an IP address for which there is no SRTP security policy configured), there is no reason to reference session agents.

-  **Note:** Oracle's Enhanced Traffic Controller (ETC) networking interface unit handles traffic differently such the issue with session agent reference is elided. That is, if you are using the ETC NIU, you do not need to be concerned about this issue.

### Security Configuration for RTP and RTCP

This section shows you how to configure your Oracle Communications Session Border Controller to support secure and non-secure flows in the same realm.

To configure a security policy to support secure and non-secure flows in the same realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **security** and press Enter.

```
ACMEPACKET(configure)# security
ACMEPACKET(security)#
```

3. Type **media-security** and press Enter.

```
ACMEPACKET(security)# media-security
ACMEPACKET(media-security)#
```

4. Type **media-sec-policy** and press Enter. If you are editing a pre-existing configuration, you need to select it before you can make changes.

```
ACMEPACKET(media-security)# media-sec-policy
ACMEPACKET(media-sec-policy)#
```

5. Type **inbound** to enter the setting for inbound flows.

```
ACMEPACKET(media-sec-policy)# inbound
ACMEPACKET(inbound)#
```

6. **mode**—Enter the mode that you want to use for inbound flows. You can choose from **rtp**, **srtcp**, and **any**.

7. **protocol**—Change this value from either **sdes** or **mikey** to **none**. Use the **done** command to save your work, and exit the **inbound** configuration.

8. Type **outbound** to enter the setting for inbound flows.

```
ACMEPACKET(media-sec-policy)# outbound
ACMEPACKET(outbound)#
```

9. **mode**—Enter the mode that you want to use for outbound flows. You can choose from **rtp**, **srtcp**, and **any**.

10. **protocol**—Change this value from either **sdes** or **mikey** to **none**. Use the **done** command to save your work, and exit the **outbound** configuration.

11. Type **done** and continue.

To set the egress offer format for an SDES profile configuration:

12. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

13. Type **security** and press Enter.

```
ACMEPACKET(configure)# security
ACMEPACKET(security)#
```

14. Type **media-security** and press Enter.

```
ACMEPACKET(security)# media-security
ACMEPACKET(media-security)#
```

15. Type **sdes-profile** and press Enter. If you are editing a pre-existing configuration, you need to select it before you can make changes.

```
ACMEPACKET(media-security)# sdes-profile
ACMEPACKET(sdes-profile)#
```

16. **egress-offer-format**—Choose an egress offer format for this profile to use when you set the outbound mode in the media security policy to **any**. You can select one of two values:

- If the value is **same-as-ingress (default)**, the Oracle Communications Session Border Controller leaves the profile of the media lines unchanged.
- If the value is **simultaneous-best-effort**, the Oracle Communications Session Border Controller inspects the incoming offer SDP and:
  - Adds an RTP/SAVP media line for any media profile that has only the RTP/AVP media profile
  - Adds an RTP/AVP media line for any media profile that has only the RTP/SAVP media profile

17. Type **done** to save your work and continue.

To set the egress offer format for an SDES profile configuration:

18. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

19. Type **security** and press Enter.

```
ACMEPACKET(configure)# security
ACMEPACKET(security)#
```

20. Type **media-security** and press Enter.

```
ACMEPACKET(security)# media-security
ACMEPACKET(media-security)#
```

21. Type **mikey-profile** and press Enter. If you are editing a pre-existing configuration, you need to select it before you can make changes.

```
ACMEPACKET(media-security)# mikey-profile
ACMEPACKET(mikey-profile)#
```

22. **egress-offer-format**—Choose an egress offer format for this profile to use when you set the outbound mode in the media security policy to any. You can select one of two values:

- If the value is **same-as-ingress (default)**, the Oracle Communications Session Border Controller leaves the profile of the media lines unchanged.
- If the value is **simultaneous-best-effort**, the Oracle Communications Session Border Controller inspects the incoming offer SDP and:
  - Adds an RTP/SAVP media line for any media profile that has only the RTP/AVP media profile
  - Adds an RTP/AVP media line for any media profile that has only the RTP/SAVP media profile


23. Type **done** to save your work and continue.

## Supporting UAs with Different SRTP Capabilities

To support UAs with different levels of SRTP capabilities, the **use-ingress-session-params** parameter appears in both the **sdes-profile** and **mikey-profile** configurations. The values for this parameter allow the Oracle Communications Session Border Controller to accept and (where applicable) mirror the UA's proposed cryptographic session parameters:

- **srtcp-auth**—Decides whether or not authentication is performed in SRTCP
- **srtcp-encrypt**—Decides whether or not encryption is performed in SRTCP
- **srtcp-encrypt**—Decides whether or not encryption is performed in SRTCP

Using these possible values, the Oracle Communications Session Border Controller accepts the corresponding incoming session parameters.

 **Note:** For MIKEY, this parameter and its function are reserved for future use.

### Receiving Offer SDP

When the Oracle Communications Session Border Controller receives offer SDP with applicable session parameters, it uses the same session parameters in its answer SDP (if it can support the same). This is true even if the value for that session parameter differs from the available media security profile.

Consider this example: An SDES profile is applied for incoming direction for a media security policy configured with the **srtcp-encrypt** value set to **enabled**. With the **use-ingress-session-params** parameter set to **srtcp-encrypt** for the SDES profile, the Oracle Communications Session Border Controller accepts the offer SDP and also sets UNENCRYPTED\_SRTCP for the cryptographic attributes in its answer SDP. When the call connects, the Oracle Communications Session Border Controller does not encrypt or decrypt SRTCP packets. Without the SDES profile set this way, the Oracle Communications Session Border Controller would reject offer SDP if any of its cryptographic attributes showed UNENCRYPTED\_SRTCP in its session parameters list.

### Receiving Answer SDP

When the Oracle Communications Session Border Controller receives answer SDP with the accepted session parameter, the value for the same session parameters that the Oracle Communications Session Border Controller uses might or might not be the same as the incoming value. Configuration of the outbound media security profile controls the value used because the Oracle Communications Session Border Controller makes offer SDP, which cannot be changed, with the session parameters based on the outgoing media security profile.

Consider this example: An SDES profile is applied for incoming direction for a media security policy configured with the **srtcp-encrypt** value set to **enabled**, so the cryptographic attributes in the SDP the system sends **do not have the UNENCRYPTED\_SRTCP** session parameters. If the UNENCRYPTED\_SRTCP appears in the corresponding answer SDP it receives, the Oracle Communications Session Border Controller accepts it if the **srtcp-encrypt** value appears in the **use-ingress-session-params** parameter. But the Oracle Communications Session Border Controller still performs SRTCP encryption. When the call connects, the Oracle Communications Session Border Controller encrypts outgoing SRTCP packets but does not decrypt incoming SRTCP packets. So if the UA (receiving the system's offer SDP) does not support SRTCP decryption, it will likely reject the offer SDP.

### SDES Profile Configuration

To set the ingress session parameters for an SDES profile configuration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **security** and press Enter.

```
ACMEPACKET (configure) # security
ACMEPACKET (security) #
```

3. Type **media-security** and press Enter.

```
ACMEPACKET (security) # media-security
ACMEPACKET (media-security) #
```

4. Type **sdes-profile** and press Enter. If you are editing a pre-existing configuration, you need to select it before you can make changes.

```
ACMEPACKET (media-security) # sdes-profile
ACMEPACKET (sdes-profile) #
```

5. **use-ingress-session-params**—Enter the list of values for which the Oracle Communications Session Border Controller will accept and (where applicable) mirror the UA's proposed cryptographic session parameters:

- **srtcp-auth**—Decides whether or not authentication is performed in SRTP
- **srtcp-encrypt**—Decides whether or not encryption is performed in SRTP
- **srtcp-encrypt**—Decides whether or not encryption is performed in SRTCP

```
ACMEPACKET(sdes-profile) # use-ingress-session-params srtp-auth srtp-
encrypt srtcp-encrypt
```

6. Type **done** to save your work and continue.

## Refining Interoperability

To refine any remaining interoperability issues, you can use the options parameter in these configurations: **media-sec-policy**, **sdes-profile**, and **mikey-profile**.

Commons value to configure an option are **include-local-id** and **include-remote-id**. By default, the Oracle Communications Session Border Controller does not include the IDi or IDr when sending the MIKEY I\_MESSAGE. And it does not set the IDr in the MIKEY R\_MESSAGE. Using the options provides these results:

- **include-local-id**—The Oracle Communications Session Border Controller includes the IDi in the I\_MESSAGE and the IDr in the R\_MESSAGE.

When used for the outbound direction of a media security policy, the IDi is included in the I\_MESSAGE the Oracle Communications Session Border Controller sends. The content of the IDi is the value of the Contact header found in the INVITE message.

When configured for the mikey-profile associated with the inbound media security policy, the Oracle Communications Session Border Controller includes the IDr in the R\_MESSAGES it sends in response to incoming I\_MESSAGES. The content of the IDr is the value of the Contact header found in the 200 OK response.

- **include-remote-id**—The system includes the IDr in the I\_MESSAGE.

When configured for the mikey-profile associated with the outbound media security policy, the Oracle Communications Session Border Controller includes the IDr in I\_MESSAGES it initiates. The content of the IDr is the value of the Request-URI from the INVITE message.

## Refining Interoperability Configuration

You can configure these options for **media-sec-policy**, **sdes-profile**, and **mikey-profile** configurations. The following uses the **mikey-profile** to demonstrate how to enter them.

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **security** and press Enter.

```
ACMEPACKET(configure)# security
ACMEPACKET(security)#
```

3. Type **media-security** and press Enter.

```
ACMEPACKET(security)# media-security
ACMEPACKET(media-security)#
```

4. Type **mikey-profile** and press Enter. If you are editing a pre-existing configuration, you needs to select it before you can make changes.

```
ACMEPACKET(media-security)# mikey-profile
ACMEPACKET(mikey-profile)#
```

5. **options**—Your entry will look like this when you add both values:

```
ACMEPACKET(mikey-profile)# options include-local-id, include-remote-id
```

You can you the plus sign (+) and the minus sign (-) to add and remove values from the options list.

To remove an value, your entry would look like this:

```
ACMEPACKET(mikey-profile)# options -include-local-id
```

To add an value, your entry would look like this:

```
ACMEPACKET(mikey-profile) # options +include-local-id
```

## Multi-system Selective SRTP Pass-through

---

Prior to Release S-CX6.3F1, Oracle Communications Session Border Controller provided a single Secure Real-time Transport Protocol (SRTP) operational mode, referred to as SRTP Media Proxy Mode. In this original processing mode, each Oracle Communications Session Border Controller in the SRTP media path served as a proxy for the media — always decrypting inbound traffic, and encrypting outbound traffic.

Release S-CX6.3F1 introduces support for a new, alternative processing mode, referred to as Multi-system Selective SRTP Pass-through Mode. In this new mode encryption and decryption of SRTP media is handled by the SRTP endpoints, the calling and called parties. Off-load of the processor-intensive encryption/decryption provides the Oracle Communications Session Border Controller with the ability to handle a larger number of simultaneous SRTP sessions.

With Multi-system Selective SRTP Pass-through enabled, the Oracle Communications Session Border Controller can be configured to selectively allow hair-pinned (spiral) SRTP packets to pass through the Oracle Communications Session Border Controller without encryption or decryption.



**Note:** hair-pinned calls are those calls where the calling and called parties are within the same realm and/or within the same sub-network.

## License Requirements

Multi-system Selective SRTP Pass-through requires only the SIP license.

## Hardware Requirements

On the NN4500 platform, Multi-system Selective SRTP Pass-through requires an SSM2 (for SIP/TLS signalling), and either an IPsec/SRTP NIU, or an ETC NIU.

## Constraints

Multi-system Selective SRTP Pass-through support has the following constraints:

1. The realm, or realms in which the calling and called parties are located are configured for Multi-system Selective SRTP Pass-through support.
2. The call session does not require SIP—INFO to RFC 2833 tone translation.
3. The call session does not require SDES/MIKEY internetworking. Both SDES and MIKEY can be used for the exchange of SRTP keying material. Both the calling and called parties must support the same key exchange protocol.
4. Multi-system Selective SRTP Pass-through support should not be enabled in topologies where core-side application servers may change the c-line to inject a media server or some other media device in the media path. In such cases, SRTP should be terminated at the SD for each endpoint, so that the media server receives unencrypted media. In the other case, where the c-line is not subject to modification, Multi-system Selective SRTP Pass-through can be enabled.

If any of these conditions are not met, Multi-system Selective SRTP Pass-through processing cannot be provided, and the call is serviced as specified by SRTP Media Proxy Mode.

## Operational Overview

To set up Multi-system Selective SRTP Pass-through, the ingress and egress Oracle Communications Session Border Controllers (which can, in fact, be a single Oracle Communications Session Border Controller) exchange the SDES or MIKEY keying material that they receive from their respective endpoint so that the Oracle Communications Session Border Controller peer can pass the material to its adjacent endpoint. The endpoint to endpoint exchange of keying material enables the endpoints themselves to generate encryption/decryption keys.

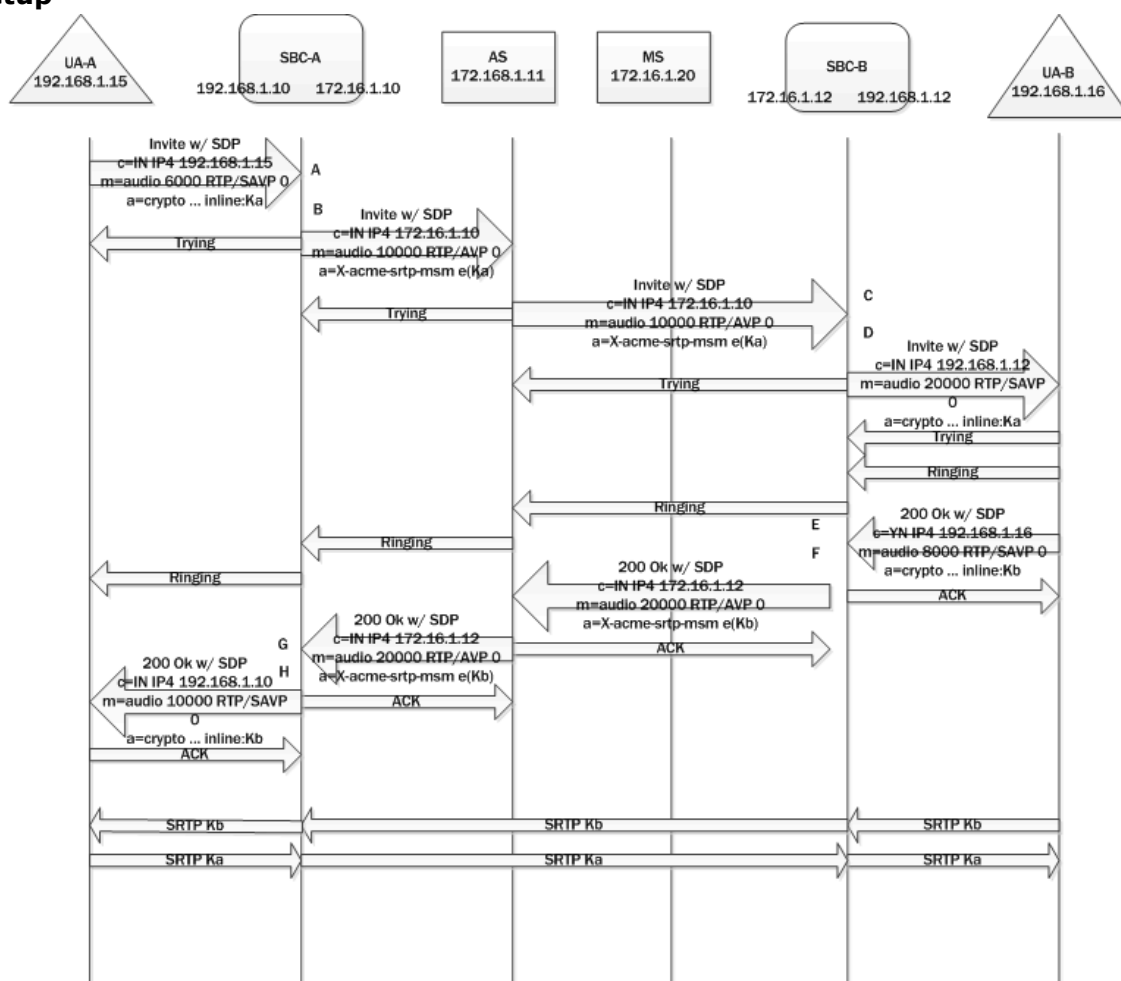
The actual exchange of keying material takes place in SIP messages (specifically, INVITE, 200 OK, and ACK) that carry offer or answer SDPs. Encrypted keying material is conveyed within a media attribute for each SRTP session. The name of the media attribute is configurable.

When either Oracle Communications Session Border Controller receives the encrypted keying material sent by its remote peer, it decrypts the media attribute and passes the plaintext attribute to its endpoint. Consequently, subsequent SRTP packets from the endpoints pass through the Oracle Communications Session Border Controllers without being decrypted and encrypted because both endpoints (now in possession of each others keying material) are able to decrypt the SRTP packets received from the other.

## Call Flows

The following three sections describe call signalling for common scenarios.

### Call Setup



192.168.1.15

A: The calling party sends an INVITE with SDP to SBC-A. The offer SDP contains an SDP crypto attribute within the SRTP media line.

B: Since Multi-system Selective SRTP Pass-through is enabled within the ingress realm, SBC-A adds an a=X-acme-srtp-msm media attribute. The a=X-acme-srtp-msm attribute contains a cookie that includes an encryption of the SDP crypto attribute present in the SDP. The encryption is done using the shared secret configured for encrypting SRTP Pass-through information.

C: SBC-B receives the offer SDP that has the cookie sent by SBC-A. It is assumed that the proxies that forward the offer SDP sent by SBC-A preserve and forward the cookie added by SBC-A.

D: SBC-B checks if the egress realm has Multi-system Selective SRTP Pass-through enabled. If so, SBC-B decrypts the cookie using the shared secret to retrieve the SDES crypto attribute. SBC-B adds the SDES crypto attribute retrieved from the cookie to the offer SDP sent to UA-B.

E: The called party sends an answer SDP with a SDES crypto attribute on the SRTP media line to SBC-B.

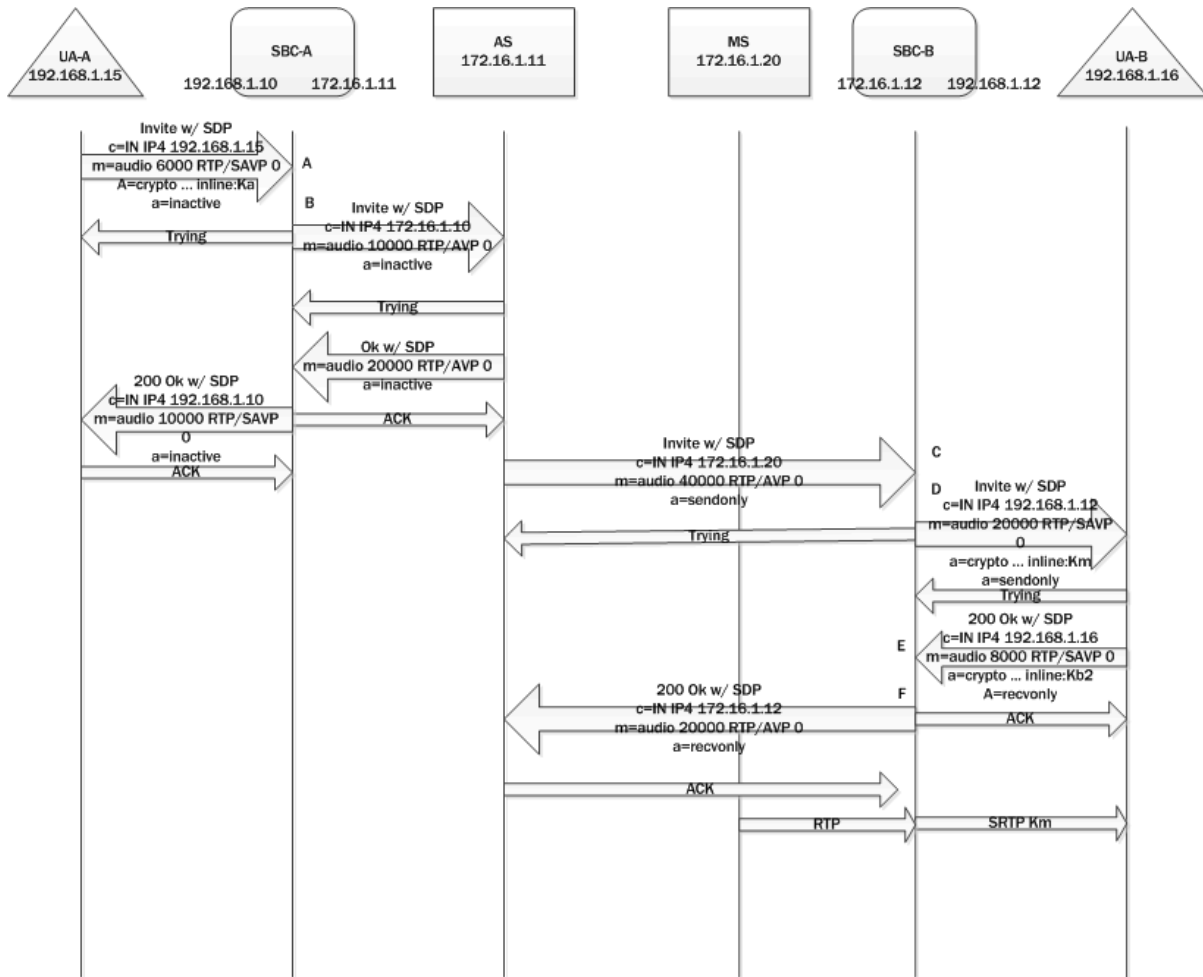
F: SBC-B checks if it has received a cookie in the offer SDP and adds the cookie to the answer SDP. The cookie contains an encryption of the SDES crypto attribute received in the answer SDP from UA-B. SBC-B does not install any SA or media policy so that SRTP packets from/to UA-B can pass through SBC-B without any decryption/encryption.

G: SBC-A receives the answer SDP that has the cookie sent by SBC-B.

H: SBC-A decrypts the cookie using the shared secret to retrieve the SDES crypto attribute. SBC-A adds the SDES crypto attribute retrieved from the cookie to the answer SDP. SBC-A does not install any SA or media policy so that SRTP packets from/to UA-A can pass through SBC-A without any decryption/encryption.

**Music on Hold**

After a call is established, one of the endpoints can put the other endpoint on hold. An application server might intercept the re-INVITE from one endpoint (for putting the other on hold) and implement MoH as follows.



A: Endpoint UA-A sends an offer SDP for hold to SBC-A.

B: SBC-A forwards offer SDP without any cookie since there will be no media going from/to UA-A.



C: SBC-B receives an offer SDP that is addressed to the MS without any cookie.

D: Because there is no cookie in the ingress offer SDP, SD-B generates its own SDES crypto attributes for the egress offer SDP sent to UA-B.

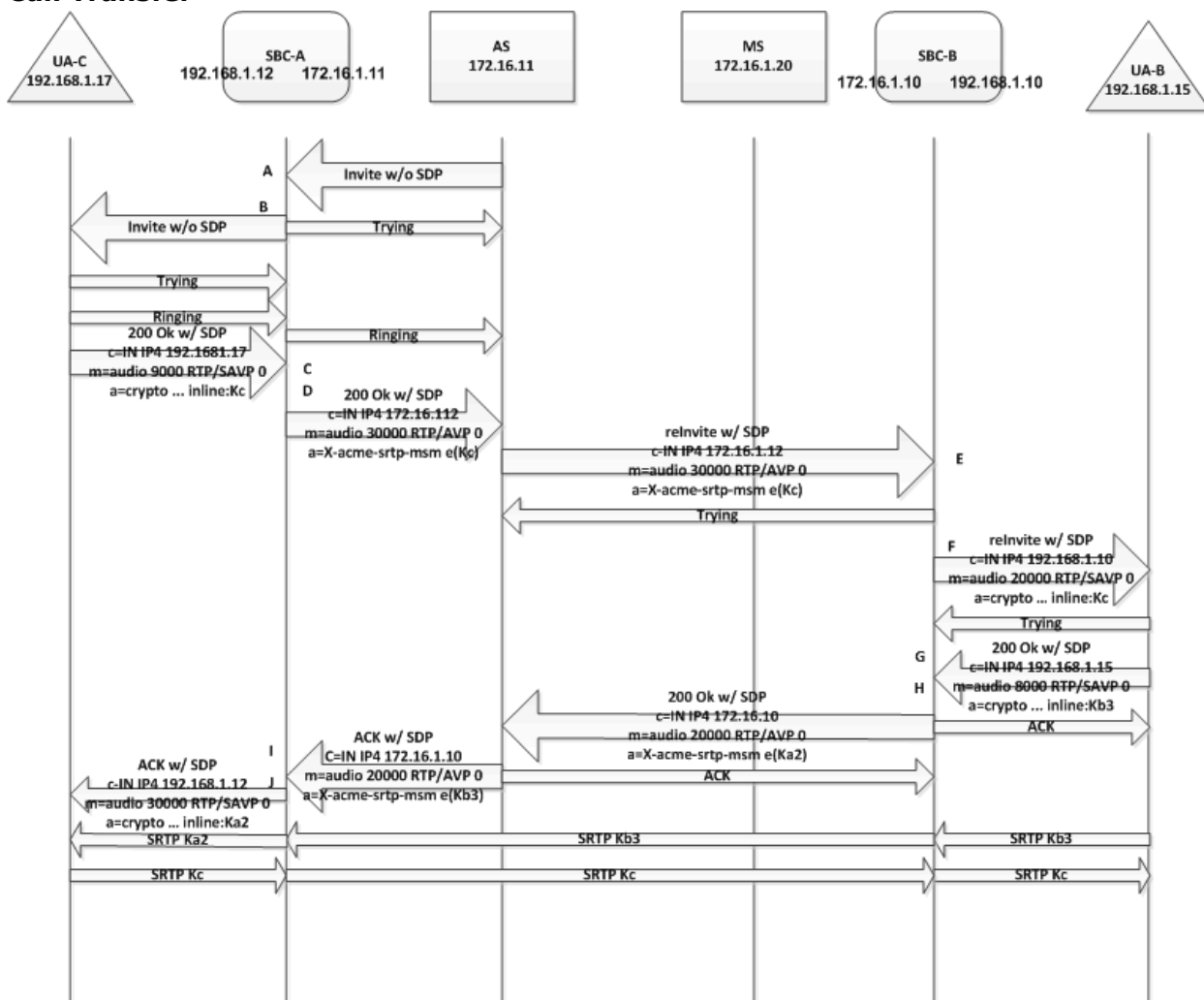
E: SBC-B receives an answer SDP from UA-B.

F: SBC-B sends its answer SDP without any cookie and encrypts SRTP packets going to UA-B. Note that there will be no SRTP packets going from UA-B to SBC-B.

As a result, MoH is heard by UA-B as it decrypts SRTP packets encrypted by SBC-B. When UA-A resumes, the steps to establish media between UA-A and UA-B will be the same as the steps used for call setup.

Once the media is reestablished between UA-A and UA-B media travelling between the two UAs will not be decrypted by either SBC.

### Call Transfer



A call transfer can also happen after a call is established. For example, endpoint UA-B can transfer UA-A to another endpoint, UA-C. UA-B can make a blind transfer or an attended transfer. The following diagram describes a blind call transfer with SRTP Pass-through enabled. Note it does not show the SIP message exchange between UA-B and the Application Server to facilitate the call transfer (i.e. the INVITE from UA-B to put the call on hold and the REFER/NOTIFY message exchange between UA-B and the Application Server). After UA-A is put on hold and the transfer target (that is, UA-C) is received from UA-B, the Application Server attempts to establish a call to UA-C. After the call to UA-C is established, the Application Server takes UA-A off hold and connects its media session to that of UA-C.

A: SBC-B receives an INVITE (from the Application Server to invite UA-C) without an offer SDP.

B: SBC-B sends an INVITE without an offer SDP to UA-C.

C: SBC-B receives a 200 OK response with an offer SDP that has a SDES crypto attribute on the SRTP media line from UA-C.

D: Since Multi-system Selective SRTP Pass-through is enabled within the ingress realm, SBC-B adds a cookie to the egress offer SDP that is sent to the core realm.

E: SBC-A receives a re-INVITE to UA-A with the offer SDP that has the cookie sent by SBC-B.

F: Since Multi-system Selective SRTP Pass-through is enabled within the ingress realm, SBC-A adds the SDES crypto attribute retrieved from the cookie to the offer SDP sent to UA-A.

G: SBC-A receives an answer SDP that has a SDES crypto attribute on the SRTP media line from UA-A.

H: SBC-A checks if it has a cookie in the offer SDP and adds the cookie to the answer SDP that is sent to the core realm. The cookie contains an encryption of the SDES crypto attribute received in the answer SDP from UA-B. SBC-A stops the encryption of any SRTP packets going to UA-B (set up for MoH) so that SRTP packets from/to UA-B can now pass through SBC-A without any decryption/encryption.

I: SBC-B receives the answer SDP that has the cookie sent by SBC-A.

H: SBC-B decrypts the cookie using the shared secret to retrieve the SDES crypto attribute. SBC-B adds the SDES crypto attribute retrieved from the cookie to the answer SDP.

After the call to UA-C is established and the call to UA-A is resumed (with the resulted media going between UA-A and UA-C) the Application Server disconnects the call with UA-A. Note that steps C-J are essentially the same steps as steps A-H in the Call Setup example. The difference is that the offer SDP from C comes to SD-B in a 200 OK response and the answer SDP sent by SBC-B to C is in the ACK.

## Early Media

Different application servers may implement early media in different ways. In general, the SBC supports early media in the following way.

If the SBC receives an SDP without any cookie in a provisional response, the SBC generates its own SRTP crypto context and exchanges it with the caller via the SDP included in the provisional response. The SBC continues to decrypt and encrypt early media packets going to and from the caller. The SBC stops decrypting and encrypting only if it receives a final response with an answer SDP that signals that SRTP Pass-through should be enabled.

## Multi-system Selective SRTP Pass-through with Media Release

When SRTP Pass-through is allowed, the hair-pinned media can also be released to the network if media release is configured — that is, if realm-config parameters **msm-release** is **enabled**, **mm-in-realm** is **disabled**, or **mm-in-network** is **disabled**. If SRTP Pass-through is not allowed, media release will not be allowed either.

## Multi-system Selective SRTP Pass-through Configuration

Use the following procedure to enable Multi-system Selective SRTP Pass-through within a specific realm.

1. Use the following command sequence to move to **realm-config** Configuration Mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

2. Use the **srtp-msm-passthrough** parameter to enable Multi-system Selective SRTP Pass-through within a specific realm.

By default, pass-through support is disabled.

```
ACMEPACKET(realm-config)# srtp-msm-passthrough enabled
ACMEPACKET(realm-config)#
```

3. Use **done**, **exit**, and **verify-config** to complete enabling Multi-system Selective SRTP Pass-through within the current realm.

**verify-config** checks that the **srtp-msm-password** parameter has been configured, and outputs an error if it has not been configured. **verify-config** also checks other configuration settings that conflict with Multi-system SRTP Pass-through operation. Among these possible mis-configurations are the following.

rfc2833-mode set to preferred on a SIP interface within a realm that has srtp-msm-passthrough enabled

rfc2833-mode set to preferred and app-protocol set to SIP on a session-agent within a realm that has srtp-msm-passthrough enabled.

4. If required, repeat Steps 1 through 3 to enable Multi-system Selective SRTP Pass-through on additional realms.

Use the following procedure to specify values need to support the exchange of SDES or MIKEY keying information.

5. Use the following command sequence to move to **security** Configuration Mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)#
```

6. Use the **srtp-msm-attr-name** parameter to specify the name of the media attribute used to convey SDES or MIKEY keying information within a SDP media description.

A valid attribute name must consist of characters from the US-ASCII subset of ISO-10646/UTF-8 as specified in RFC 2327, *SDP: Session Description Protocol*. IANA-registered names should not be used. Values should begin with an X-1 prefix to prevent collision with registered values.

In the absence of a specified attribute name, the SD provides a default value of X-acme-srtp-msm.

```
ACMEPACKET(security)# srtp-msm-attr-name X-key-material
ACMEPACKET(security)#
```

7. Use the **srtp-msm-password** parameter to provide the shared secret used to derive the key for encrypting SDES or MIKEY keying material that is placed in the media attribute of an SDP media description. Ingress keying material is encrypted using this shared secret before being forwarded to the network core. On egress, the encrypted keying material is decrypted with this same key.

Allowable values are characters strings that contain a minimum of 8 and a maximum of 16 characters.

```
ACMEPACKET(security)# srtp-msm-password IsHeEleemosynary
ACMEPACKET(security)#
```

8. Use **done**, **exit**, and **verify-config** to complete necessary configuration.

**verify-config** checks that the **srtp-msm-password** parameter has been configured, and outputs an error if it has not been configured. **verify-config** also checks other configuration settings that conflict with Multi-system SRTP Pass-through operation. Among these possible mis-configurations are the following.

rfc2833-mode set to preferred on a SIP interface within a realm that has srtp-msm-passthrough enabled

rfc2833-mode set to preferred and app-protocol set to SIP on a session-agent within a realm that has srtp-msm-passthrough enabled.

## Statistics

The number of media sessions set up with Multi-systems Selective SRTP Pass-through is tracked and included in the output of the **show mbcid statistics** command.

## IPSec Support


The Oracle Communications Session Border Controller offers IPSec for securing signaling, media, and management traffic at the network layer.

### Supported Protocols

The Oracle Communications Session Border Controller's IPSec implementation supports all required tools for securing Internet communication via the IPSec protocol suite. The following paragraphs list and explain the protocols within the IPSec suite that the Oracle Communications Session Border Controller supports. This chapter does not explain how to design and choose the best protocols for your application.

#### AH vs. ESP


The Oracle Communications Session Border Controller supports the two encapsulations that IPSec uses to secure packet flows. Authentication Header (AH) is used to authenticate and validate IP packets. Authentication means that the packet was sent by the source who is assumed to have sent it.

 **Note:** AH is incompatible with NAT. Validation means that the recipient is assured that the packet has arrived containing the original, unaltered data as sent.

ESP (Encapsulating Security Payload) provides AH's authentication and validations and extends the feature set by ensuring that the IP packet's contents remain confidential as they travel across the network. Using an encryption algorithm that both peers agree upon, ESP encrypts a full IP packet so that if intercepted, an unauthorized party cannot read the IPSec packet's contents.

#### Tunnel Mode vs. Transport Mode

In addition to its security encapsulations, the IPSec suite supports two modes: tunnel mode and transport mode. Tunnel mode is used most often for connections between gateways, or between a host and a gateway. Tunnel mode creates a VPN-like path between the two gateways and encapsulates the entire original IP packet. Transport mode is used to protect end-to-end communications between two hosts providing a secured IP connection and encrypts just the original payload.

 **Note:** Traffic sent through the inner IPSec tunnel must be on the same VLAN-slot-port network-interface combination as where the outer tunnel is configured. This is because IPSec tunnel mode does not carry any L2 information for the inner packet. Once the SBC decrypts (de-tunnel) the received packet, it uses the L2 header from the original packet for the lookup. Therefore, if the SBC uses different vlan/slot/port for the inner network, lookups will fail.

#### Cryptographic Algorithms

IPSec works by using a symmetric key for validation and encryption. Symmetric key algorithms use the same shared secret key for encoding and decoding data on both sides of the IPSec flow. The Oracle Communications Session Border Controller's IPSec feature supports the following encryption algorithms:

- DES
- 3DES
- AES128CBC
- AES256CBC
- AES128CTR
- AES256CTR

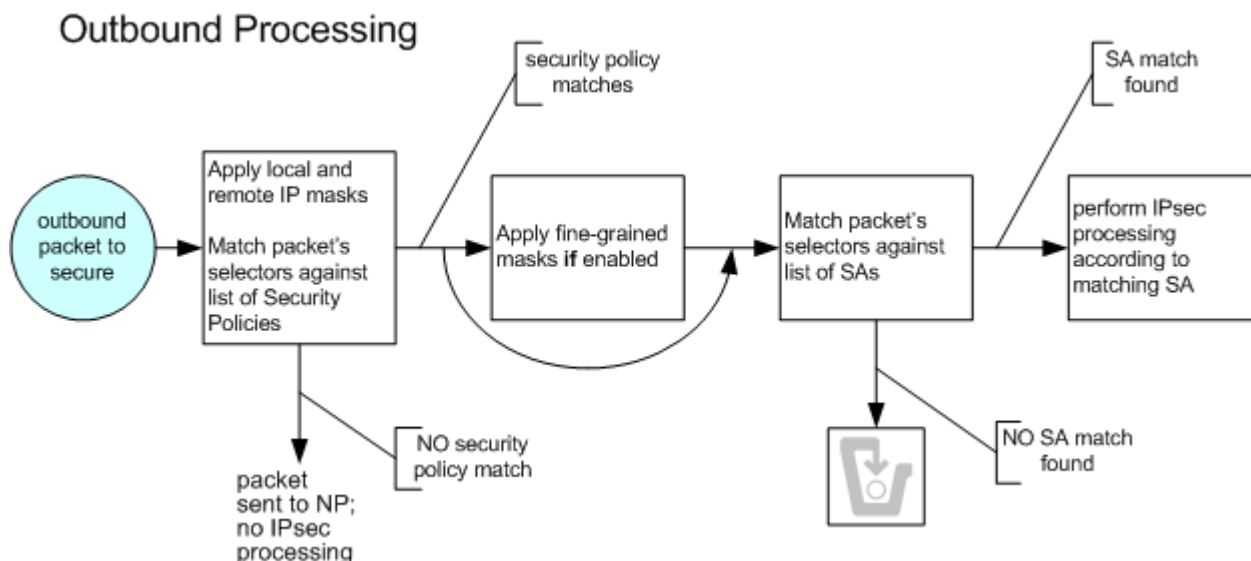
The Oracle Communications Session Border Controller can quickly generate keys for all of the above mentioned algorithms from the CLI. It can additionally support HMAC-SHA1 or HMAC-MD5 keyed-hash message authentication codes. Only manual keying is currently supported for both hash authentication and data encryption. Therefore, all keys must be provisioned on the Oracle Communications Session Border Controller by hand.

### IPSec Implementation

The Oracle Communications Session Border Controller uses separate logic for processing IPSec packets based on whether the traffic is inbound or outbound. The configuration is divided into two pieces, the security policy and the security association (SA). Both the SA and security policies have a directional attribute which indicates if they can be used and/or reused for inbound and outbound traffic.

## Outbound Packet Processing

The following diagrams show the steps the Oracle Communications Session Border Controller follows when processing outbound IPsec traffic. Details of each step are described in the following sections.



### Security Policy

The Oracle Communications Session Border Controller first performs a policy lookup on outbound traffic to test if it should be subjected to IPsec rules. A security policy, local policy applicable for IPsec functionality, defines the matching criteria for outgoing network traffic to secure. It is configured on a network interface basis.

Configuring a security policy is similar to a local policy, with additional IPsec-specific parameters. Unlike a local policy, used for routing, a security policy is used for packet treatment. As with a local policy, a set of selector values is matched against the outbound flow's following characteristics:

- VLAN
- Source IP address (plus mask)
- Source IP port
- Destination IP address (plus mask)
- Destination IP port
- Transport Protocol

Each of these selection criteria can be defined by a wildcard except for the VLAN ID, which can be ignored. This flexibility aids in creating selection criteria that ranges from highly restrictive to completely permissive. In addition to the main traffic matching criteria, a priority parameter is used to prioritize the order that configured security policies are checked against. The #0 policy is checked first, #1 policy is checked next, continuing to the lowest prioritized policy being checked last.

Once the outbound traffic matches a policy, the Oracle Communications Session Border Controller proceeds to the next step of outbound IPsec processing. If no matching security policy is found, the default pass-through policy allows the packet to be forwarded to the network without any security processing.

### Fine-grained policy Selection

After a positive match between outbound traffic and the configured selectors in the security policy, the Oracle Communications Session Border Controller can perform a calculation between a set of fine-grained packet selectors and the outbound packet. The fine-grained policy masking criteria are:

- Source IP subnet mask
- Destination IP subnet mask
- VLAN mask

By default, the fine-grained security policy is set to match and pass all traffic untouched to the security association (SA) portion of IPSec processing.

Fine-grained policy selection works by performing a logical AND between outbound traffic's fine-grained selectors and the traffic's corresponding attributes. The result is then used to find the matching SA. Applying a fine-grained mask has the effect of forcing a contiguous block of IP addresses and/or ports to appear as one address and or port. During the next step of IPSec processing, when an SA is chosen, the Oracle Communications Session Border Controller in effect uses one SA lookup for a series of addresses. Without fine-grained policy selection, unique SAs must always be configured for outbound packets with unique security policy selectors.

### Security Associations

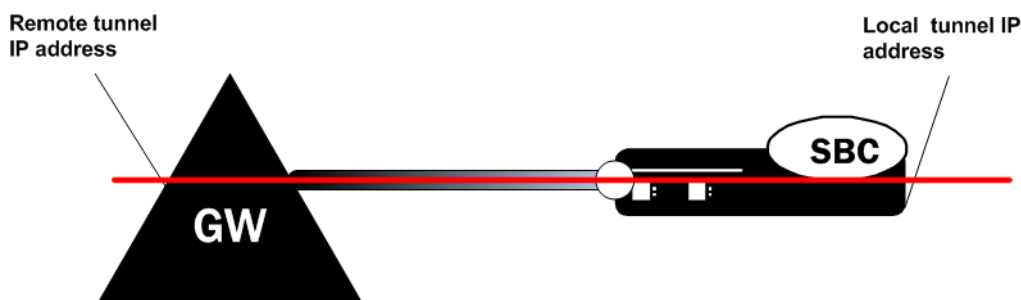
After the Oracle Communications Session Border Controller determines that outgoing traffic is subject to IPSec processing, and optionally applies fine-grained masking, an SA lookup is performed on the traffic. An SA is the set of rules that define the association between two endpoints or entities that create the secured communication. To choose an SA, the Oracle Communications Session Border Controller searches for a match against the outgoing traffic's SA selectors. SA selectors are as follows:

- VLAN
- Source IP address
- Source IP port
- Destination IP address
- Destination IP port
- Transport Protocol

If there is a match, the Oracle Communications Session Border Controller secures the flow according to security parameters defined in the SA that the Oracle Communications Session Border Controller chooses. The packet is then forwarded out of the Oracle Communications Session Border Controller. If no match is found, the packets are discarded, and optionally dumped to secured.log if the log-level is set to DEBUG.

### Secure Connection Details

Several parameters define an IPSec connection between the Oracle Communications Session Border Controller and a peer. When planning an IPSec deployment, the primary architectural decisions are which IPSec protocol and mode to use. The two choices for IPSec protocol are ESP or AH, and the two choices for IPSec mode are either tunnel or transport. IPSec protocol and mode are both required for an SA configuration. When creating an IPSec tunnel (tunnel mode), the SA must also define the two outside IP addresses of the tunnel.

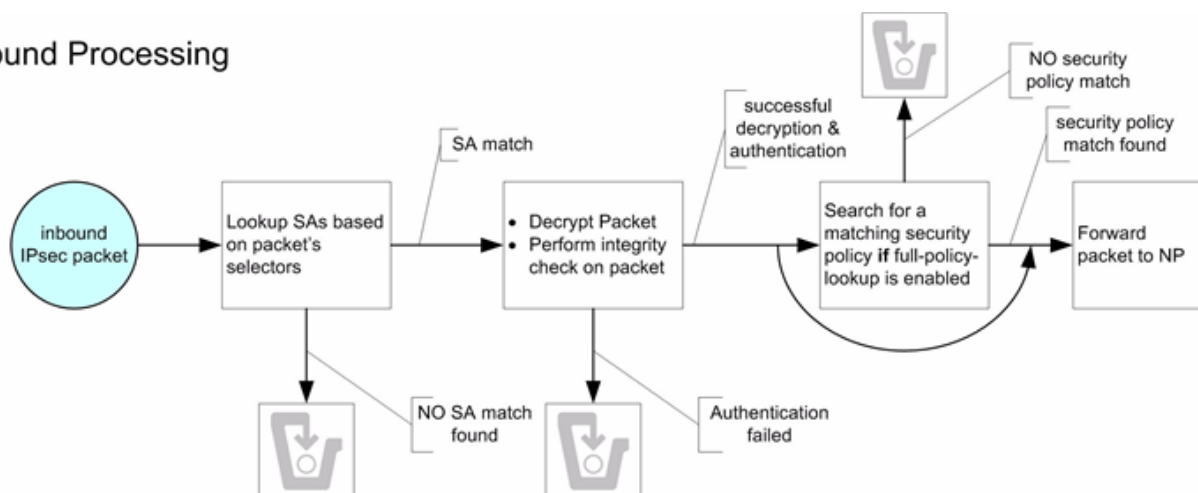


The authentication algorithm and the authentication key must always be configured. The Oracle Communications Session Border Controller supports hmac-md5 or hmac-sha1 authentication algorithms. Because only manual keying is supported, the key must be entered by hand. When encryption is required, the encryption algorithm and the encryption key must be configured. The Oracle Communications Session Border Controller supports des, 3des, aes-128-cbc, aes-256-cbc, aes-128-ctr, and aes-256-ctr encryption algorithms. When using the two encryption protocols that operate in AES counter mode (RFC 3686), an additional nonce value is required. In addition, the security parameter index (SPI) must be configured for each SA. All SPI values must be unique as well, across all SAs.

## Inbound Packet Processing

The following diagram shows the steps the system follows when processing inbound IPsec traffic. Details of each step are described in the following sections.

### Inbound Processing



### IP Header Inspection

Processing inbound IPsec packets begins by the Oracle Communications Session Border Controller inspecting an inbound IP packet's headers. If the packet is identified as IPsec traffic, as determined by the presence of an AH or ESP header, an SA policy lookup is performed. If the traffic is identified as non-IPsec traffic, as determined by the lack of an IPsec-type (AH or ESP) header, it still is subject to a policy lookup. However, due to the default allow policy, the packet is forwarded directly to the NP, without any security processing.

### SA Matching

The Oracle Communications Session Border Controller proceeds to match the inbound IPsec traffic's selectors against configured SAs. Inbound selector masking is performed where noted. These selectors are:

- VLAN (plus mask)
- Source IP address (plus mask)
- Source IP port
- Destination IP address (plus mask)
- Destination IP port
- Transport Protocol
- SPI

If no matching SA is found, the packets are discarded, and optionally dumped to `secured.log` if the log-level is set to `DEBUG`. When the Oracle Communications Session Border Controller finds a matching SA, the packet is authenticated and decrypted according to the configuration and sent to the Oracle Communications Session Border Controller's NP for continued processing.

### Inbound Full Policy Lookup

Inbound traffic can optionally be subjected to a full policy lookup, after decryption and authentication. A full policy lookup checks if a security policy exists for this inbound traffic before the Oracle Communications Session Border Controller sends the decrypted packet to the NP. If no matching security policy is found, even after a successful SA match, the packets are discarded, and optionally dumped to `secured.log` if the log-level is set to `DEBUG`.

Full policy lookups are useful for traffic filtering. If you wish to drop traffic not sent to a specific port (e.g. drop any traffic not sent to port 5060), a security policy with specific `remote-port-match` parameter would be used to define what is valid (i.e., not dropped).

## HA Considerations

Anti-replay mechanisms, running on IPSec peers, can cause instability with the Oracle Communications Session Border Controllers configured in an HA pair. The anti-replay mechanism ensures that traffic with inconsistent (non-incrementing) sequence numbers is labeled as insecure, assuming it could be part of a replay attack. Under normal circumstances, this signature causes the remote endpoint to drop IPSec traffic.

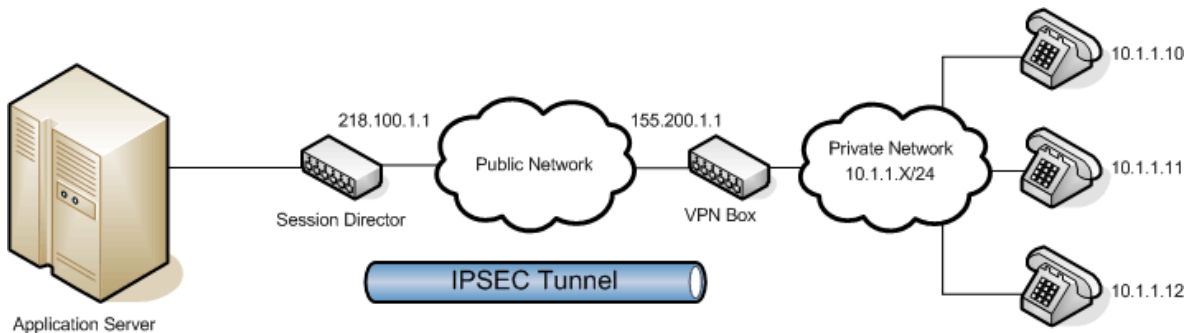
When a failover occurs between HA peers, the newly-active system starts sending traffic with the IPSec sequence number starting at 0. A remote system's anti-replay mechanism observes this and labels the traffic as insecure. It is therefore recommended that anti-replay protection not be used with Oracle Communications Session Border Controllers in an HA configuration. This situation does not create any problems as long as IPSec peers are not configured to use anti-replay mechanisms.

## Packet Size Considerations

The security processor supports receipt of jumbo frames up to 9K (9022 bytes with VLANs, 9018 without). Under normal operation the default outgoing maximum packet size of 1500 bytes is used. This packet size includes the IPSec headers, which will result in less space for packet data (SIP signaling, RTP, etc...).

## IPSec Application Example

In this example, the Oracle Communications Session Border Controller terminates an IPSec tunnel. The remote side of the tunnels is a dedicated VPN appliance in the public Internet. Behind that VPN appliance are three non-IPSec VoIP phones. In this scenario, the VPN box maintains the IPSec tunnel through which the phones communicate with the Oracle Communications Session Border Controller.



Without the fine-grained option (or alternatively IKE), an SA entry would need to be configured for each of the three phones, communicating over the IPSec tunnel (resulting in 3 tunnels).

This does not scale for manual-keying with a large number of endpoints. Using the fine-grained configuration as well as the inbound SA mask allows any number of endpoints on the 10.1.1.X network to use a single security association (a coarse-grain configuration). The configuration in this example follows:

A packet sent from the Oracle Communications Session Border Controller to any of the phones will match the policy `poll`. The `remote-ip-mask` parameter of the fine-grained configuration will then be masked against the `remote-ip`, resulting in a SA selector value of 10.1.1.0. This matches security-association `sa1`, and the packet will be secured and sent over the tunnel. The tunnel-mode addresses in the security-association represent the external, public addresses of the termination points for the IPSec tunnel.

Packets returning from the 10.1.1.0 side of the IPSec tunnel will first match the tunnel-mode `local-ip-addr` of 218.100.1.1. The packets will then be decrypted using the SA parameters, and the tunneled packet will be checked against the `remote-ip-addr` field of the SA.

If the fine-grained mask had not been used, three discrete SAs would have to be configured: one for each of the three phones.

```
ACMEPACKET (manual) # show
manual
      name          assoc1
      spi           1516
```




```

network-interface      lefty:0
local-ip-addr          100.20.50.7
remote-ip-addr        100.25.56.10
local-port             60035
remote-port           26555
trans-protocol         ALL
ipsec-protocol         esp
direction              both
ipsec-mode             tunnel
auth-algo              hmac-md5
encr-algo              des
auth-key
encr-key
aes-ctr-nonce          0
tunnel-mode
    local-ip-addr      100.20.55.1
    remote-ip-addr     101.22.54.3
last-modified-date    2007-04-30 16:04:46

```

## IPSec Configuration

The following example explains how to configure IPSec on your Oracle Communications Session Border Controller.

 **Note:** If you change the phy-interface slot and port associated with any SAs or SPDs, the Oracle Communications Session Border Controller must be rebooted for the changes to take effect.

### Configuring an IPSec Security Policy

To configure an IPSec security policy:

1. In Superuser mode, type **configure terminal** and press Enter.

```

ACMEPACKET# configure terminal
ACMEPACKET(configure)#

```

2. Type **security** and press Enter to access the security path of the configuration menu.

```

ACMEPACKET(configure)# security
ACMEPACKET(security)#

```

3. Type **ipsec** and press Enter.

```

ACMEPACKET(security)# ipsec
ACMEPACKET(ipsec)#

```

4. Type **security-policy** and press Enter. The prompt changes to let you know that you can begin configuration.

```

ACMEPACKET(ipsec)# security-policy
ACMEPACKET(security-policy)#

```

5. **name**—Enter a name for this security policy. This parameter is required and has no default.
6. **network-interface**—Enter the network interface and VLAN where this security policy applies in the form: interface-name:VLAN
7. **priority**—Enter the priority number of this security policy. The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—254
8. **action**—Enter the action the Oracle Communications Session Border Controller should take when this policy matches outbound IPSec traffic. The default value is **ipsec**. The valid values are:
  - **ipsec**—Continue processing as IPSec traffic
  - **allow**—Forward the traffic without any security processing
  - **discard**—Discard the traffic

9. **direction**—Enter the direction of traffic this security policy can apply to. The default value is **both**. The valid values are:

- **in**—This security policy is valid for inbound traffic
- **out**—This security policy is valid for outbound traffic
- **both**—This security policy is valid for inbound and outbound traffic

To define the criteria for matching traffic selectors for this security policy:

10. **local-ip-addr-match**—Enter the source IP address to match. The default value is **0.0.0.0**.

11. **remote-ip-addr-match**—Enter the destination IP address to match. The default value is **0.0.0.0**.

12. **local-port-match**—Enter the source port to match. A value of 0 disables this selector. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—65535

13. **remote-port-match**—Enter the destination port to match. A value of 0 disables this selector. The default value is zero (0). The valid range is:

- Minimum—0
- Maximum—65535

14. **trans-protocol-match**—Enter the transport protocol to match. The default value is **all**. The valid values are:

- **UDP | TCP | ICMP | ALL**

15. **local-ip-mask**—Enter the source IP address mask, in dotted-decimal notation. The default value is **255.255.255.255**.

16. **remote-ip-mask**—Enter the remote IP address mask, in dotted-decimal notation. The default value is **255.255.255.255**.

17. Save your work using the ACLI **done** command.

### Defining Outbound Fine-Grained SA Matching Criteria

To define outbound fine-grained SA matching criteria:

1. From within the security policy configuration, type **outbound-sa-fine-grained-mask** and press Enter. The prompt changes to let you know that you can begin configuration.

2. **local-ip-mask**—Enter the fine-grained source IP address mask to apply to outbound IP packets for SA matching. Valid values are in dotted-decimal notation. The default mask matches for all traffic.

3. **remote-ip-mask**—Enter the fine-grained destination IP address mask to apply to outbound IP packets for SA matching. Valid values are in dotted-decimal notation. The default mask matches for all traffic.

4. **local-port-mask**—Enter the local port mask for this security policy. The default value for this parameter is 0. The valid range is:

- Minimum—0
- Maximum—65535

5. **remote-port-mask**—Enter the remote port mask for this security policy. The default value for this parameter is 0. The valid range is:

- Minimum—0
- Maximum—65535

6. **trans-protocol-mask**—Enter the transport protocol mask for this security policy. The default value for this parameter is 0. The valid range is:

- Minimum—0
- Maximum—255

7. **vlan-mask**—Enter the fine-grained VLAN mask to apply to outbound IP packets for SA matching. The default is **0x000 (disabled)**. The valid range is:
  - 0x000 - 0xFFF
8. Save your work using the ACLI **done** command.

### Configuring an IPsec SA

To configure an IPsec SA:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **security** and press Enter to access the security path of the configuration menu.

```
ACMEPACKET (configure) # security
ACMEPACKET (security) #
```

3. Type **ipsec** and press Enter.

```
ACMEPACKET (security) # ipsec
ACMEPACKET (ipsec) #
```

4. Type **security-association** and press Enter.

```
ACMEPACKET (ipsec) # security-association
ACMEPACKET (security-association) #
```

5. Type **manual** and press Enter. The prompt changes to let you know that you can begin configuration.

```
ACMEPACKET (security-association) # manual
ACMEPACKET (manual) #
```

6. **name**—Enter a name for this security policy.
7. **network-interface**—Enter the network interface and VLAN where this security policy applies in the form: `interface_name:VLAN`
8. **direction**—Enter the direction of traffic this security policy can apply to. The default value is **both**. Valid values are:
  - in | out | both
9. Save your work using the ACLI **done** command.



### Defining Criteria for Matching Traffic Selectors per SA

To define the criteria for matching traffic selectors for this SA:

1. From within the **manual** portion of the security association configuration, you need to set the parameters described in this process.

```
ACMEPACKET (security-association) # manual
ACMEPACKET (manual) #
```

2. **local-ip-addr**—Enter the source IP address to match.
3. **remote-ip-addr**—Enter the destination IP address to match.
4. **local-port**—Enter the source port to match. A value of **0** disables this selector. The default value is **0**, disabling this parameter. The valid range is:
  - Minimum—0
  - Maximum—65535
5. **remote-port**—Enter the destination port to match. A value of **0** disables this selector. The default value is **0**, disabling this parameter. The valid range is:
  - Minimum—0
  - Maximum—65535

6. **trans-protocol**—Enter the transport protocol to match for traffic selectors for this SA. The default value is **ALL**. The valid values are:
  - **UDP | TCP | ICMP | ALL**
7. **ipsec-protocol**—Select the IPsec protocol to use for this SA configuration. The default value for this parameter is **esp**. Valid values are:
  - **esp | ah**
8. **spi**—Enter the security parameter index. The default value is **256**. The valid range is:
  - Minimum—256
  - Maximum—2302
9. **ipsec-mode**—Enter the IPsec mode of this SA. The default value is **transport**. The valid values are:
  - **tunnel | transport**
10. **auth-algo**—Enter the IPsec authentication algorithm for this SA. The default value is **null**. The valid values are:
  - **hmac-md5 | hmac-sha1 | null**
11. **auth-key**—Enter the authentication key for the previously chosen authentication algorithm for this SA.
  -  **Note:** The specified auth-key value will be encrypted in the configuration and will no longer be visible in clear-text.
12. **encr-algo**—Enter the IPsec encryption algorithm for this SA. The default value is **null**. The valid values are:
  - **des | 3des | aes-128-cbc | aes-256-cbc | aes-128-ctr | aes-256-ctr | null**
13. **encr-key**—Enter the encryption key for the previously chosen encryption algorithm for this SA.
  -  **Note:** The specified encr-key value will be encrypted in the configuration and will no longer be visible in clear-text.
14. **aes-ctr-nonce**—Enter the AES nonce if aes-128-ctr or aes-256-ctr were chosen as your encryption algorithm. The default value is **0**.

### Defining Endpoints for IPsec Tunnel Mode

To define endpoints for IPsec tunnel mode:

1. From within the **manual** portion of the security association configuration, you need to set the parameters described in this process.

```
ACMEPACKET (security-association) # manual
ACMEPACKET (manual) #
```

2. **local-ip-addr**—Enter the local public IP address which terminates the IPsec tunnel.
3. **remote-ip-addr**—Enter the remote public IP address which terminates the IPsec tunnel.
4. Save your work using the ACLI **done** command.

### Real-Time IPsec Process Control

The **notify secured** commands force the IPsec application to perform tasks in real-time, outside of the Oracle Communications Session Border Controller reloading and activating the running configuration. The **notify secured** usage is as follows:

```
notify secured [activateconfig | nolog | log | debug | nodebug]
```

The following arguments perform the listed tasks:

- **nolog**—Disables secured logging
- **log**—Enables secured logging
- **debug**—Sets secured log level to DEBUG

- **nodebug**—Sets secured log level to INFO

## Key Generation

The **generate-key** command generates keys for the supported encryption or authentication algorithms supported by the Oracle Communications Session Border Controller's IPSec implementation. The generate-key commands generate random values which are not stored on the Oracle Communications Session Border Controller, but are only displayed on the screen. This command is a convenient function for users who would like to randomly generate encryption and authentication keys. The generate-key usage is as follows:

```
generate-key [hmac-md5 | hmac-sha1 | aes-128 | aes-256 | des | 3des]
```

## IDS Reporting

The Oracle Communications Session Border Controller supports a wide range of intrusion detection and protection capabilities for vulnerability and attack profiles identified to date. The IDS reporting feature is useful for enterprise customers requirement to report on intrusions and suspicious behavior that it currently monitors.

## IDS Licensing

This feature requires the IDS Reporting license.



**Note:** The following capabilities and restrictions of the license:

- The following configuration parameters located in the **access control** and **media manager configuration** elements are only visible after installing the license:
  - trap-on-demote-to-deny
  - syslog-on-demote-to-deny
  - cac-failure-threshold
  - untrust-cac-failure-threshold
- Endpoint demotions based on admission control failures are only a valid option with the IDS License.
- The presence of the IDS license makes the apSysMgmtInetAddrWithReasonDOSTrap trap available and the apSysMgmtExpDOSTrap unavailable. Without an IDS licence installed, only the apSysMgmtExpDOSTrap trap is available.
- The **Trust->Untrust** and **Untrust-Deny** counters in the SIP ACLs' statistics are visible regardless of the IDS license's presence.
- The **Demote Trust-Untrust** and **Demote Untrust-Deny** collect records in the SIP ACL HDR groups are visible regardless of the IDS license's presence.
- A GET operation can be preformed on the two MIB entries to view the global endpoint counter for Demotions from Trusted to Untrusted and from Untrusted to Deny regardless of the IDS license's presence
- On Acme Packet 3820 systems, the DOS license must be installed in addition to the IDS license in order to enable all features described in this section.

## Basic Endpoint Demotion Behavior

Each session agent or endpoint is promoted or demoted among the trusted, untrusted, and denied queues depending on the **trust-level** parameter of the session agent or realm to which it belongs. Users can also configure access control rules to further classify signaling traffic so it can be promoted or demoted among trust queues as necessary.

An endpoint can be demoted in two cases:

1. Oracle Communications Session Border Controller receiving too many signaling packets within the configured time window (**maximum signal threshold** in **realm config** or **access control**)
2. Oracle Communications Session Border Controller receiving too many invalid signaling packets within the configured time window. (**invalid signal threshold** in **realm config** or **access control**)

## Endpoint Demotion Reporting

The Oracle Communications Session Border Controller counts the number of endpoint or session agent promotions and demotions. Further, the Oracle Communications Session Border Controller counts when endpoints or session agents transition from trusted to untrusted and when endpoints transition from untrusted to denied queues. These counts are maintained for SIP signaling applications. They appear as the Trust->Untrust and Untrust->Deny rows in the **show sipd acls** command.

### SNMP Reporting

These per-endpoint counters are available under APSYSMGMT-MIB -> acmepacketMgmt -> apSystemManagementModule -> apSysMgmtMIBObjects -> apSysMgmtMIBGeneralObjects.

MIB NAME	MIB OID	PURPOSE
apSysSipEndptDemTrustToUntrust	.1.3.6.1.4.1.9148.3.2.1.1.19	Global counter for SIP endpoint demotions from trusted to untrusted.
apSysSipEndptDemUntrustToDeny	.1.3.6.1.4.1.9148.3.2.1.1.20	Global counter for SIP endpoint demotions from untrusted to denied.

### HDR Reporting

The SIP (sip-ACL-oper) HDR ACL status collection groups include the following two metrics:

- Demote Trust-Untrust (Global counter of endpoint demotion from trusted to untrusted queue)
- Demote Untrust-Deny (Global counter of endpoint demotion from untrusted to denied queue)

## Endpoint Demotion SNMP Traps

An SNMP trap can be sent when the Oracle Communications Session Border Controller demotes an endpoint to the denied queue. This is set by enabling the **trap on demote to deny** parameter located in the **media manager config** configuration element.

When the IDS license is installed and the **trap on demote to deny** parameter is enabled, apSysMgmtInetAddrWithReasonDOSTrap trap is sent. This trap supersedes the apSysMgmtInetAddrDOSTrap trap.

When the IDS license is installed and the **trap on demote to deny** parameter is disabled the apSysMgmtInetAddrWithReasonDOSTrap trap is not sent from the Oracle Communications Session Border Controller, even when an endpoint is demoted to the denied queue.

This apSysMgmtInetAddrWithReasonDOSTrap contains the following data:

- apSysMgmtDOSInetAddressType—Blocked IP address family (IPv4 or IPv6)
- apSysMgmtDOSInetAddress—Blocked IP address
- apSysMgmtDOSRealmID—Blocked Realm ID
- apSysMgmtDOSFromURI—The FROM header of the message that caused the block (If available)
- apSysMgmtDOSReason—The reason for demoting the endpoint to the denied queue: This field can report the following three values:
  - Too many errors
  - Too many messages
  - Too many admission control failures



**Note:** By default, this parameter is enabled for upgrade configurations, as the current behavior is to send a trap for every endpoint that is demoted to deny. However, for a new configuration created, the value to this configuration is disabled.

### Trusted to Untrusted Reporting

Endpoints, however, transition to an intermediate state, untrusted, prior to being denied service. The Oracle Communications Session Border Controller provides an ACLI parameter, **trap-on-demote-to-untrusted**,

that generates an SNMP trap when a previously trusted endpoint transitions to the untrusted state. Trap generation is disabled by default.

## SNMP Reporting

Endpoint state transitions continue to be tracked by two counters available under APSYSMGMT-MIB -> acmepacketMgmt -> apSystemManagementModule -> apSysMgmtMIBObjects -> apSysMgmtMIBGeneralObjects.

MIB NAME	MIB OID	PURPOSE
apSysSipEndptDemTrustToUntrust	.1.3.6.1.4.1.9148.3.2.1.1.19	Global counter for SIP endpoint demotions from trusted to untrusted.
apSysSipEndptDemUntrustToDeny	.1.3.6.1.4.1.9148.3.2.1.1.20	Global counter for SIP endpoint demotions from untrusted to denied.

## Endpoint Demotion Trusted-to-Untrusted SNMP Trap

The system can generate an SNMP trap when an endpoint transitions from the trusted to the untrusted state. The trap is structured as follows.

```
apSysMgmtInetAddrTrustedToUntrustedDOSTrap NOTIFICATION-TYPE
OBJECTS { apSysMgmtDOSInetAddressType,
apSysMgmtDOSInetAddress,
apSysMgmtDOSRealmID,
apSysMgmtDOSFromUri,
apSysMgmtDOSReason }
STATUS current
DESCRIPTION
"This trap is generated when an IP is placed on a untrusted list from trusted
list, and provides the ip address that has been demoted, the realm-id of that
IP, (if available) the URI portion of the SIP From header of the message that
caused the demotion."
 ::= { apSysMgmtDOSNotifications 5 }
```

The trap OID is 1.3.6.1.4.1.9148.3.2.8.0.5.

## Endpoint Demotion Syslog Message

A Syslog message can be generated when an endpoint is demoted. Setting the **media manager config -> syslog-on-demote-to-deny** parameter to enabled writes and endpoint demotion warning to the syslog every time an endpoint is demoted to the denied queue. By default, this configuration option is set to disabled. The syslog message has a WARNING Level and looks like this:

```
Jan 15 12:22:48 172.30.60.12 ACMESYSTEM sipd[1c6e0b90] WARNING SigAddr[access:
168.192.24.40:0=low:DENY] ttl=3632 guard=798 exp=30 Demoted to Black-List
(Too many admission control failures)
```

## Event Log Notification Demotion from Trusted to Untrusted

You can enable your Oracle Communications Session Border Controller to provide event log notification (a syslog message) any time it demotes an endpoint from trusted to untrusted. The log message contains this data: IP address of the demoted endpoint, the endpoint's configured trust level, and the reason for demotion.

To use this feature, the intrusion detection system (IDS) Reporting license must be installed and your media manager configuration must be enabled to send the log messages. This feature is enabled with the **syslog-on-demote-to-untrusted** parameter in the media manager.

## Endpoint Demotion Configuration

To configure the Oracle Communications Session Border Controller to send traps and/or write syslog messages on endpoint demotion:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-level configuration elements.

```
ACMEPACKET (configure) # media-manager  
ACMEPACKET (media-manager) #
```

3. Type **media-manager** and press Enter.

```
ACMEPACKET (media-manager) # media-manager  
ACMEPACKET (media-manager-config) #
```

4. **trap-on-demote-to-deny**—Set this parameter to enabled for the Oracle Communications Session Border Controller to send the apSysMgmtInetAddrWithReasonDOSTrap trap when applicable.
5. **syslog-on-demote-to-deny**—Set this parameter to enabled for the Oracle Communications Session Border Controller to write an endpoint demotion warning message to the syslog.
6. **syslog-on-demote-to-untrusted**—Change this parameter from **disabled** (default), to **enabled** so the Oracle Communications Session Border Controller will generate event notifications (syslog messages) when an endpoint becomes untrusted. For this capability to work, the IDS license must be installed on your system.
7. **trap-on-demote-to-untrusted**—Set this parameter to enabled for the Oracle Communications Session Border Controller to send the apSysMgmtInetAddrTrustedToUntrustedDOSTrap when the endpoint identified within the trap transitions from the trusted to untrusted state.
8. Save your work.

## Endpoint Demotion due to CAC overage

The Oracle Communications Session Border Controller can demote endpoints from trusted to untrusted queues when CAC failures exceed a configured threshold. The Oracle Communications Session Border Controller can also demote endpoints from untrusted to denied queues when CAC failures exceed a another configured threshold.

The Oracle Communications Session Border Controller maintains CAC failures per-endpoint. The CAC failure counter is incremented upon certain admission control failures only if either one of **cac-failure-threshold** or **untrust-cac-fail-threshold** is non-zero.

The **cac failure threshold** parameter is available in the access control and realm configuration elements. Exceeding this parameter demotes an endpoint from the trusted queue to the untrusted queue. The **untrust cac-failure-threshold** parameter is available in the access control and realm configuration elements. Exceeding this parameter demotes an endpoint from the untrusted queue to the denied queue.

If both the **cac failure threshold** and **untrusted cac failure threshold** are configured to 0, then admission control failures are considered and counted as invalid signaling messages for determining if the **invalid-signal-threshold** parameter value has been exceeded.

### CAC Attributes used for Endpoint Demotion

The Oracle Communications Session Border Controller determines CAC failures only by considering the calling endpoint's signaling messages traversing the calling realms' configuration. If an endpoint exceeds the following CAC thresholds, the Oracle Communications Session Border Controller will demote the endpoint when the CAC failure thresholds are enabled.

1. sip-interface user CAC sessions (realm-config > user-cac-sessions)
2. sip-interface user CAC bandwidth (realm-config > user-cac-bandwidth)
3. External policy server rejects a session

### Authentication Failures used for Endpoint Demotion

If an endpoint fails to authenticate with the Oracle Communications Session Border Controller using SIP HTTP digest authentication OR endpoint fails authentication with an INVITE with authentication incase registration-caching is disabled, and receives back a 401 or 407 response from the registrar



When the Oracle Communications Session Border Controller receives a 401 or 407 message from the registrar in response to one of the following conditions, the endpoint attempting authentication is demoted.

- endpoint fails to authenticate with the Oracle Communications Session Border Controller using SIP HTTP digest authentication
- endpoint fails to authenticate with the Oracle Communications Session Border Controller using INVITE message when registration-caching is disabled

## Endpoint Demotion Configuration on CAC Failures

To configure endpoint demotion on CAC failures:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **access-control** and press Enter.

```
ACMEPACKET(session-router)# access-control
ACMEPACKET(access-control)#
```

If you are adding this feature to an existing configuration, then you will need to select the configuration you want to edit.

4. **cac-failure-threshold**—Enter the number of CAC failures for any single endpoint that will demote it from the trusted queue to the untrusted queue.
5. **untrust-cac-failure-threshold**—Enter the number of CAC failures for any single endpoint that will demote it from the untrusted queue to the denied queue.
6. Save your work.

## IDS Phase 2 (Advanced Reporting)

This feature supplements the IDS reporting and protection services. IDS Phase 2 provides enterprise users with additional tools to identify, monitor, and control suspicious, and possibly, malicious traffic patterns. IDS Phase 2 requires the IDS Advanced Reporting license.

### License Requirements

IDS Phase 2 requires the IDS Advanced Reporting license, which is different from the IDS license introduced in the Net-Net S-C6.2.0 release. Access to new services described in this Technical Notice requires that the original IDS license be upgraded to the IDS Advanced Reporting license.

### Rejected SIP Calls

IDS Phase 2 provides tools to monitor and record rejected SIP calls. A sudden or gradual increase in such calls can, but need not, indicate malicious intent.

IDS Phase 2 provides a global counter that increments with each SIP INVITE or REGISTER message that is rejected by the Acme Packet Oracle Communications Session Border Controller, and offers the option of generating a syslog message in response to call rejection.

### Rejected Calls Counter

The rejected calls counter is a 32-bit global counter that records the total number of rejected SIP calls. Such calls have been rejected by the Oracle Communications Session Border Controller with the following response codes: 400, 403,

404, 405, 408, 413, 416, 417, 420, 423, 480, 481, 483, 484, 485, 488, 494, 500, 503, 505, and 604. These response codes may change with future software revisions.

The current value of the rejected calls counter is accessible via SNMP, Historical Data Recording (HDR), or the ACLI.

apSysMgmtGeneralObjects Table (1.3.6.1.4.1.9148.3.2.1.1)		
Object Name	Object OID	Description
apSysSipTotalCallsRejected	1.3.6.1.4.1.9148.3.2.1.1.25	Global counter for SIP calls that are rejected by the SBC

The sip-error HDR collection group contains a new reporting field, Call Rejects, which contains the value of the global rejected calls counter.

The ACLI command **show sipd errors** displays the contents of the rejected calls counter.

```
ACMEPACKET# show sipd errors
12:29:13-131
SIP Errors/Events          ----- Lifetime -----
                          Recent      Total    PerMax
SDP Offer Errors           0         0        0
SDP Answer Errors         0         0        0
Drop Media Errors         0         0        0
Transaction Errors        0         0        0
Application Errors        0         0        0
Media Exp Events          0         0        0
Early Media Exps          0         0        0
Exp Media Drops            0         0        0
Expired Sessions           0         0        0
Multiple OK Drops         0         0        0
Multiple OK Terms         0         0        0
Media Failure Drops       0         0        0
Non-ACK 2xx Drops         0         0        0
Invalid Requests          0         5         2
Invalid Responses         0         0        0
Invalid Messages          0         0        0
CAC Session Drop          0         0        0
Nsep User Exceeded        0         0        0
Nsep SA Exceeded          0         0        0
CAC BW Drop               0         0        0
Calls Rejected            0         0        0 <--
```

### Syslog Reporting of Rejected Calls

Users can choose to send a syslog message in response to the rejection of a SIP call. In the default state, rejected calls are not reported to syslog.

Use the following ACLI command sequence to enable syslog reporting of rejected SIP calls.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# media-manager
ACMEPACKET(media-manager-config)# syslog-on-call-reject enable
```

The syslog-on-call-reject attribute, which is disabled by default, enables the generation of a syslog message in response to the rejection of a SIP call.

Use **done**, **exit**, and **verify-config** to complete this configuration.

Syslog messages issued in response to call rejection contain the following call-related information.

- SIP status code indicating rejection cause
- SIP method name (INVITE or REGISTER)

- Reason for denial
- Realm of calling endpoint
- Applicable local response map
- Content of Reason header (if present)
- From URI of calling endpoint
- Target URI of called endpoint
- Source and Destination IP address and port
- Transport type

The following are sample syslog messages issued in response to call rejections.

```
Dec 8 06:05:42 172.30.70.119 deimos sipd[205bfee4] ERROR [IDS_LOG]INVITE from source 172.16.18.100:5060
to dest 172.16.101.13:5060[UDP] realm=net172; From=sipp
<sip:sipp@172.16.18.100:5060>;tag=13890SIPpTag001; target=sip:service@172.16.101.13:5060 rejected!;
status=483 (Too Many Hops)
```

```
Dec 10 15:09:28 172.30.70.119 deimos sipd[2065ace8] ERROR [IDS_LOG]INVITE from source 172.16.18.5:5060
to dest 172.16.101.13:5060[UDP] realm=net172; From=sipp <sip:sipp@172.16.18.5:5060>;tag=10015SIPpTag001;
target=sip:service@172.16.101.13:5060 rejected!; status=488 (sdp-address-mismatch); error=sdp address mismatch
```

IDS syslog messages that report rejected calls and those that report endpoint demotions now contain a string `IDS_LOG`, to facilitate their identification as IDS-related messages. With IDS Phase 2, IDS messages reporting either endpoint demotions or call rejections can be sent to specific, previously-configured syslog servers.

In topologies that include multiple syslog servers, use the following procedure to enable delivery of IDS-related messages to one or more specific syslog servers.

1. Use the following command sequence to move to **syslog-config** Configuration Mode.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) # system
ACMEPACKET (system) # system-config
ACMEPACKET (system-config) # syslog-servers
ACMEPACKET (syslog-config) #
```

2. From the existing pool of syslog servers select the server or servers that will receive syslog messages.
3. Ensure that all selected servers are configured with the same value for the **facility** attribute.

Allowable values are integers within the range 0 through 23.

4. Use the following command sequence to move to **system-config** Configuration Mode.

```
ACMEPACKET (syslog-config) # done
ACMEPACKET (syslog-config) # exit
ACMEPACKET (system-config) #
```

5. Use the **ids-syslog-facility** attribute to enable message transfer to specific syslog servers.

The default value, -1, disables selective message transfer. To enable transfer to a designated syslog server or servers, enter the facility value (an integer within the range 0 through 23) that you confirmed or set in Step 3.

The following example enables the transfer of IDS syslog messages to all servers with a facility value of 16.

```
ACMEPACKET (system-config) # ids-syslog-facility 16
ACMEPACKET (system-config) #
```

6. Use **done**, **exit**, and **verify-config** to complete this configuration.

## TCA Reporting of Denied Entries

Users can construct a Threshold Crossing Alarm (TCA) which issues minor, major, and critical system alarms when the count of denied entries exceeds pre-configured values. For each issued alarm, the TCA also transmits an SNMP trap that reports the alarm state to remote SNMP agents.

1. Use the following command sequence to move to **media-manager-config** Configuration Mode.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) # system
ACMEPACKET (system) # system-config
ACMEPACKET (system-config) # alarm-threshold
ACMEPACKET (alarm-threshold) #
```

2. Use the **type** attribute to specify the TCS type (**deny-allocation** for denied entries TCAs), the **severity** attribute to specify the criticality of the alarm, and the **value** attribute to specify the alarm threshold.

The following ACLI sequence defines three alarm thresholds (minor, major, and critical), although it is not necessary to configure all thresholds. Given the static deny allocation value of 32000 on a Net-Net 4500 or 16000 on a Net-Net 3800, you can determine what the percentage value maps to.

```
ACMEPACKET (alarm-threshold) # type deny-allocation
ACMEPACKET (alarm-threshold) # severity minor
ACMEPACKET (alarm-threshold) # value 80
ACMEPACKET (alarm-threshold) # done
ACMEPACKET (alarm-threshold) # type deny-allocation
ACMEPACKET (alarm-threshold) # severity major
ACMEPACKET (alarm-threshold) # value 90
ACMEPACKET (alarm-threshold) # done
ACMEPACKET (alarm-threshold) # type deny-allocation
ACMEPACKET (alarm-threshold) # severity critical
ACMEPACKET (alarm-threshold) # value 95
ACMEPACKET (alarm-threshold) # done
```

3. Use **exit** and **verify-config** to complete this configuration.

After issuing a system alarm and accompanying SNMP trap, the TCA continues to monitor the number of denied entries. If the number of denied entries rises to the next threshold value, a new, and more severe, system alarm/SNMP trap is generated. If the number of denied entries falls below the current threshold level, and remains there for a period of at least 10 seconds, a new, and less severe system alarm/SNMP trap is generated.

### Syslog Reporting of Denied Entries

Syslog reporting of endpoint demotions was introduced as part of IDS Phase 1 in S-C6.2.0. With IDS Phase 2, such syslog messages contain the last SIP message from the endpoint that caused the transition to the denied state. If the included SIP message increases the length of the syslog beyond 1024 bytes, the SIP message is truncated so that the syslog is no larger than 1024 bytes.

### CPU Load Limiting

The transmission of IDS-related system alarms and SNMP traps is disabled when the CPU utilization rate surpasses a configured threshold percentage. When the threshold is exceeded, a syslog message (MINOR level) announces the termination of IDS reporting. No additional syslog messages or SNMP traps are generated until the CPU utilization rate falls below the configured threshold. The resumption of IDS reporting is announced by another syslog message, also issued at the MINOR level.

By default the CPU utilization rate is 90%. This value can be changed by the following ACLI command sequence

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) # options +load-limit="80"
ACMEPACKET (sip-config) # done
```

### Denied Endpoints

IDS Phase 2 provides a denied endpoint counter that includes SIP endpoints. The global counter value is available via SNMP or HDR.

The global counter value is available to SNMP under APSYSMGMT-MIB -> acmepacketMgmt -> apSystemManagementModule -> apSysMgmtMIBObjects -> apSysMgmtMIBGeneralObjects.

apSysMgmtGeneralObjects Table (1.3.6.1.4.1.9148.3.2.1.1)		
Object Name	Object OID	Description
apSysCurrentEndptsDenied	1.3.6.1.4.1.9148.3.2.1.1.26	Global counter for current endpoints denied

The system HDR collection group contains a new reporting field, Current Deny Entries Allocated, which contains the value of the global endpoints denied counter.

## Maintenance and Troubleshooting

### show sipd acls

The **show sipd acls** command includes counters that track the number of endpoints demoted from trusted to untrusted and the number of endpoints demoted from untrusted to denied. For example:

```
ACMEPACKET# show sipd acls
...
ACL Operations          ---- Lifetime ----
                   Recent      Total  PerMax
...
Trust->Untrust         0           1       1
Untrust->Deny          0           1       1
```



---

## Lawful Intercept

This section summarizes options for configuring the lawful intercept feature. It describes how the Oracle Communications Session Border Controller interoperates with mediation equipment from vendors who build lawful intercept equipment.

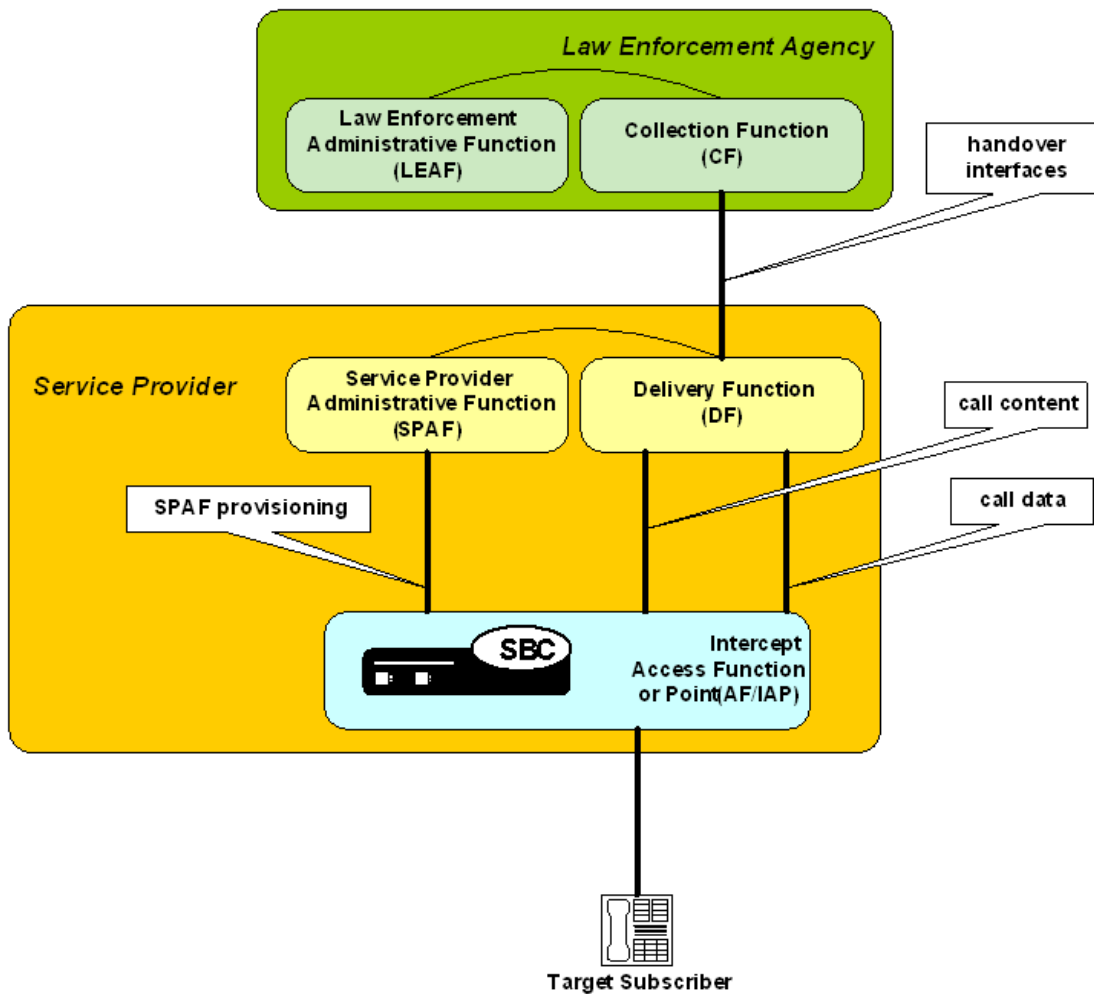
LI/CALEA consists of the interception of call content and/or the interception of call-identifying information. It requires that call information and media streams be sent to one or more law enforcement agencies in accordance with warrants for lawful interception.

You can configure your Oracle Communications Session Border Controller to support LI/CALEA functionality, enabling the Oracle Communications Session Border Controller to play a role in your Lawful Interception solution. Acting as an intercept access point (IAP), the Oracle Communications Session Border Controller can provide call data and can replicate media when signaling and media are anchored at the Oracle Communications Session Border Controller.

The Oracle Communications Session Border Controller supports LI/CALEA functionality that:

- Ensures unobtrusive intercept by hiding the network-based intercept of call information and content through topology hiding and media relay or NAT
- Intercepts and forwards call information and call content
- Interfaces with the mediation equipment [service provider administrative function (SPAF) and delivery function (DF)] for legal intercept

The following diagram provides one example of the Oracle Communications Session Border Controller deployed as part of a service provider's lawful intercept solution.



## Recommendations

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Calls may be lawfully intercepted by different devices in the service provider's network based on specific call flows, involvement of the device in the invoked service and where devices sit in the flow. Oracle recommends that you contact our Professional Services department to plan your use of the lawful intercept feature on your Oracle Communications Session Border Controller. Oracle Professional services can assist you with network/call flow analysis to determine which types of calls will involve the Oracle Communications Session Border Controller as an intercept access point and to recommend proper configuration.

## Interoperability Using X1 X2 X3

---

This document describes how the Oracle Communications Session Border Controller supports X1, X2, and X3 interfaces for lawful interception of SIP calls. In this deployment, the Oracle Communications Session Border Controller acts as an interception point that receives provisioning information from an administrative function and, based on that information, provides call data and content. As with the other LI interoperability solutions that the Oracle Communications Session Border Controller supports, the X1, X2, and X3 interfaces ensure unobtrusive call intercept by hiding network-based intercept of call data and content. The Oracle Communications Session Border Controller supports intercept of call data only, or of call data and call content.



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## External Policy Servers

### Diameter-based External Policy Servers

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The Diameter base protocol (RFC 3588) is supported by the Oracle Communications Session Border Controller and is used for Resource and Admission Control Function (RACF) and Connectivity Location Function (CLF) applications. The existing licenses for COPS based CLF and RACF support Diameter and COPS.

#### Diameter Connection

The Oracle Communications Session Border Controller supports Diameter (RFC 3588) connections to a Diameter server over TCP. The base Diameter protocol runs on TCP port 3868. Diameter-based RACF and CLF are available from the media interfaces on the Oracle Communications Session Border Controller.

The Diameter connection is always initiated from the Oracle Communications Session Border Controller to the Diameter server. The Oracle Communications Session Border Controller begins the connection by sending a Capabilities-Exchange-Request (CER) to the server, which replies with Capabilities-Exchange-Answer (CEA) message.

#### HA Support

The Oracle Communications Session Border Controller's high availability (HA) capabilities support CAC. When one Oracle Communications Session Border Controller in an HA configuration goes out of service, the MAC addresses are reassigned to a healthy Oracle Communications Session Border Controller. IP addresses follow the MAC addresses to provide a seamless switchover between HA nodes.

After an HA failover, the Diameter connection on the primary Oracle Communications Session Border Controller is either gracefully torn down, or times out depending on behavior of the PDP. The backup Oracle Communications Session Border Controller attempts to create a new Diameter connection with the PDP.

#### FQDN Support

FQDNs are configured in the address parameter in the external policy server configuration element. These FQDNs must conform to RFC 1035. If the port parameter in external policy server configuration element is not zero, then it will be used to connect to the group of applicable external policy servers. If the port parameter is set to zero, then the port number returned in SRV RR from the DNS server will be used. When the Oracle Communications Session Border Controller queries a DNS server for an FQDN configured in the external policy server, it always appends the `_diameter._tcp.` to request only TCP based Diameter servers.

### IPv6 Support

An external policy server configuration element with an application mode parameter set to Rx may be configured with an IPv6 address in the address parameter (in addition to an IPv4 address or FQDN).

### Diameter Heartbeat

Device-Watchdog-Request (DWR) and Device-Watchdog-Answer (DWA) messages are used to detect transport failures at the application layer between the Oracle Communications Session Border Controller communicating with a policy server via Diameter. The request/answer message pair forms a heartbeat mechanism that can alert the requesting side if the answering side is not reachable.

The Oracle Communications Session Border Controller always responds to a DWR by replying with a DWA message. In addition, the Oracle Communications Session Border Controller can be configured to initiate DWR messages toward a policy server or other Diameter-based network device.

You configure the **watchdog ka timer** with a timeout value that determines the number of seconds a DWA is expected in response to the Oracle Communications Session Border Controller sending a DWR.

If the Oracle Communications Session Border Controller fails to receive a DWA response from a Policy Server within the configured interval, then the connection towards that Policy Server is considered failed and torn down. The Oracle Communications Session Border Controller attempts to recreate the TCP connection, followed by the recreating the Diameter connection by issuing a CEA toward the policy server.

### Diameter Failures

During periods of application inactivity on the Diameter interface, Device-Watchdog-Request (DWR) and Answer (DWA) messages are exchanged between the client and server to provide an application-level heartbeat. DWRs may be sent toward the Oracle Communications Session Border Controller, which responds with a DWA message.

Prior to establishing a connection, the system tries to connect to a configured Diameter server using the default TCP retry interval of 10 seconds. This is also true if either side of the connection receives a TCP FIN or RST. The Oracle Communications Session Border Controller labels a connection as failed when it does not receive a DWR from the Diameter server within the guard timer period.

If an operational Diameter connection fails, the Oracle Communications Session Border Controller tries to re-open the TCP socket and Diameter connection to the Diameter server at 30 second intervals, and increases its retry interval to 5 minutes until a successful Diameter connection is made.


### Application IDs and Modes

Diameter messages include an application ID to indicate the application and standards' body interface. The following table lists the different Application-IDs for the corresponding standards' and applications. Application IDs must be provisioned manually.

	Standards Reference Point			
	RACF			CLF
Reference Point/ Standards Body	Gq 3GPP R6 29.209	Rx 3GPP R7 29.214	Rq ETSI 283 026	e2 ETSI 283 035
Application-ID	16777222	16777229	16777222	16777231

You also set the application mode to specify the interface more precisely. Doing so avoids the potential for collision between interface types that can occur when you only configure the application identifier. By setting both the application mode and application identifier for the interface, you tell the Oracle Communications Session Border Controller the format for Diameter messages it sends and receives.

The following table describes the application mode settings.

Application Mode Type	Description
Rq	As the default mode for the interface, Rq is the Oracle Communications Session Border Controller's base RACF interface. Even when you leave the application mode set to none (default), the Oracle Communications Session Border Controller runs in Rq mode. The only exception to this rule is if you set the application identifier to 16777236 and leave the application mode set to none; in this instance, the interface runs in Rx mode.
Rx	The interface runs in Rx mode when you either: Set the application mode to Rx and the application identifier to 16777236 Leave the application mode set to none, and set the application identifier to 16777236
Gq	The interface runs in Gq mode when you set the application mode to Gq.  <b>Note:</b> The application identifier 1677722 is no longer unique, but applies both to Rq and Gq interface modes.
e2	The interface runs in e2 mode, the base CLF interface, when you set the application mode to e2. Even if you leave the application mode set to none, the interface will run in e2 mode when the external policy server is configured as a CLF interface.
none	The interface runs in Rq mode when you do not configure an application identifier, or in Rx mode if you set the application identifier to 16777236.

## IPv6 Support

The Oracle Communications Session Border Controller supports Diameter-based CLF and RACF external policy servers in both IPv4 and IPv6 networks. There are three areas of enhanced functionality where the Oracle Communications Session Border Controller's Diameter external policy server offerings have changed.

- AVP support of IPv6 addresses encoded in UTF-8 format
- Extra bandwidth allocation in policed flows to compensate for longer addresses
- Framed-IPv6-Prefix AVP (97) support

### IPv6 Addresses in UTF-8 Format

When necessary, the Oracle Communications Session Border Controller checks that IPv6 addresses are formatted correctly in UTF-8 when they are inserted into relevant AVPs. The applicable AVPs are

- Flow-Description AVP (507)
- Subscription-Id-Data AVP (444)
- Destination-Realm AVP (283) only if applicable

### Framed-IPv6-Prefix AVP

The Oracle Communications Session Border Controller supports the Framed-IPv6-Prefix AVP (97) as defined in RFC4005.

The Diameter interface on the Oracle Communications Session Border Controller substitutes this AVP for the Framed-IP-Address AVP (8) in a Diameter message when carrying IPv6 addresses in the AVP.

The IP Address is encoded as Octet-String. Although the IPv6 address is a 128 bit number, it will not fall on a 4 byte boundary due to the formatting of this particular AVP. Additional whitespace will be added per RFC 3588 to the end of the octet-string to pad the ending of the AVP.

### Bandwidth Allocation Compensation for IPv6

Transporting IPv6 packets requires extra bandwidth because of their larger packet size. This needs to be taken into account when allocating bandwidth and policing media. In order to reserve the necessary bandwidth for signaling messages, the **standard pkt rate** parameter has been added to the **media profile** configuration element.

The Oracle Communications Session Border Controller needs a baseline media packet size for bandwidth requests. It first checks if the SDP includes a **ptime** value. That value will be used as the baseline if present in the request. If this value is not included in the SDP, the **standard pkt rate** parameter is used as the baseline. The chosen value is then multiplied times 20 (the difference between an IPv4 and IPv6 packet).

This value is sent to the external policy server in the bandwidth request when sending media into an IPv6 realm. From that point it is used when allocating bandwidth or media policing.

### Diameter: RACF

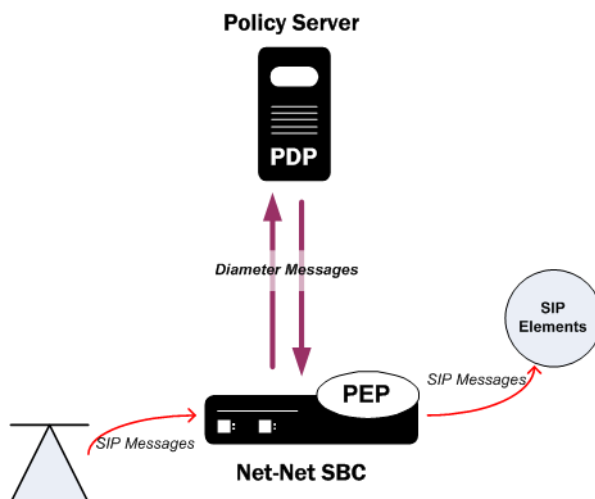
As the Oracle Communications Session Border Controller proxies and forwards calls, caller media information is known before the INVITE reaches the callee. The Oracle Communications Session Border Controller, acting as a P-CSCF, requests a specific amount of bandwidth for a call, and the RACF can reserve this amount of bandwidth for a call before the called phone rings. A call's required bandwidth can also be reserved by network devices along the path from the caller to callee if the QoS admission criteria is pushed from the RACF to other edge nodes (PEPs) such as routers, along this path to the callee.

### Implementation Features

Bandwidth-based CAC is performed according to the following typical scenario. When the Oracle Communications Session Border Controller, known as the Policy Enforcement Point (PEP), receives a SIP INVITE, it sends a Diameter Authentication Authorization Request (AAR) message to the Policy Decision Point (PDP) or Resource and Admission Control Function (RACF). The Oracle Communications Session Border Controller does not forward the INVITE to its destination at this point.

The AAR message includes call identification information and the SDP-based bandwidth requirements for the call. The RACF responds with a Diameter Authentication Authorization Answer (AAA) message to either the install or remove the call. An install command directs the Oracle Communications Session Border Controller to forward the INVITE to the next SIP device. A remove command directs the Oracle Communications Session Border Controller send a SIP 503 Service Unavailable message sent back to the UA and reject the call.

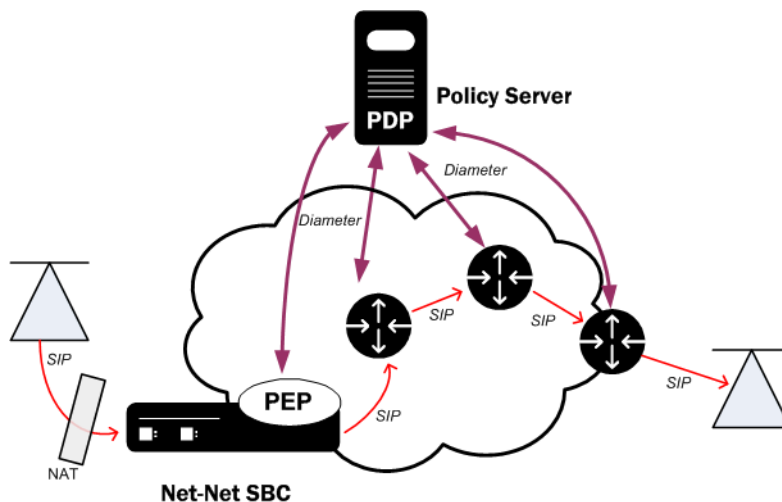
When the RACF is unreachable, incoming calls are rejected by default with a 503 message, as bandwidth can not be reserved. It is possible to configure the Oracle Communications Session Border Controller to allow all calls when the RACF is unreachable if this is the desired behavior.



The Oracle Communications Session Border Controller can be configured so that both sides of a call, based on realm, are subject to bandwidth enforcement. Each flow is treated as a unique call/event, because from a media and signaling perspective, they are distinct. As the Oracle Communications Session Border Controller functions as one side of a call, its IP address is inserted into the AAR message regardless of whether it is the calling or called party. This allows for the Diameter install or remove decision to be made before the Oracle Communications Session Border Controller receives the 200 OK response, and before ringing the far-end phone. Only one external policy server can be used within a single realm.

When a call ends, either with the expected SIP BYE or CANCEL methods, or due to other error conditions, the Oracle Communications Session Border Controller alerts the RACF by sending it a Diameter Session Termination Request (STR) message. All ended calls must be deleted from the RACF in order to accurately track used and available bandwidth.

The RACF can apply its hosted policies for calls originating at SIP UAs located behind NATs. This is a standard part of the Oracle Communications Session Border Controller's ability to provide seamless HNT.



## Bandwidth Negotiation

Because the decision whether to admit or reject a call is made before the INVITE is forwarded to the called party, some information is not available to the PDP at the initial request. The final IP Address, UDP port number, that transport the RTP flow, and the codec used are not known by the Oracle Communications Session Border Controller until the called party responds with its own SDP information (either in the 180 or 200 response).

The Oracle Communications Session Border Controller examines the Session Description Protocol (SDP) value in the body of the SIP INVITE to determine what codecs are available for the call. If the INVITE specifies more than one codec, the Oracle Communications Session Border Controller bases its request to the RACF on the most bandwidth-hungry codec to ensure that all bandwidth requests will succeed or fail on the first attempt.

**Note:** The amount of bandwidth requested depends on the configured media profiles.

If the call is admitted, and when the called party returns finalized SDP information, the Oracle Communications Session Border Controller modifies the original reservation with the chosen codec's bandwidth requirements. This ensures the RACF has current and accurate information with which to make policy decisions.

## Session Lifetime

When receiving a successful Diameter response message for bandwidth from the RACF, a session lifetime timer may be included in the message. If included, this timer states how long the session can last. If the session continues past 3/4 of session lifetime, the Oracle Communications Session Border Controller sends another bandwidth request for that session to ultimately refresh the lifetime timer. If the RACF grants this bandwidth request, the Oracle Communications Session Border Controller continues to allow the session to proceed uninterrupted. If a lifetime timer for a session is not returned to the Oracle Communications Session Border Controller by the RACF, the Oracle

## External Policy Servers

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Communications Session Border Controller assumes the session can last forever and never issues a refresh in this manner.

### Opening for RTCP Flows

When the Oracle Communications Session Border Controller functions as the AF (i.e., A-SBC or P-CSCF), you can configure it to explicitly open RTCP ports, just as it does for RTP. Without explicitly opening these ports, the Oracle Communications Session Border Controller relies on possibly unreliable PDN-GWs and BRAs to open RTCP ports and it sends only RTP information in AARs.

In external bandwidth managements configurations where the application mode is Rx and the **include-rtcp-in-request** parameters is **enabled**, the Oracle Communications Session Border Controller ensures RTCP ports are opened. It sends AAR requests to the policy server (PCRF) that contain both RTP and RTCP, opening the gates (ports) for both RTP and RTCP flows. RTCP information in the AAR is the number of the RTP port plus one (RTP port + 1 = RTCP ports) for all sessions. Flow information for RTCP is part of a different Media-Sub-component AVP as the RTP, but under the same Media-Component-Description AVP. RTCP flow information will also include the Flow-Usage AVP with RTCP (1). The RTCP port is set to 0 when the RTP ports is also unknown and therefore set to 0.

## RACF-only AVPs

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### Diameter AAR Query Post SDP Exchange

The Oracle Communications Session Border Controller supports sending the Authentication-Authorize-Request (AAR) query upon SDP answer instead of the SDP offer. This change can useful in WiMax environments where mobile phones go idle and become semi-detached from their base stations and from the WiMax access controller (WAC). In such a case, the WAC receives an AAR from the idle user but, because it cannot determine that user's base station, rejects the request.

You enable this behavior by setting the **reserve-incomplete** parameter to **orig-realm-only**.

### The Proxy Bit

When a signaling protocol receives an event request, the Oracle Communications Session Border Controller must ensure that the external policy server on the other end has enough bandwidth to maintain the requested call. The SDP information from the signaling message is stripped and encoded into the Diameter Band Request to be forwarded onto the external policy server. This feature is used with the Gq and other Diameter based interfaces.

The proxy bit allows the Oracle Communications Session Border Controller to tell the external policy server whether it wants the main server to handle the Diameter message, or if it is okay to proxy it to another server on the network. A parameter in the **ext-policy-server** configuration element called **allow-srv-proxy** has been developed. When this parameter is enabled, the proxy bit is set and the external policy server must process this Diameter request. When the parameter is disabled, the Oracle Communications Session Border Controller gives the external policy server permission to proxy the request along.

If you do not use this feature, this external policy server either handles the Diameter message on its own or proxies it to another server, depending on how much traffic it is handling at the time. This is done without any input from the Oracle Communications Session Border Controller.

### Experimental-Result-Code AVP: RACF

The Diameter RACF interface takes special actions based upon what AVPs are present inside the Experimental-Result AVP in the User-Data-Answer (UDA) message. If the Experimental-Result-Code AVP found within an Experimental-Result AVP has any value that is not considered successful, per RFC 3588, the response will be handled as non-successful response and a 503 error code will be issued back to the endpoint. If the value is a successful one, the normal call flow will proceed.

## Transport-Class AVP

When the Oracle Communications Session Border Controller, running as a Diameter-based RACF, receives a SIP INVITE triggering external bandwidth management, the Oracle Communications Session Border Controller performs SDP stripping and—through internal processes—selects an external bandwidth manager to use. If the options parameter in the selected external bandwidth manager is set to transport-class, the Oracle Communications Session Border Controller's Diameter RACF interface will issue authentication authorization requests (AARs) with the transport class AVP. The Oracle Communications Session Border Controller does not insert the transport-class AVP messages when the option is not configured.

The transport-class AVP will:

- Be identified with the AVP code 311
- Always have the vendor (V) bit set in the AVP flags
- Never have the mandatory (M) bit set in the AVP flags
- Have the Vendor-Id field set to 13019 (a value specified by ETSI TISPAN)
- Be formatted as an unsigned integer
- Reside in the Media-Component AVP, a grouped AVP

In addition, the transport-class AVP's payload field will be a 32-bit unsigned integer corresponding to a specific media type. The Oracle Communications Session Border Controller learns the specific media type from the m-line of the SDP it received. The following table shows how the Oracle Communications Session Border Controller evaluates the SDP's m= lines and concludes a default service type classification.

Service Class Classification	SDP evaluation	Default transport-class value
video	At least 1 m=video line	1
audio	No m= video At least 1m=audio	0
application	No m=video, No m=audio At least 1 m=application	3
data	No m=video, No m=audio, No m=application At least 1 m=data	2
control	No m=video, No m=audio, No m=application, No m=data At least 1 m=control	4
image	No m=video, No m=audio, No m=application, No m=data, No m=control At least 1 m=image	10
text	No m=video, No m=audio, No m=application, No m=data, No m=control, No m=image At least 1 m=text	5
message	No m=video, No m=audio, No m=application, No m=data, No m=control, No m= image, No m=text At least 1 m=message	6

## External Policy Servers

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After the service class classification has been chosen, the Oracle Communications Session Border Controller inserts the corresponding default transport class value in the transport-class AVP to send to the RACF.

### Overriding Transport- Class AVP Value

You can override the Transport class AVP value sent to the RACF in a Diameter message by configuring the **service class options** parameter. Custom service class option values are configured as a <service-class-option>=<user-entered-value> value pair. For example, to configure the Oracle Communications Session Border Controller to send the value 80 instead of the value 8 for a message service class classification, you would configure **message=80** in the service class options parameter.

### Transport-class AVP Configuration

You configure the Oracle Communications Session Border Controller to send the Transport-Class AVP in the external bandwidth manager's **options** parameter.

To set the transport-class AVP support for an external bandwidth manager:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **ext-policy-server** and press Enter.

```
ACMEPACKET (media-manager) # ext-policy-server
ACMEPACKET (ext-policy-server) #
```

4. **options**—Set the options parameter by typing options, a Space, the option name **transport-class** with a plus sign in front of it. Then press Enter.

```
ACMEPACKET (ext-policy-server) # options +transport-class
```

If you type **options** and then the option value without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. **service-class-options**—Set this options parameter by typing **service-class-options**, a <space>, the service-class type, an equal sign, your custom value. Then press Enter. Enter multiple service class/value pairs separated by a comma. For example:

```
ACMEPACKET (ext-policy-server) # service-class-options message=80 ,text=70
```

6. Save and activate your configuration.

## RACF and CLF AVPs

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### Frame-IP-Address AVP

The Diameter CLF and RACF interfaces can send a Frame-IP-Address AVP. You can configure the value to appear in either an ascii string (e.g., 192.168.10.1) or an octet string (e.g.,0xC0A80A01) with the **framed ip addr encoding** parameter.

### 1637 - Diameter Destination Realm AVP

As of S-C6.2.0, the Destination Realm AVP's value does not contain the realm of the incoming SIP message. Now, it contains the realm where the Policy Server resides as learned from the Origin-Realm AVP received in a CEA message from the Policy Server.



The Oracle Communications Session Border Controller can be configured with an option, `include-gua` , to retain the previous behavior of sending an incoming SIP message’s realm to a policy server. This is accomplished by sending the Globally Unique AVP in the AAR message to the policy server, by adding an option parameter to the external policy server configuration.

The following table summarizes the effect of provisioning the external policy server with the Globally Unique AVP option on each Diameter interface, as configured.

Diameter Interface	No <code>include-gua</code> option Configured (default)	Add <code>include-gua</code> option
Rq	AAR sends Globally Unique Address AVP	AAR sends Globally Unique Address AVP
Rx	AAR does not send Globally Unique Address AVP	AAR will contain Globally Unique Address AVP
Gq	AAR does not send Globally Unique Address AVP	AAR will contain Globally Unique Address AVP
E2	AAR sends Globally Unique Address AVP	AAR sends Globally Unique Address AVP

### Legacy Destination-Realm AVP Behavior

The Diameter CLF and RACF interfaces can change the format of the payload string in the Destination-Realm AVP for any Diameter message it originates and sends to an external server. The payload field for this AVP can be constructed in any the following formats:

Format	Description
<user>@<realm>	<user>—IP address of the endpoint initiating the call with the Oracle Communications Session Border Controller  <realm>—Name of the realm on which the Oracle Communications Session Border Controller received the INVITE from a user
<user>	<user>—IP address of the endpoint initiating the call with the Oracle Communications Session Border Controller
<realm>	<realm>—Name of the realm on which the Oracle Communications Session Border Controller received the INVITE from a user

When either the Diameter CLF or RACF interface sends any message with the Destination-Realm AVP, it determines from the external policy server configuration how to construct the payload string for this AVP.

You can set the format to use in the `dest-realm-format` parameter in the external policy server configuration. The parameter can be set to any value in the table above and defaults to <user>@<realm>. By treating the format this way, the policy server and the Oracle Communications Session Border Controller can easily communicate this value; if sent to the policy server in any AVP, the policy server can simply return the full value.

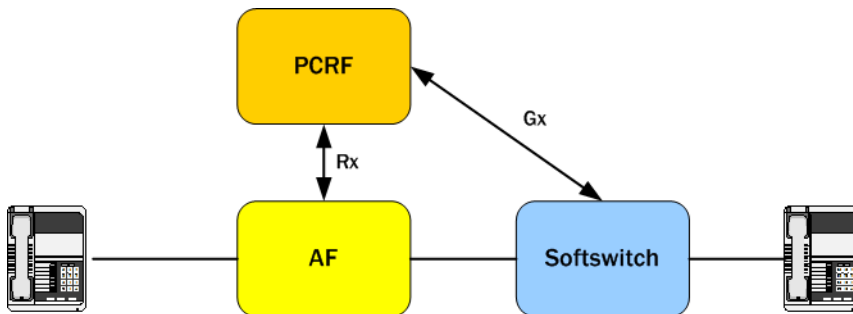
### Origin-Host AVP

The Diameter CLF and RACF interfaces can change the suffix for Origin-Realm and Origin-Host AVPs that have a payload string constructed as a domain name.

You can set the suffix you want appended to payload strings using the `domain-name-suffix` parameter in the external policy server configuration. This parameter can be set to any string (default is .com), and the Oracle Communications Session Border Controller automatically adds a dot (.) to the front of this entry if you do not include one. The policy server and the Oracle Communications Session Border Controller can easily communicate this value; if sent to the policy server in any AVP, the policy server can simply return the full value.

### Wildcard Transport Protocol

The Oracle Communications Session Border Controller external bandwidth management solution provides for an Rx interface that supports the Flow-Description AVP (507). Rather than use a numerical value, this Flow-Description AVP uses an IP filter rule with the keyword “ip.” The ip keyword means any transport protocol matches the Flow-Description AVP when issuing AARs to the PCRF. Before it forwards a Gx RAR messages to the softswitch, the PCRF decodes the audio codec into the correct speed and classification. The PCRF passes the Oracle Communications Session Border Controller's Flow-Description AVP to the softswitch untouched. But not all softswitches accommodate the ip keyword, resulting in rejected requests.



When you enable the **wildcard-transport-protocol** parameter, however, you can essentially format the Flow-Description AVP to suit your network.

For sessions that need to allocate media and have applicable external bandwidth management associations, the Oracle Communications Session Border Controller's Diameter interface checks for the necessary bandwidth. The Diameter interface, with an Rx or Rq application mode, constructs an AAR with Flow-Description AVP of ip when the **wildcard-transport-protocol** parameter is **enabled**. The flow description would look like this:

```
<Flow-Description-AVP(507) | Avp Flags=128 | AVP Length=72 | Vendor-Id=10415  
Data = permit out ip from 168.192.24.20 49500 to 168.192.24.0 8000  
<Flow-Description-AVP(507) | Avp Flags=128 | AVP Length=72 | Vendor-Id=10415  
Data = permit in ip from 168.192.24.0 8000 to 168.192.24.20 49500
```

With the **wildcard-transport-protocol** set to **disabled**, the Oracle Communications Session Border Controller does not use the ip wildcard keyword. Instead, it uses the specific media stream transport protocol in the Flow-Description AVP—and only falls by to the ip keyword when the transport protocol is unknown. The flow description with this parameter disabled would look like this:

```
<Flow-Description-AVP(507) | Avp Flags=128 | AVP Length=72 | Vendor-Id=10415  
Data = permit out 17 from 168.192.24.20 49500 to 168.192.24.0 8000  
<Flow-Description-AVP(507) | Avp Flags=128 | AVP Length=72 | Vendor-Id=10415  
Data = permit in 17 from 168.192.24.0 8000 to 168.192.24.20 49500
```

#### New Configurations and Upgrading

To comply with 2GPP TS 29.213, **wildcard-transport-protocol** parameter is **disabled** by default in new configurations. So if the transport protocol is known, the Oracle Communications Session Border Controller uses it in the Flow-Description AVP.

To maintain default behavior for existing configurations, the Oracle Communications Session Border Controller performs a check at the time of upgrade to set this parameter to **enabled**. This setting means the Oracle Communications Session Border Controller does use the ip keyword in the Flow-Description AVP.

### Configuring Diameter-based RACF

In the following configuration examples, we assume that your baseline configuration passes SIP traffic, with the Oracle Communications Session Border Controller in the role of an Access N Oracle Communications Session Border Controller. In this example, you will configure additions to the ream configuration and the new external bandwidth manager configuration. You must also configure media profiles to accept bandwidth policing parameters.

## Diameter Support Realm Configuration

To configure the realm configuration for Diameter support in a CAC scenario:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

4. Type **select** and the number of the pre-configured sip interface you want to configure.

```
ACMEPACKET (realm-config) # select 1
ACMEPACKET (realm-config) #
```

5. **mm-in-realm**—Set this parameter to **enabled** so that calls from devices in the same realm have their media flow through the Oracle Communications Session Border Controller to be subject to CAC. The default value is **disabled**. The valid values are:
  - enabled | disabled
6. **mm-in-network**—Set this parameter to **enabled** so that the Oracle Communications Session Border Controller will steer all media traveling between two endpoints located in different realms, but within the same network. If this field is set to **disabled**, then each endpoint will send its media directly to the other endpoint located in a different realm, but within the same network. The default value is **enabled**. The valid values are:
  - enabled | disabled
7. **ext-bw-manager**—Enter the name of the external bandwidth manager configuration instance to be used for external CAC for this Realm.
8. Save your work using the ACLI **done** command.

## External Bandwidth Manager Configuration

To configure the external bandwidth manager:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # media-manager
```

3. Type **ext-policy-server** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # ext-policy-server
ACMEPACKET (ext-policy-server) #
```

4. **name**—Enter the name for this external bandwidth manager instance. This parameter is used to identify the PDP that will be used in each Realm configuration.
5. **state**—Set the state of this ext-policy-server configuration to **enabled** to run this CAC. The default value is **enabled**. The valid values are:
  - enabled | disabled
6. **operation-type**—Enter **bandwidth-mgmt** for this external policy server configuration element to perform RACF/External Policy Server functions. The default value is **disabled**. The valid values are:
7. **protocol**—Enter **Diameter** to support Diameter-based CAC. The default value is **C-SOAP**.
8. **address**—Enter the IP Address or FQDN of the external RACF.

## External Policy Servers

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9. **port**—Enter the port number the diameter connection connects to on the RACF. The default value is **80**. The valid range is:
  - Minimum—0
  - Maximum—65535
10. **realm**—Enter the name of the Realm in which this Oracle Communications Session Border Controller defines the RACF to exist. This is NOT necessarily the Realm where the Oracle Communications Session Border Controller performs admission control.
11. **permit-conn-down**—Enter **enabled** if this external policy server configuration can permit new calls into the network when the policy server connection is down. The default value is **disabled**. The valid values are:
12. **product-name**—Enter text string that describes the vendor-assigned name for the RACF. This parameter is required.
13. **application-mode**—Enter the type of interface you want to use. Your choices are: **Rq, Rx, Gq, e2**, and **none**.
14. **application-id**—Enter a numeric application ID that describes the interface used to communicate with the RACF. The default value is zero (**0**). The valid range is:
  - Minimum—0
  - Maximum—999999999
15. **framed-ip-addr-encoding**—Enter the format of the Frame-IP-Address (AVP 8) value in Diameter messages. The default value is **octet-string**. The valid values are:
  - **ascii-string**—Example: 192.168.10.1
  - **octet-string**—Example: 0xC0A80A01
16. **dest-realm-format**—Enter the format you want to use for the Destination-Realm AVP. The default value is **user\_with\_realm**. The valid values are:
  - **user\_with\_realm | user\_only | realm\_only**
17. **domain-name-suffix**—Enter the suffix you want to use for Origin-Realm and Origin-Host AVPs that have a payload string constructed as a domain name Your value can be any string, to which the system will prepend with a dot if you do not include one. The default value is **.com**.
18. **allow-srv-proxy**—Set to **enabled** in order to include the proxy bit in the header. The presence of the proxy bit allows the Oracle Communications Session Border Controller to tell the external policy server whether it wants the main server to handle the Diameter message, or if it is okay to proxy it to another server on the network (disabled). The default is **enabled**. The valid values are:
  - **enabled | disabled**
19. **wildcard-trans-protocol**—Set this parameter from **enabled** if you want to use transport protocol wildcarding for Rx/Rq Flow-Description AVP (507) generation. Enabled sends a flow description of ip. Set this parameter to **disabled** if you want to use the specific media stream transport protocol.
20. **reserve-incomplete**—Set this parameter to **enabled** when communicating with a PDP via Diameter. The parameter allows the Oracle Communications Session Border Controller to make admission requests before learning all the details of the flows and devices (e.g., not knowing the final UDP port numbers for the RTP media streams until after the RTP has begun). The default value is **enabled**. The valid values are:
  - **enabled** (default)—This mode supports the usual behavior when the AAR is sent upon SDP offer as well as SDP answer. This mode ensures backwards compatibility.
  - **orig-realm-only**—This mode allows calls originating from a realm with a policy server associated with it to send the AAR upon SDP offer. However, calls terminating at a realm with a policy server associated with it send the AAR post SDP exchange.
  - **disabled**—This mode allows no bandwidth reservation for incomplete flows.
21. **include-rtcp-in-request**—Change this parameter from **disabled** (default), to **enabled** so the Oracle Communications Session Border Controller will include RTCP information in AARs. RTCP information is the number of the RTP port plus one (RTP port + 1 = RTCP ports) for all sessions.

22. **trans-expires**—Set the amount of time, in seconds, that the Oracle Communications Session Border Controller waits before performing an expiration if a Diameter base protocol transaction does not occur. The default value is 15 seconds. Valid values range between 1 and 15.
23. Save your work using the ACLI **done** command.

## Media Profile Configuration

Values for the following parameters can be found in the PacketCable™ Audio/Video Codecs Specification PKT-SP-CODEC-I06-050812 document.

To configure the media profile configuration for Diameter support in a CAC scenario:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router path.

```
ACMEPACKET (configure) # session-router
```

3. Type **media-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # media-profile
ACMEPACKET (media-profile) #
```

4. Type **select** and the number of the pre-configured media profile you want to configure.

```
ACMEPACKET (media-profile) # select 1
ACMEPACKET (media-profile) #
```

5. **req-bandwidth**—Enter the required bandwidth in Kbps for the selected media profile. This is the bandwidth that the Oracle Communications Session Border Controller will request from the policy server. The default value is zero (0). The valid values are:

- Minimum—0
- Maximum—4294967295

6. **standard-pkt-rate**—Enter the value to use for the standard packet rate for this codec when sending a request to the RACF for a bandwidth request.

7. Save your work using the ACLI **done** command.

## CAC Debugging

A new argument has been added to the show command for viewing CAC statistics. From the user prompt, type **show ext-band-mgr**.

```
ACMEPACKET# show ext-band-mgr
10:11:38-194
EBM Status
```

	Active	-- Period --		----- Lifetime -----		
		High	Total	Total	PerMax	High
Client Trans	0	0	0	0	0	0
Server Trans	0	0	0	0	0	0
Sockets	1	1	1	1	1	1
Connections	0	0	0	0	0	0

```

----- Lifetime -----
Recent      Total    PerMax
Reserve      0         0         0
Modify       0         0         0
Commit       0         0         0
Remove       0         0         0
EBM Requests 0         0         0
EBM Installs 0         0         0
EBM Errors   0         0         0
EBM Rejects  0         0         0
```

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EBM Expires	0	0	0
EBMD Errors	0	0	0

You can also refer to the log.ebmd log file located in the /ramdrv/logs/ directory on the Oracle Communications Session Border Controller. This file must be retrieved via FTP or SFTP.

## 2127 - Configurable Subscription ID Types

### Subscriber Information AVP

Certain policy servers rely on having the user's URI information available as means to identify the endpoint/subscriber. In addition to conveying the L3 IP address of a user, the Oracle Communications Session Border Controller supports RFC 4006: The Subscription-Id AVP. It identifies the end user's subscription and is used in 3GPP Rx reference point. This feature can be enabled regardless of the selected application mode of the external policy server.

You can enable the Oracle Communications Session Border Controller to include the Subscription-ID-Type based on the received message's request line's Request URI, or you can set the Subscription ID type on a per-realm basis.

### Subscription-ID AVP

The Subscription-Id AVP (AVP Code 443) includes a Subscription-Id-Data AVP that holds the identifier and a Subscription-Id-Type AVP that defines the identifier type. The external policy server configuration element is configured with an option to enable sending the Subscription-Id AVP to the policy server in an AA-Request (AAR) message.

### Subscription-Id-Type

The Oracle Communications Session Border Controller can send a Subscription-Id-Types AVP to an external policy server:

Value	Name	Description
0	END_USER_E164	Identifier is in international E.164 format (e.g., MSISDN), according to the ITU-T E.164 numbering plan.
1	END_USER_IMSI	Identifier is in international IMSI format, according to ITU-T E.212 numbering plan as defined in [E212] and [CE212].
2	END_USER_SIP_URI	Identifier is in the form of a SIP/SIPS URI.

To send the Subscription-ID-Types AVP in an AAR, you must add the include-sub-info option in the external policy server configuration element. If this option is enabled, and you do not configure a **subscription-id-type** in the realm-config, the request line's Request URI of the received message is sent in the AAR.

You can also set the Subscription-ID-Type on a per-realm basis by configuring the parameter **subscription-id-type** in the realm-config element to determine the value of the Subscription ID in the AAR.

The default value for **subscription-id-type** is END\_USER\_NONE. If the value of subscription-id-type is set to END\_USER\_NONE, and the external policy server option is enabled to send the Subscription ID AVP, then the Oracle Communications Session Border Controller relies on the request line's Request URI of the received message.

If the value for **subscription-id-type** is set to one format, such as END\_USER\_IMSI, but the request line's Request URI is set to another format, such as END\_USER\_SIP\_URI, the value of **subscription-id-type** is used to send in the AAR message.

### Subscription ID AVP in AA-Request Message Configuration

To configure the Oracle Communications Session Border Controller to send Subscription ID AVP in AA-Request Messages:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **ext-policy-server** and press Enter.

```
ACMEPACKET (session-router) # ext-policy-server
ACMEPACKET (ext-policy-server) #
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI select command) the external policy server that you want to edit.

4. **options**—Set the options parameter by typing **options**, a Space, the option-name **include-sub-info** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET (ext-policy-server) # options +include-sub-info
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to the policy server configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

### Subscription ID AVPs in AA-Request Message Configuration

To set the Subscription ID AVPs in AA-Request Messages:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter.

```
ACMEPACKET (session-router) # realm-config
ACMEPACKET (realm-config) #
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI select command) the realm that you want to edit.

4. **subscription-id-type**—Set the Subscription ID Type included in the AA-Request.

- END\_USER\_E164
- END\_USER\_IMSI
- END\_USER\_SIP\_URI
- END\_USER\_NONE

5. Save and activate your configuration.

## Specific Action AVP support

When acting as a P-CSCF, Oracle Communications Session Border Controller sends the Specific-Action AVP to the PCRF in an AAR message to indicate the subscription types it supports.

The Oracle Communications Session Border Controller can be configured to subscribe to one or more of the following subscription types:

- LOSS OF BEARER

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- RECOVERY OF BEARER
- RELEASE OF BEARER
- OUT OF CREDIT
- SUCCESSFUL RESOURCES ALLOCATION
- FAILED RESOURCES ALLOCATION

When no subscription types are configured, the Oracle Communications Session Border Controller does not include the Specific-Action AVP in its AAR.

Specific Action AVP subscription is configured in the **Specific action subscription** parameter located in the External policy server configuration element.

## ACLI Examples - specific-action-subscription

1. Navigate to the ext-policy-server configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# ext-policy-server
ACMEPACKET(ext-policy-server)#
```

2. Select an existing external policy server configuration element.

```
ACMEPACKET(ext-policy-server)# select
name:
1: extpoll

selection: 1
ACMEPACKET(ext-policy-server)#
```

3. specific-action-subscription - Configure 1 or more specific actions. When configuring 2 or more specific actions, enclose them in quotation marks, with the values separated by spaces. The following are valid specific actions: loss-of-bearer, recovery-of-bearer, release-of-bearer, out-of-credit, successful-resources-allocation, failed-resources-allocation

```
ACMEPACKET(ext-policy-server)# specific-action-subscription "loss-of-bearer
recovery-of-bearer"
```

4. Type done to save your work.
5. Save and activate your configuration to begin using this feature.

## Diameter STR Timeouts

---

When a call ends, the Oracle Communications Session Border Controller alerts the RACF by sending it a Diameter Session Termination Request (STR) message. You can enable the Oracle Communications Session Border Controller to resend STR messages at the application layer if the STR messages time out. A new option **STR-retry=x** has been created that allows you to configure the number of times the STR messages are resent.

## Diameter STR Timeouts Configuration

To configure Diameter request timeouts:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **ext-policy-server** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.



```
ACMEPACKET (media-manager) # ext-policy-server
ACMEPACKET (ext-policy-server) #
```

4. Set the options parameter by typing **options**, a Space, the option name **STR-retry=x** with a plus sign in front of it. Then press Enter.

```
ACMEPACKET (ext-policy-server) # options +STR-retry=x
```

If you type options and then the option value without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration’s options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save and activate your configuration.

## Gq Interface Features

The Oracle Communications Session Border Controller can run the Gq interface over a Diameter connection and act as a P-CSCF (AF) communicating with a PDF. The application ID field must be set to 16777222 to run the Gq reference point.

## Rx Interface Features

The Oracle Communications Session Border Controller can run the Rx interface over a Diameter connection and act as a P-CSCF communicating with a PCRF. The application ID field must be set to 16777236 to run the Rx reference point. When running in this mode, the Oracle Communications Session Border Controller will send two AVPs, in addition to other information:

- The Codec-Data AVP is then included for non-priority calls. This AVP is one of several that together comprise a Group AVP structure.
- The Reservation-Priority AVP is included for priority calls. This AVP will be the main AVP within the AAR message.

## Non-Priority Call Handling

When a SIP signaling event triggers external bandwidth management use, the Oracle Communications Session Border Controller removes all SDP information from the signaling message that was the trigger. The Oracle Communications Session Border Controller repackages this bandwidth information so that it can form a Bandwidth Request and decide on an external bandwidth manager to which it should be sent. If the appropriate external bandwidth manager is configured for Rx interface use, then Oracle Communications Session Border Controller then reformats the SDP information to construct a Codec-Data AVP.

If the external bandwidth manager that receives the request ignores the SDP information, then it does not include the Codec-Data AVP in the AAR.

For calls that do not require special treatment, the Codec-Data AVP is required to have the:

- AVP code 599
- 3GPP vendor identification number (10415)
- V (vendor) bit set in the AVP
- M (mandatory) bit set when sending this AVP
- Type octet string

In addition, the Codec-Data AVP appears as described in the following table.

AVP section/line	Requirement
Line 1	Must specify the direction of the flow by including the ASCII “uplink” or downlink: uplink—Identifies the SDP as having come from the UE and sent to the network

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AVP section/line	Requirement
	downlink—Identifies the SDP as having come from the network and sent to the UE
Line 2	Must specify whether the offer or answer codec is at issue by including the ASCII “offer” (from an SDP offer according to RFC 3264) or answer (from an answer according to RFC 3264)
Remainder of the AVP	Must include lines found in the signaling SDP, formatted in ASCII and separated by new-line characters; the first line of this section must be the m line, followed by any “a” or “b” lines related to that m line

## Priority Call Handling

The Oracle Communications Session Border Controller determines that a call is priority call when it matches a defined network management control (NMC) priority rule. No other scenario triggers the priority call handling treatment described in this section.

When a SIP signaling event triggers external bandwidth management use for a priority call, the Oracle Communications Session Border Controller sends the Reservation-Priority AVP in the AAR message. The Reservation-Priority AVP is required to:

- Use the ETSI Vendor identification number (13019)
- Have the V (vendor) bit set in the AVP
- Not to have the M (mandatory) bit set when sending this AVP
- Be of type enumeration
- Set to PRIORITY-SEVEN (7)

## Rx bearer plane event

The Rx reference point is used to exchange application level session information between the PCRF and the AF which is the Oracle P-CSCF/SBC. The PCRF exchanges the Policy and Charging Control (PCC) rules with the PCEF (Policy and Charging Enforcement Function) and QoS rules with the BBERF (Bearer Binding and Event Reporting Function). The role of the former is typically performed by a 3GPP/GGSN, 3GPP-R9/PGW, 3GPP2/PDSN, WiFi/Max/ASN-GW. The role of the latter may be a standalone element or typically it’s incorporated within the system acting as the PCEF. The Oracle P-CSCF/SBC offers applications that require policy and charging control of traffic plane resources (e.g. UMTS PS domain/GPRS domain resources). Currently support for Rx within the Oracle Communications Session Border Controller supports policy, the present requirement will enhance Rx to incorporate traffic plane event notifications north bound. The PCRF receives session and media related information from the Oracle Communications Session Border Controller currently and with this enhancement the PCRF will report traffic plane events to the Oracle Communications Session Border Controller.

BN-Bearer-0040: When not all the service data flows within the AF session are affected, the PCRF will send an RAR command that includes the deactivated IP flows encoded in the Flows AVP and the cause encoded in the Specific-Action AVP.

BN-Bearer-0050: The Oracle SBC/P-CSCF shall acknowledge the receipt of this RAR command by sending an RAA. It shall then proceed to release the entire session at the SIP signaling layer and proceed to send the corresponding Diameter STR command to the PCRF and receive the STA.

### AA-Request (AAR) Command

The following Specific-Action notifications will be added to the AA-Request command:

- INDICATION\_OF\_LOSS\_OF\_BEARER (2)
- INDICATION\_OF\_RECOVERY\_OF\_BEARER (3)
- INDICATION\_OF\_RELEASE\_OF\_BEARER (4)
- INDICATION\_OF\_OUT\_OF\_CREDIT (7)
- INDICATION\_OF\_SUCCESSFUL\_RESOURCES\_ALLOCATION (8)

- INDICATION\_OF\_FAILED\_RESOURCES

### Re-Auth-Request (RAR) Command Handling

The Flows AVP (510) and Abort-Cause AVP (500) will be ignored if they are received in the RAR command.

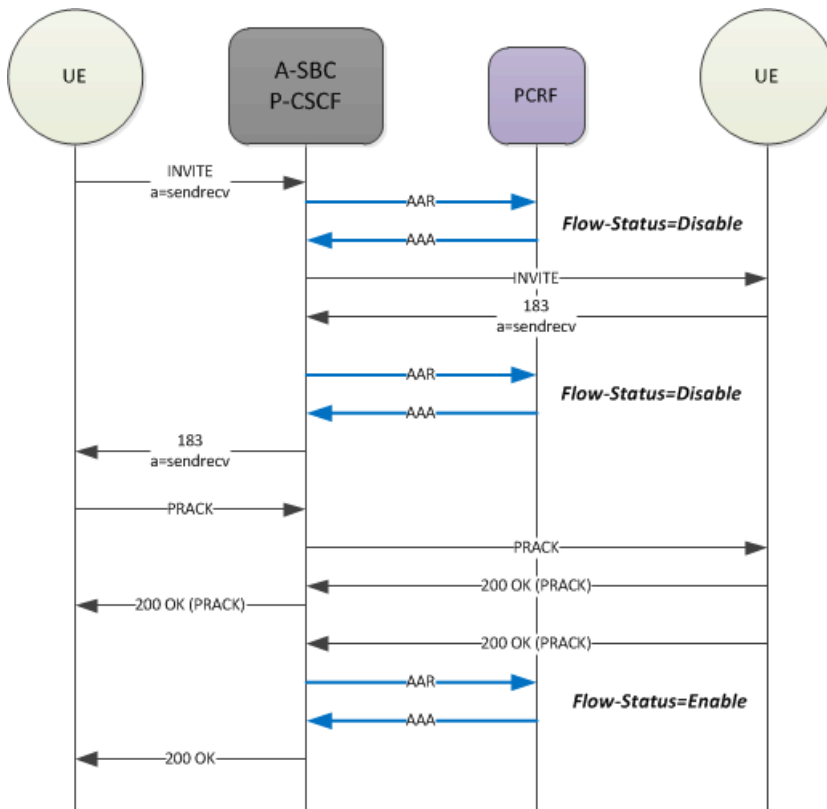
The Specific-Action AVP (513) will be handled as follows:

- Notification: INDICATION\_OF\_LOSS\_OF\_BEARER (Action: Terminate SIP Session \* Send RAA BYE will be sent to both the EU and the core. This is similar to the Abort-Session--Request (ASR) handling).
- Notification: INDICATION\_OF\_RELEASE\_OF\_BEARER (Action: Terminate SIP Session \* Send RAA BYE will be sent to both the EU and the core. This is similar to the Abort-Session--Request (ASR) handling).
- Notification: INDICATION\_OF\_OUT\_OF\_CREDIT (Action: Terminate SIP Session \* Send RAA BYE will be sent to both the EU and the core. This is similar to the Abort-Session--Request (ASR) handling).
- Notification: INDICATION\_OF\_FAILED\_RESOURCES\_ALLOCATION (Action: Terminate SIP Session \* Send RAA BYE will be sent to both the EU and the core. This is similar to the Abort-Session--Request (ASR) handling).
- Notification: INDICATION\_OF\_SUCCESSFUL\_RESOURCES\_ALLOCATION (Action: Send RAA).
- Notification: INDICATION\_OF\_RECOVERY\_OF\_BEARER (Action: Send RAA)

## 2836 - Early Media Suppression for Rx

The Oracle Communications Session Border Controller supports early media suppression where it does not allow media to flow between the endpoints it connects until the session is fully established as noted by receiving a 200 OK. Early media suppression is configured by setting the **early-media-allow** parameter on a realm, realm group, or session agent to **none**.

The Oracle Communications Session Border Controller also relays the state of early media suppression in the Flow-Status AVP in an AAR message to the PCRF as the call is being set up. When the call's media is still in a suppressed state, the Flow-Status AVP is set to Disabled. The following image illustrates when the Flow-Status changes from disable to enable.



### Media-Component-Number and Flow-Number AVP Values

The values assigned to the Media-Component-Number and Flow-Number AVPs are not derived in compliance with 3GPP standards. Version S-CZ7.2.0 provides the capability to derive compliant values.

AA-Request (AAR) messages are sent, via the Rx Interface, by an Access Function (AF) to a Policy and Charging Rules Function (PCRF) in order to provide it with the session information. Among other fields, these messages contain a Media-Component-Number AVP and a Flow-Number AVP, used to identify specific media sessions, and the individual IP flows associated with these sessions.

The values assigned to the Media-Component-Number and Flow-Number AVPs are not derived in compliance with 3GPP standards.

Release S-CZ7.2.0 provides a new option that enables compliance with 3GPP requirements. Note that this support is limited to the Rx Interface

#### Media-Component-Number AVP

The Media-Component-Number AVP (AVP code 518) is of type Unsigned32; it contains the ordinal number of the media component (the SDP m= line) as specified in 3GPP TS 29.214. When this AVP refers to AF signalling, it contains the value 0. The Media-Component-Number AVP provides an index to the grouped Media-Component-Description AVP (AVP code 517).

3GPP compliance requires that:

1. Number should start from 1
2. It should contain the ordinal number of the position of the m= line in the SDP.
3. When this AVP refers to AF Signalling, this is indicated by using the value 0.
4. The ordinal number of a media component shall not be changed when the session description information is modified.

#### Flow-Number AVP

The Flow-Number AVP (AVP code 509) is of type Unsigned32; it contains the ordinal of the IP flows created to support a specific SDP m= line as specified in 3GPP TS 29.214. The Flow-Number AVP provides an index to the grouped Media-Sub-Component AVP (AVP code 519).

3GPP compliance requires that:

1. Number should start from 1
2. It should contain the ordinal number of the IP flow(s) within the m= line assigned in the order of increasing downlink destination port numbers, if downlink destination port numbers are available.
3. It should contain the ordinal number of the IP flow(s) within the m= line assigned in the order of increasing uplink destination port numbers, if no downlink destination port numbers are available.
4. The ordinal number assigned should not be changed when the session description information is modified.

#### Flow Examples

The following example illustrates the derivation of flow identifiers from SDP as specified by 3GPP TS 29.214 3rd Generation Partnership Project; Technical Specification Group Core Network and Terminals; Policy and Charging Control over Rx reference point (Version 9.3.0).

A flow identifier consists of two digits in the form #,# with the digit identifying a specific SDP m= line (defining a media session), and the second digit identifying a specific IP flow (for example, an RTP stream) supporting the media session. 3GPP standards require that the Media-Component-Number AVP be populated with the first digit if the flow identifier and the Flow-Number AVP be populated with the second digit.

A UE, as the offerer, sends a SDP session description (relevant sections shown below) to an application server.

```
v=0
o=ecsreid 3262464865 3262464868 IN IP6 2001:0646:00F1:0045:02D0:59FF:FE14:F33A
s=MM01
```

```
i=One unidirectional audio media and one unidirectional video media and one
bidirectional application
media
t=3262377600 3262809600
m=video 50230 RTP/AVP 31 Ordinal 1 (3 media flows, 1
RTP & 2 RTCP)
c=IN IP6 2001:0646:00F1:0045:02D0:59FF:FE14:F33A
a=recvonly
m=audio 50330 RTP/AVP 0 Ordinal 2 (3 media flows, 1
RTP & 2 RTCP)
c=IN IP6 2001:0646:00F1:0045:02D0:59FF:FE14:F33A
a=sendonly
m=application 50430 udp wb Ordinal 3 (2 media flows,
up & down links)
c=IN IP6 2001:0646:00F1:0045:02D0:59FF:FE14:F33A
a=sendrecv
```

The application returns (relevant sections):

```
v=0
o=ecsreid 3262464865 3262464868 IN IP6 2001:0646:00F1:0045:02D0:59FF:FE14:F33A
s=MM01
i=One unidirectional audio media and one unidirectional video media and one
bidirectional application
media
t=3262377600 3262809600
m=video 51372 RTP/AVP 31 Ordinal 1
c=IN IP6 2001:0646:000A:03A7:02D0:59FF:FE40:2014
a=sendonly
m=audio 49170 RTP/AVP 0 Ordinal 2
c=IN IP6 2001:0646:000A:03A7:02D0:59FF:FE40:2014
a=recvonly
m=application 32416 udp wb Ordinal 3
c=IN IP6 2001:0646:000A:03A7:0250:DAFF:FE0E:C6F2
a=sendrecv
```

From this offer–answer exchange of SDP parameters the UE and the application server each creates a list of flow identifiers as shown below.

Order of m= line	Type of IP Flows	Destination IP Address/Port Number of Flows	Flow Identifier
1	RTP (Video) DL	2001:0646:00F1:0045:02D0:59FF:FE14:F33A / 50230	<1,1>
1	RTCP DL	2001:0646:00F1:0045:02D0:59FF:FE14:F33A / 50231	<1,2>
1	RTCP UL	2001:0646:000A:03A7:02D0:59FF:FE40:2014 / 51373	<1,2>
2	RTP (Audio) UL	2001:0646:000A:03A7:02D0:59FF:FE40:2014 / 49170	<2,1>
2	RTCP DL	2001:0646:00F1:0045:02D0:59FF:FE14:F33A / 50331	<2,2>
2	RTCP UL	2001:0646:000A:03A7:02D0:59FF:FE40:2014 / 49171	<2,2>
3	UDP (application) DL	2001:0646:00F1:0045:02D0:59FF:FE14:F33A / 50430	<3,1>
3	UDP (application) UL	2001:0646:000A:03A7:0250:DAFF:FE0E:C6F2 / 32416	<3,1>

### Media-Component AVP 3GPP Compliance Configuration

Use this procedure to enable compliance with 3GPP requirements. Compliance is enabled with the **format-flow-id** option available in **ext-policy-server** configuration mode.

1. Access the **ext-policy-server** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
```

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```
ACMEPACKET(media-manager)# ext-policy-server
ACMEPACKET(ext-policy-server)#
```

2. Select the **ext-policy-server** object to edit.

```
ACMEPACKET(ext-policy-server)# select
<name>:1: name=extpoll

selection: 1
ACMEPACKET(ext-policy-server)#
```

3. **format-flow-id**—Add this option to enable 3GPP compliance.

```
ACMEPACKET(ext-policy-server)# options +format-flow-id
```

4. Type **done** to save your configuration.

## Rx Interface Reason Header Usage

The Oracle Communications Session Border Controller has increased capability to map network events to a configurable SIP Reason header. The extended capability provides for the mapping of specific disconnect events on the Rx Interface.

Release S-CZ7.2.0, and later releases expands the usage of the Reason header and its associated parameters to include mapping of disconnect events on the Rx interface between an SBC (functioning in the P-CSCF role) and a Policy and Charging Rules Function (PCRF). With the expanded capability enabled, contents of the Reason header are based on either the contents of the Specific-Action AVP in Re-Authorization Request (RAR) messages issued by the PCRF, or the contents of the Abort-Cause AVP in Abort Session Request (ASR) messages issued by the PCRF.

To use this feature, set the **session-router > sip-interface > rx-sip-reason-mapping** parameter to enabled.

### Specific-Action AVP

The Specific-Action AVP (AVP code 513) is of type Enumerated.

Within a PCRF initiated Re-Authorization Request, the Specific-Action AVP determines the type of the action.

The following reported disconnected events can be mapped to SIP Reason headers using the corresponding value to configure in a **local-response-map-entry > local-error**.

Specific-Action AVP value	local-error configuration
2 -- INDICATION_OF_LOSS_OF_BEARER	rx-rar-loss-of-bearer
4 -- INDICATION_OF_RELEASE_OF_BEARER	rx-rar-release-of-bearer
7 -- INDICATION_OF_OUT_OF_CREDIT	rx-rar-out-of-credit
9 -- INDICATION_OF_FAILED_RESOURCES_ALLOCATION	rx-rar-failed-resources-allocation

### Abort-Action AVP

The Specific-Action AVP (AVP code 500) is of type Enumerated.

Within a PCRF initiated Abort Session Request (ASR), the Abort-Cause AVP identifies the cause of the ASR.

The following reported disconnected events can be mapped to SIP Reason headers using the corresponding value to configure in a **local-response-map-entry > local-error**.

Abort-Cause AVP value	local-error configuration
0 -- BEARER_RELEASED	rx-asr-bearer-released
1 -- INSUFFICIENT_SERVER_SERVICES	rx-asr-insufficient-server-resources
2 -- INSUFFICIENT_BEARER_SERVICES	rx-asr-insufficient-bearer-resources
3 -- PS_TO_CS_HANDOVER	rx-asr-ps-to-cs-handover

Abort-Cause AVP value	local-error configuration
4 -- SPONSORED_DATA_CONNECTIVITY_DISALLOWED	rx-asr-sponsored-data-connectivity-disallowed

### Message Flows

Mapped headers will appear in either a BYE or a CANCEL message from the P-CSCF. As shown in the following illustrations, a BYE message is issued when the disconnect event occurs after establishing the SIP session. A CANCEL message is issued when the disconnect event occurs before establishing the SIP session.

### RAR Loss-of-Bearer After Session Establishment

If mapping is not enabled, the Reason header defaults to SIP;cause=503;text="Service Unavailable".

## Network Provided Location Information

Network provided location information (NPLI) is a service commonly supported within the IMS network architecture. NPLI is most commonly provided as a geographic identifier which specifies a location either in terms of geodetic coordinates (latitude and longitude), for example 42°25'33 N 71°18'16 W; or in terms of a recognized civic identifier, for example Lincoln, Massachusetts.

Network provided location information (NPLI) is a service commonly supported within the IMS network architecture. NPLI is most commonly provided as a geographic identifier which specifies a location either in terms of geodetic coordinates (latitude and longitude), for example 42°25'33 N 71°18'16 W; or in terms of a recognized civic identifier, for example Lincoln, Massachusetts.

Support for NPLI was introduced in two phases. Initial, Phase 1 support for NPLI, was provided by Version S-CZ7.1.2. Enhanced, Phase 2 NPLI support, is provided by Version S-CZ7.2.0 and later releases.

Phase 1 and Phase 2 support is described in the following sections.

### Basic Implementation

Basic Phase 1 NPLI support is provided with Version S-CZ7.2.0.

While implementation details and the granularity of provided services differ among IMS providers, Phase 1 NPLI implementation triggered by an originating mobile device typically involves a message exchange similar to the following.

1. Upon receiving the initial INVITE from the mobile handset, the P-CSCF sends an AA-Request (AAR) command to the Policy Charging/Rule Function (PCRF) via the Rx interface.
2. The P-CSCF buffers the INVITE pending the receipt of an AA-Answer (AAA) command from the PCRF. Assuming NPLI is provided by the IMS core, the AAA contains a 3GPP-User-Location-Info Attribute Value Pair (AVP) along with a RAT-Type AVP. Depending upon network equipment, the AAA may also contain a proprietary MSISDN (Mobile Subscriber Integrated Services Digital Network Number), AVP although this is not usually the case.
3. Upon receiving the AAA, the P-CSCF, using values contained in the 3GPP-User-Location-Info attribute, and the RAT-Type attribute, adds an abbreviated P-Access-Network-Info (PANI) header to the buffered INVITE. If an MSISDN is included within the AAA, the P-CSCF also adds a P-Subscription-MSISDN header to the buffered INVITE.
4. The P-CSCF caches received attribute values, and forwards the altered INVITE to the IMS core.
5. All subsequent re-INVITES and UPDATES contain the PANI and P-Subscription-MSISDN headers populated with the cached values.

NPLI provisioning triggered by a terminating mobile device typically involves a message exchange similar to the following.

1. Upon receiving the initial INVITE from the IMS core, the P-CSCF sends an AAR command to the PCRF via the Rx interface.

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2. The P-CSCF buffers the INVITE pending the receipt of an AAA command from the PCRF. Assuming NPLI is provided by the IMS core, the AAA contains the 3GPP-User-Location-Info and RAT-Type AVPs. Depending upon network equipment, the AAA may also contain a proprietary MSISDN AVP although this is not usually the case.
3. Upon receiving the AAA, the P-CSCF, using values contained in the 3GPP-User-Location-Info attribute, and RAT-Type attributes, adds a PANI header to the buffered INVITE. If an MSISDN is included within the AAA, the P-CSCF also adds a P-Subscription-MSISDN to the buffered INVITE.
4. The P-CSCF caches received attribute and MSISDN values, and forwards the altered INVITE to the mobile handset.
5. All subsequent 1xx and 200OK, and all subsequent re-INVITES and UPDATES that are generated by the mobile handset contain the PANI and P-Subscription-MSISDN headers populated with the cached values.

## Enhanced Implementation

Enhanced Phase 2 NPLI support is provided by Version S-CZ7.2.0, and later releases. Phase 2 provides for an explicit subscription to NPLI changes resulting in the dynamic update of NPLI as opposed to the single static transaction provided by the initial Phase 1.

Phase 2 NPLI implementation triggered by an originating mobile device typically involves a message exchange similar to the following.

1. Upon receiving the initial INVITE from the mobile handset, the P-CSCF sends an AAR command to the PCRF via the Rx Interface. This AAR contains a Required-Access-Info AVP that requests user location reports as they become available to the PCRF.
2. The P-CSCF buffers the INVITE pending the receipt of an AAA command from the PCRF. The AAA contains a 3GPP-User-Location-Info AVP along with a RAT-Type AVP. Depending upon network design, the AAA may also contain a proprietary MSISDN AVP, although this is not usually the case.
3. Upon receiving the AAA, the P-CSCF, using values contained in the 3GPP-User-Location-Info attribute, and the RAT-Type attribute, adds an extended PANI header to the buffered INVITE. If an MSISDN is included within the AAA, the P-CSCF also adds a P-Subscription-MSISDN header to the buffered INVITE.
4. The P-CSCF caches received attribute values, and forwards the altered INVITE to the IMS core.
5. At any point during the call (from initial session establishment to session termination) the PCRF can send a Re-Auth Request (RAR) that contains updated NPLI conveyed by the 3GPP-User-Location-Info, RAT-Type, IP-CAN-Type, or MSISDN AVPs.
6. After receiving the RAR, the P-CSCF updates the cached attribute values.
7. All subsequent re-INVITES and UPDATES contain the PANI and P-Subscription-MSISDN headers populated with the most recent cache values.

The following call flow depicts a successful, mobile-originated call containing a Re-INVITE. NPLI updates are requested in the initial AAR (P-CSCF-to-PCRF). NPLI is first provided with the initial AAA (PCRF-to-P-CSCF). Updated NPLI is provided with the initial RAR (PCRF-to-P-CSCF). All requests/responses toward the IMS core contain the latest available NPLI.

## 3GPP-User-Location-Info Attribute

### Phase I Implementation

The 3GPP-User-Location-Info Attribute (sub-attribute number 22) is defined in section 16.4.7.2, Coding 3GPP Vendor-Specific RADIUS attributes, of 3GPP TS29.061, Interworking between the Public Land Mobile Network (PLMN) supporting packet based services and Packet Data Networks (PDN). This AVP contains the location type, for example a cell global identifier (CGI), and the actual location data for the specified cell.

### Phase II Implementation

The 3GPP-User-Location-Info AVP (type 22) is defined in section 16.4.7.2, Coding 3GPP Vendor-Specific RADIUS attributes, of 3GPP TS29.061, Interworking between the Public Land Mobile Network (PLMN) supporting packet based services and Packet Data Networks (PDN).



3GPP-User-Location-Info contains the location type, for example a cell global identifier (CGI), and the actual location data for the specified cell.

This attribute is supported by both the Phase 1 and Phase 2 implementations. It occurs in AAR and RAR messages received by the P-CSCF.

The AVP consists of four fields.

The 1-octet 3GPP-type field contains the value 22.

The 1-octet 3GPP-length field contains the attribute length in octets.

The 1-octet Geographic Location Type field identifies the equipment type whose location is being provided.

```
0 = CGI
1 = SAI
2 = RAI
128 = TAI
129 = ECGI
130 = TAI and ECGI
```

Other values are reserved for future use.

The variable length Geographic Location field contains the geographic coordinates of the equipment specified in the Geographic Location Type field.

## IP-CAN-Type AVP (Phase 2)

The IP-CAN-Type AVP (type 1027) specifies the type of Connectivity Access Network to which the user is connected.

This attribute is supported by the Phase 2 implementation. It occurs in RAR messages received by the P-CSCF.

The following values are defined.

```
0 = 3GPP-GPRS
    indicates 3GPP General Packet Radio Service (GPRS) access and is further
    detailed by the RAT-Type AVP that contains an applicable value (not EUTRAN).
1 = DOCSIS
    indicates Data Over Cable Service Interface Specification (DOCSIS) access.
2 = xDSL
    indicates Data Over Cable Service Interface Specification (DOCSIS) access.
3 = WiMAX
    indicates Worldwide Interoperability for Microwave Access (WiMAX -- IEEE
    802.16).
4 = 3GPP2
    indicates CDMA2000 access as specified in 3GPP2 X.S0011 and is further
    detailed by the RAT-Type AVP.
5 = 3GPP-EPS
    indicates 3GPP Evolved Packet System (EPS) access and is further detailed
    by the RAT-Type AVP.
6 = Non-3GPP-EPS
    indicates Evolved Packet System (EPS) based on non-3GPP access technology
    and is further detailed by the RAT-Type AVP.
```

## MSISDN AVP

The Vodafone proprietary MSISDN AVP is supported in both NPLI Phases 1 (Release S-CZ7.1.2) and 2 (Release s-CZ7.2.0 and later).

This attribute can be found in AAR and RAR messages received by the P-CSCF. It is constructed as follows.

```
<Name>MSISDN-VF:21274</Name>
<AvpName>MSISDN-VF</AvpName>
<AvpCode>3701</AvpCode>
```

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```
<VendorId>21274</VendorId>
<MandatoryFlag>>false</MandatoryFlag>
<ProtectFlag>>false</ProtectFlag>
<MayencryptFlag>>false</MayencryptFlag>
<VendorSpecificFlag>>true</VendorSpecificFlag>
<AvpType>utf8String</AvpType>
```

## RAT-Type Attribute

### Phase I Implementation

The RAT-Type Attribute (attribute type 1032) specifies the radio access technology type. For purposes of NPLI, the following values are supported:

- 1000 — Universal Terrestrial Radio Access Network (UTRAN)
- 1001 — GSM Edge Radio Access Network (GERAN)
- 1004 — Evolved UMTS Radio Access Network (EUTRAN)

In the event that a received RAT-Type attribute contains an unsupported type (something other than 1000, 1001, or 1004), an error condition is declared and the default PANI is used.

### Phase II Implementation

This attribute is supported by both the Phase 1 and Phase 2 implementations.

It occurs in AAR and RAR messages received by the P-CSCF.

The following values are supported.

```
0 - WLAN
1 - Virtual (indicates that RAT is unknown)
1000 - Universal Terrestrial Radio Access Network (UTRAN)
1001 - GSM Edge Radio Access Network (GERAN)
1002 - GAN (Generic Access Network)
1003 - HSPA_EVOLUTION (Evolved High Speed Packet Access)
1004 - Evolved UMTS Radio Access Network (EUTRAN)
2000 - CDMA2000_1X (core CDMA2000 wireless air interface)
2001 - HRPD (High Rate Packet Data)
2002 - UMB (Ultra Mobile Broadband)
2003 - EHRPD (Enhanced High Rate Packet Data)
```

In the event that a received RAT-Type attribute contains an unsupported type, an error condition is declared and the default PANI is used.

## NPLI Required-Access-Info AVP (Phase 2)

The Required-Access-Info AVP (type 536) conveys a subscription request for NPLI.

This attribute is supported by the Phase 2 implementation. It occurs in AAR messages originated by the P-CSCF.

The following values are defined.

```
0 - subscribe to NPLI
1 - subscribe to user timezone information
```

## NPLI Specific-Action AVP (Phase 2)

The Specific-Action AVP (type 513) works in conjunction with the Required-Access-Info AVP (type 536) to convey an NPLI subscription request to the PCRF.

This attribute is supported by the Phase 2 implementation. It occurs in AAR messages originated by the P-CSCF.

The following value is used to request NPLI:

```
12 - ACCESS_NETWORK_INFO_REPORT
```

## NPLI Enabling NPLI Notifications

The P-CSCF sends the PCRF AAR messages containing the Specific-Action and Required-Access-Info AVPs to indicate a desired notification/subscription type. Use the following procedure to request NPLI notifications.

1. Move to `ext-policy-server` configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# ext-policy-server
ACMEPACKET(ext-policy-server)#
```

2. Use the `select` command to identify the target PCRF.

```
ACMEPACKET(ext-policy-server)# select
name:
...
...
...
4. PCRF_10A
...
...
...
selection: 4

ACMEPACKET(ext-policy-server)#
```

3. Use the `specific-action-subscription` parameter to specify one or more specific actions. When designating 2 or more actions, enclose them in quotation marks, with the values separated by spaces. Use `access-network-info-report` to enable NPLI support.

```
ACMEPACKET(ext-policy-server)# specific-action-subscription
access-network-info-report
ACMEPACKET(ext-policy-server)#
```

4. Use `done`, `exit` and `verify-config` to complete configuration.

## P-Access-Network-Info SIP Header

The P-Access-Network-Info SIP header is defined in section 4.4 of RFC 3455, *Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)*. This header contains information on the access network that the user handset is using to get IP connectivity, and is typically ignored by intermediate proxies between the SBC that adds the header and the SIP proxy that is providing services. The proxy providing services can inspect the header and make use of the information contained there to provide appropriate services, depending on the value of the header. Before proxying the request onwards, this proxy strips the header from the message.

The P-Access-Network-Info header contains an `access-type` field identifying the radio access technology as received in the `RAT-Type` attribute.

- 3GPP-UTRAN — when the `RAT-Type` AVP value is 1000
- 3GPP-GERAN — when the `RAT-Type` AVP value is 1001
- 3GPP-E-UTRAN — when the `RAT-Type` AVP value is 1004

The `access-type` field is followed by the `access-info` field which identifies the mobile phone cell that provides network access. The location of this cell provides the means to geographically locate connected mobile handsets.

- `utran-cell-id-3gpp` — when the `RAT-Type` AVP value is 1000
- `cgi-3gpp` — when the `RAT-Type` AVP value is 1001
- `utran-cell-id-3gpp` — when the `RAT-Type` AVP value is 1004

The `access-info` field is followed by an EQUAL sign and a `CellID/CGI` (Cell Global Identifier) field that contains an octet string which provides the location value contained in the 3GPP-User-Location-Info AVP. The location value can be either an CGI or an Extended CGI (ECGI).

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- CGI field — when the RAT- Type AVP value is 1000
- CGI field — when the RAT- Type AVP value is 1001
- ECGI field— when the RAT- Type AVP value is 1004

Examples of P-Access-Network-Info headers are as follows:

If the RAT-Type is UTRAN:

```
P-Access-Network-Info: 3GPP-UTRAN;utran-cell-id-3gpp=C359A3913B20E
```

If the RAT-Type is GERAN:

```
P-Access-Network-Info: 3GPP-GERAN;cgi-3gpp=62F8100005C599
```

If the RAT-Type is EUTRAN:

```
P-Access-Network-Info: 3GPP-E-UTRAN;utran-cell-id-3gpp=C359A3913B20E
```

### Abbreviated PANI SIP Header (Phase 1)

NPLI Phase 1 (Version S-CX7.1.2) provides an abbreviated PANI header consisting of three fields -- access-type, access-info, and location-info.

The access-type field identifies the radio access technology as identified by the RAT-Type attribute.

```
3GPP-UTRAN    - when the RAT-Type AVP value is 1000
3GPP-GERAN    - when the RAT-Type AVP value is 1001
3GPP-E-UTRAN - when the RAT-Type AVP value is 1004
```

The access-info field works in conjunction with the location-info field to identify the type and location of the mobile phone cell that currently provides network access. The location of this cell provides the means to geographically locate connected mobile handsets.

```
utran-cell-id-3gpp - when the RAT-Type AVP value is 1000
cgi-3gpp           - when the RAT-Type AVP value is 1001
utran-cell-id-3gpp - when the RAT-Type AVP value is 1004
```

The access-info field is followed by an EQUAL sign and a CellID/CGI (Cell Global Identifier) field that contains an octet string which provides the location value contained in the 3GPP-User-Location-Info AVP. The location value can be either an CGI or an Extended CGI (ECGI).

```
CGI field    - when the RAT-Type AVP value is 1000
CGI field    - when the RAT-Type AVP value is 1001
ECGI field   - when the RAT-Type AVP value is 1004
```

Examples of abbreviated headers are as follows.

If the RAT-Type is UTRAN:

```
P-Access-Network-Info: 3GPP-UTRAN;utran-cell-id-3gpp=C359A3913B20E
```

If the RAT-Type is GERAN:

```
P-Access-Network-Info: 3GPP-GERAN;cgi-3gpp=62F8100005C599
```

If the RAT-Type is EUTRAN:

```
P-Access-Network-Info: 3GPP-E-UTRAN;utran-cell-id-3gpp=C359A3913B20E
```

### Extended PANI SIP Header (Phase 2)

NPLI Phase 2 (Version S-CX7.2.0 and later) provides an extended PANI header as defined in 3GPPTS 24.229, IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3.

Extended PANI header syntax is as shown below.

```
P-Access-Network-Info = "P-Access-Network-Info" HCOLON
                        access-net-spec *(COMMA access-net-spec)
```

```

access-net-spec          = (access-type / access-class) * (SEMI access-
access-info)
access-type              = "IEEE-802.11" / "IEEE-802.11a" /
"IEEE-802.11b" / "IEEE-802.11g" /
                        "IEEE-802.11n" / "3GPP-GERAN" / "3GPP-UTRAN-
FDD" / "3GPP-UTRAN-TDD" /
                        "3GPP-E-UTRAN-FDD" / "3GPP-E-UTRAN-TDD" /
"ADSL" / "ADSL2" / "ADSL2+" /
                        "RADSL" / "SDSL" / "HDSL" / "HDSL2" /
"G.SHDSL" / "VDSL" / "IDSL" /
                        "3GPP2-1X" / "3GPP2-1X-Femto" / "3GPP2-1X-
HRPD" / "3GPP2-UMB" /
                        "DOCSIS" / "IEEE-802.3" / "IEEE-802.3a" /
"IEEE-802.3e" / "IEEE-802.3i" /
                        "IEEE-802.3j" / "IEEE-802.3u" /
"IEEE-802.3ab" / "IEEE-802.3ae" /
                        "IEEE-802.3ah" / "IEEE-802.3ak" /
"IEEE-802.3aq" / "IEEE-802.3an" /
                        "IEEE-802.3y" / "IEEE-802.3z" / GPON/
XGPON1 / "GSTN"/ "DVB-RCS2"/ token
access-class             = "3GPP-GERAN" / "3GPP-UTRAN" / "3GPP-E-
UTRAN" / "3GPP-WLAN" / "3GPP-GAN" /
                        "3GPP-HSPA" / "3GPP2" / token
np                       = "network-provided"
access-info              = cgi-3gpp / utran-cell-id-3gpp / dsl-
location / i-wlan-node-id / ci-3gpp2 /
                        ci-3gpp2-femto / eth-location / fiber-
location / np/ gstn-location /
                        local-time-zone / dvb-rcs2-node-id /
extension-access-info
extension-access-info    = generic-param
cgi-3gpp                 = "cgi-3gpp" EQUAL (token / quoted-string)
utran-cell-id-3gpp       = "utran-cell-id-3gpp" EQUAL (token / quoted-
string)
i-wlan-node-id           = "i-wlan-node-id" EQUAL (token / quoted-
string)
dsl-location              = "dsl-location" EQUAL (token / quoted-string)
eth-location              = "eth-location" EQUAL (token / quoted-string)
fiber-location            = "fiber-location" EQUAL (token / quoted-
string)
ci-3gpp2                 = "ci-3gpp2" EQUAL (token / quoted-string)
ci-3gpp2-femto           = "ci-3gpp2-femto" EQUAL (token / quoted-
string)
gstn-location             = "gstn-location" EQUAL (token / quoted-string)
dvb-rcs2-node-id         = "dvb-rcs2-node-id" EQUAL quoted-string
local-time-zone           = "local-time-zone" EQUAL (token / quoted-
string)

```

## P-Subscription-MSISDN SIP Header

The Proprietary P-Subscription-MSISDN SIP header is supported in both NPLI Phases 1 (Release S-CZ7.1.2) and 2 (Release s-CZ7.2.0 and later).

Its syntax is as follows.

```

P-Subscription-MSISDN = "P-Subscription-MSISDN" HCOLON
                        MSISDN
                        MSISDN          =          msidsn value (string)

```

### Handling User-Endpoint-Provided Location Information

Handling of user-endpoint-provided location information is user configurable. By default, this information is not included in the PANI header constructed by the P-CSCF. This default state can be over-ridden, in which case the PANI header will contain both user-endpoint-provided location information and NPLI.

In the case where a user-endpoint PANI header contains an **np** parameter, the location information contained in the header is ignored, and is not included in the PANI constructed by the P-CSCF.

The following sample Phase 2 PANI headers illustrate the inclusion of both user-endpoint-provided and network-provided location information.

In this case, the UE -provided location information and NPLI matched.

```
P-Access-Network-Info: 3GPP-E-UTRAN-FDD; utran-cell-id-3gpp=26207FFFDF1B9601,
3GPP-E-UTRAN-FDD; utran-cell-id-3gpp=26207FFFDF1B9601; network-provided
```

In this case, the cell-IDs did not match.

```
P-Access-Network-Info: 3GPP-E-UTRAN-FDD; utran-cell-id-3gpp=26207FFFDF1B9601,
3GPP-E-UTRAN-FDD; utran-cell-id-3gpp=26207abcde7A7777; network-provided
```

In this case, neither the network access type, nor the cell type matched.

```
P-Access-Network-Info: 3GPP-E-UTRAN-FDD; utran-cell-id-3gpp=26207FFFDF1B9601,
3GPP-GERAN; cgi-3gpp=26207FFFDF1B9601; network-provided
```

In this case, NPLI was not available.

```
P-Access-Network-Info: 3GPP-E-UTRAN-FDD; utran-cell-id-3gpp=
```

### User-Endpoint-Provided Location Information Configuration

Use the following procedure to include user-endpoint-provided location information in the PANI constructed by the P-CSCF. This capability is disabled by default.

1. Move to sip-config configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

2. **include-ue-loc-info**—Use this parameter to enable the inclusion of user-endpoint-provided location information in PANI headers.

With the parameter enabled, the PANI header contains both user-endpoint-provided location information in addition to NPLI.

By default, **include-ue-loc-info** is disabled.

```
ACMEPACKET(sip-config)# include-ue-loc-info enable
```

3. Use **done**, **exit** and **verify-config** to complete configuration.

### NPLI Emergency Call Processing

Currently, when the P-CSCF sends AAR requests to the PCRF for an emergency call, the P-CSCF does not wait for the response (AAA) to forward the INVITE to the core. Instead, it forwards the INVITE as soon as it sends the AAR to the PCRF. Even if the subsequent AAA reports a failure response, the P-CSCF ignores the failure and allows emergency call processing.

Recent requirements issued by Bundesnetzagentur (BNetzA), the German telecom regulator, affect the processing of emergency calls originated by mobile handsets. The German requirements mandate that INVITES for establishment of emergency calls, be buffered, for some specified period of time, pending receipt of NPLI from the PCRF. If the PCRF fails to provide NPLI within that time period, the P-CSCF forwards the emergency INVITE to the network core with a PANI header populated with both the default-location-str value (if configured) and with any UE-provided

location information (if available, and if configured to do so). If location information is unavailable, the P-CSCF forwards the INVITE without a PANI header.

Use the following procedure to configure a hold-timer that specifies the time (in seconds) that emergency INVITES are buffered while waiting for NPLI from the PCRF. Such a timer ensures compliance with BNetzA regulations. Users who are not subject to BNetzA authority can choose to implement a minimum timer value, which provides the network some period of time to report possibly more accurate location information.

## NPLI Emergency Call Configuration

Use this procedure to buffer emergency INVITES pending the receipt of NPLI from the PCRF. This procedure is required for users subject to German telecommunications regulations. For other users, the procedure is optional.

1. Navigate to sip-config configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

2. **hold-emergency-calls-for-loc-info**—Use this parameter to both enable the buffering of emergency INVITES (thus complying with German regulations), and to specify the time (in seconds) that INVITES are buffered.

By default **hold-emergency-calls-for-loc-info** is set to the value 0, which disables the buffering of emergency INVITES.

Set **hold-emergency-calls-for-loc-info** to a non-zero value, within the boundaries acceptable to the German regulator, to enable the timer.

```
ACMEPACKET(sip-config)# hold-emergency-calls-for-loc-info 5
ACMEPACKET(sip-config)#
```

3. Use **done**, **exit** and **verify-config** to complete this configuration

## Configuring Default PANI Contents

Users can configure default contents of the P-Access-Network-Info header in the event of any failure during the NPLI transfer. By default, such failure results in the generation of an empty header. In some circumstances a more informative header may be of benefit to the user.

Default PANI contents can be configured at either the SIP interface level, or at the realm level. If values are configured at both levels, the realm value takes precedence. A value configured at the sip-interface level takes precedence over a default value (empty string) specified at the realm level.

Use the following procedure to configure default PANI contents.

1. Navigate to either the sip-interface or realm configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

or

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

2. Use the **select** command to specify the target SIP interface or realm.
3. Use the **default-location-string** parameter to specify the default contents of the P-Access-Network-Info SIP header in the event that NPLI is not available.

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```
ACMEPACKET(sip-interface) # default-location-sting "Location services temporarily unavailable."  
ACMEPACKET(sip-interface) #
```

or

```
ACMEPACKET(realm-config) # default-location-sting "Location services not unavailable."  
ACMEPACKET(realm-config) #
```

4. Use **done**, **exit**, and **verify-config** to complete configuration.

## Caveats

Within environments that provide NPLI support:

- **reserve-incomplete** (ext-policy-srv mode) — set to enabled
- **optimize-aar** (ext-policy-srv mode) — set to disable

## Rf Interface Features

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### Node-Functionality AVP Support

The Oracle Communications Session Border Controller sends the Node-Functionality (862) AVP in all Rf ACR messages.

The Node-Functionality AVP indicates the function that the message's source plays in the network. The CDF/CGF function that collects the ACR messages can use the information in the Node Functionality AVP for billing or analysis purposes.

In an IMS network, the Oracle Communications Session Border Controller may perform the following functions: P-CSCF, E-CSCF, IBCF, BGCF (when configured as an Oracle Communications Session Router). In fact, the system might perform different roles simultaneously, so that on a call-by-call basis, the value in the Node-Functionality might change.

To accurately reflect multiple, simultaneous functions that the Oracle Communications Session Border Controller performs, the value inserted into the Node-Functionality AVP may be defined per realm. The node functionality value for a call's ACR is taken from the configuration in the ingress realm. Each realm may only be marked with a single Node Functionality value.

The system can still be configured with a single, global Node Functionality value. This is done in the SIP config's node functionality parameter. When configured, all system-generated ACRs include this value. However, if the node functionality parameter is also configured in a realm config, the ingress realm's node functionality value supersedes the global value.

The node-functionality in the **realm-config** may be configured with an empty string (). This indicates that this realm should revert to the global node-functionality value.

### Node Functionality AVP Configuration

To configure a global Node Functionality AVP value:

1. Navigate to the **sip-config** configuration element.

```
ACMEPACKET# configure terminal  
ACMEPACKET(configure) # session-router  
ACMEPACKET(session-router) # sip-config  
ACMEPACKET(sip-config) #
```

2. Type **select** to begin configuring this object.

```
ACMEPACKET(sip-config) # select  
ACMEPACKET(sip-config) #
```



3. **node-functionality** — Enter a global value to insert into the Node-Functionality AVP when the system sends ACRs over the Rf interface to an appropriate destination. The default is P-CSCF. Valid values are:
  - P-CSCF
  - BGCF
  - IBCF
  - E-CSCF
4. Save and activate your configuration.

## Node Functionality Per Realm Configuration

To configure a Node Functionality value per realm:

1. Navigate to the **realm-config** configuration element.

```
ACMEPACKET(session-router)# exit
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```

2. Select the ingress realm where calls generate ACR's whose node functionality AVP will be marked as configured.

```
ACMEPACKET(realm-config)# select
identifier:
1: realm01 left-left:0 0.0.0.0
selection: 1
ACMEPACKET(realm-config)#
```

3. **node-functionality** — Enter the value to insert into the Node-Functionality AVP when the system sends ACRs for calls that enter the system from this realm. The default is empty which uses the global node functionality value configured in the sip-config configuration element. Valid values are:
  - P-CSCF
  - BGCF
  - IBCF
  - E-CSCF
4. Save and activate your configuration.

## 2733 - Optimization of AAR Messages

When the Oracle Communications Session Border Controller acts as a P-CSCF, the Diameter Rx interface updates bandwidth and addressing changes during a session. Many transactions between the P-CSCF and a PCRF server trigger a new SDP offer resulting in the transmission of an P-CSCF-initiated AA-Request (AAR) sent to the PCRF for the purpose of updating bearer parameters. However, not all transactions need to be reported to the PCRF, for example transactions that do not carry any bandwidth or addressing changes. In such instances, the issuance of an AAR is unnecessary and wasteful.

AAR message optimization (the suppression of unnecessary AARs) is enabled by setting the **optimize-aar** parameter in the ext-policy-server configuration element to **enable**.

If optimization is enabled, AARs are suppressed when

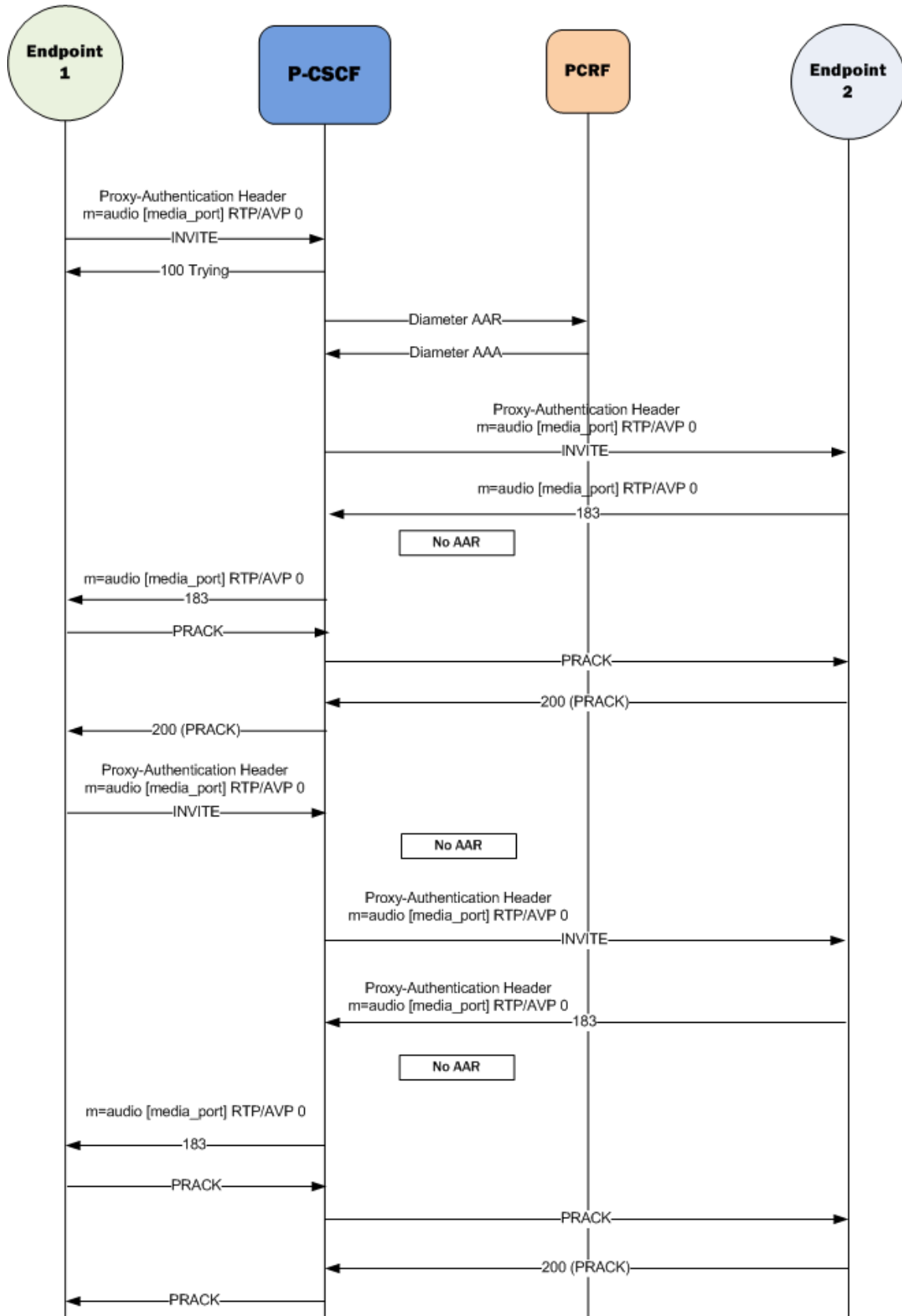
- the originating INVITE does not contain a proxy-authorization header;
- the codec provided in the m-line of an SDP response has not changed since the last SDP message;
- the INVITE sent to the P-CSCF does not contain the IP address and port of the endpoint, or the SDP response to the P-CSCF does not contain the same codec in the m-line as the INVITE, the P-CSCF only sends an AAR message upon receiving the first SDP response;
- a reINVITE or UPDATE request or response is sent to the P-CSCF and neither the bandwidth or codec has changed — such reINVITEs or UPDATES may include SDP offers/answers with a=sendonly, a=recvonly, or a=inactive

## External Policy Servers

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These SDP parameters are directly related to the provision of Supplementary Services such as Call Hold. Consequently, the ACLI provides an options command that allows the generation of an AAR when a reINVITE or UPDATE offer or answer contains a sendonly, recvonly, or inactive parameter, even when AAR optimization is enabled.

The following call flow indicates where the AAR optimization omits AARs:



- Endpoint 1 sends an INVITE containing a Proxy-Authorization header.
  - The P-CSCF sends Diameter AAR.
  - The PCRF replies with Diameter AAA.

## External Policy Servers

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- The P-CSCF forwards the INVITE to Endpoint 2.
- Endpoint 2 sends a 183 response.
  - The m-line and codec have not changed, and therefore no AAR is sent.
- Endpoint 1 sends a reINVITE. The parameters of the SDP have not changed, and therefore no AAR is sent.

## AAR Message Configuration

To optimize the use of AAR messages:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **ext-policy-server** and press Enter.

```
ACMEPACKET(media-manager)# ext-policy-server
ACMEPACKET(ext-policy-server)#
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI select command) the external policy server that you want to edit.

4. **optimize-aar**—Type enabled to optimize the use of AAR messages.
5. Save and activate your work.

## AF-Application-Identifier AVP Generation

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By default, the Oracle Communications Session Border Controller populates the AF-Application-Identifier AVP with the hostname parameter of the external policy server object that receive's the AAR for the call.

The Oracle Communications Session Border Controller can populate the AF-Application-Identifier AVP with the IMS Communication Service Identifier (ICSI). 3GPP TS 24.173 defines the ICSI format as:

```
urn:urn-xxx:3gppservice.ims.icsi.mmtel
```

An ICSI appears in one of the following three SIP headers in an INVITE:

- P-Assured-Service
- P-Preferred-Service
- Accept-Contact

This AF-Application-Identifier AVP is inserted into an AAR and sent to the PCRF when the Oracle Communications Session Border Controller (as P-CSCF) receives an INVITE.

When the Oracle Communications Session Border Controller has an ICSI value from the access UE and the core, it uses the value received from the core. ICSI values received from the network supercede a configured default ICSI value. When the INVITE is received from the access side, ICSI values are taken from the access side until values are received from the core side. If no valid ICSI value is received in a message, and the Oracle Communications Session Border Controller is not configured with a default-icsi value, the external policy server object's hostname is used for the ICSI value sent to the PCRF. If an INVITE does not contain SDP, no AAR is generated.

AF-Application-Identifier AVP generation occurs for access and core initiated calls.

## AVP Generation on Initial INVITE from UE

The Oracle Communications Session Border Controller receives the UE's INVITE. When configured to send an AAR on receipt of the INVITE, the AF-Application-Identifier AVP is populated with the contents of one of the following headers (in order of precedence) if present:

- P-Assured-Service (this header is only expected from trusted UEs)
- P-Preferred-Service
- Accept-Contact

If none of the above three headers are present, the default value of AF-Application-Identifier shall be used.

If the Oracle Communications Session Border Controller is not configured to send an AAR on receipt of an INVITE, the value from the 3 headers will be cached for later AF-Application-Identifier generation.

### AVP Generation on response to Initial INVITE from UE

The Oracle Communications Session Border Controller receives the response (1xx or 200 OK) to the original INVITE. When configured to send an AAR on receipt of the response, the AF-Application-Identifier AVP is populated with the contents of one of the following headers (in order of precedence) if present:

- P-Assured-Service
- P-Preferred-Service
- Accept-Contact header is not expected

Values obtained in this step supercede any value set from the initial INVITE step. If none of the three headers are present, the value determined in the INITIAL invite step is used for the AF-Application-Identifier AVP contents.

### AVP generation on reINVITE from UE

The Oracle Communications Session Border Controller can receive a re-INVITEs from the core. The Oracle Communications Session Border Controller populates the AF-Application-Identifier AVP with one of the following headers (in order of precedence) if present:

- P-Assured-Service (this header is only expected from trusted UEs)
- P-Preferred-Service
- Accept-Contact

If none of the three headers are present, the default AF-Application-Identifier value is used.

Responses to the Re-Invites will be treated in the similar manner.

### Examples

Example 1:

Received from access UE	Received from core	Sent to PCRF
INVITE with SDP P-A-S, P-P-S, or A-C header contains "P1"		AAR with AF-A-I AVP contains "P1"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P2"	AAR with AF-A-I AVP contains "P2"
re-INVITE with SDP P-A-S, P-P-S, or A-C header contains "P3"		AAR with AF-A-I AVP contains "P2"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P4"	AAR with AF-A-I AVP contains "P4"

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Example 2:

Received from access UE	Received from core	Sent to PCRF
INVITE with no SDP		No AAR generated
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P2"	AAR with AF-A-I AVP contains "P2"
re-INVITE with SDP P-A-S, P-P-S, or A-C header contains "P3"		AAR with AF-A-I AVP contains "P2"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P4"	AAR with AF-A-I AVP contains "P4"

Example 3:

Received from access UE	Received from core	Sent to PCRF
INVITE with no SDP		No AAR generated
	200 OK with SDP No P-A-S, P-P-S, or A-C headers	AAR with AF-A-I AVP contains default AF-A-I value.
re-INVITE with SDP P-A-S, P-P-S, or A-C header contains "P3"		AAR with AF-A-I AVP contains "P3"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P4"	AAR with AF-A-I AVP contains "P4"

Example 4:

Received from access UE	Received from core	Sent to PCRF
INVITE with SDP P-A-S, P-P-S, or A-C header contains "P1"		AAR with AF-A-I AVP contains "P1"
	200 OK with no SDP	AAR with AF-A-I AVP contains "P1"
re-INVITE with SDP P-A-S, P-P-S, or A-C header contains "P3"		AAR with AF-A-I AVP contains "P1"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P4"	AAR with AF-A-I AVP contains "P4"

Example 5:

Received from access UE	Received from core	Sent to PCRF
INVITE with no SDP		No AAR generated
	183 with SDP No P-A-S, P-P-S, or A-C headers	AAR with AF-A-I AVP contains ext-policy-server's "hostname"
PRACK with SDP P-A-S, P-P-S, or A-C header contains "P1"		AAR with AF-A-I AVP contains "P1"
	200 OK with no SDP	No AAR generated
	200 OK (INVITE) with SDP P-A-S, P-P-S, or A-C header contains "P4"	AAR with AF-A-I AVP contains "P4"

Example 6:

Received from access UE	Received from core	Sent to PCRF
INVITE with SDP P-A-S, P-P-S, or A-C header contains "P1"		AAR with AF-A-I AVP contains "P1"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P2"	AAR with AF-A-I AVP contains "P2"
	re-INVITE with SDP P-A-S, P-P-S, or A-C header contains "P3"	AAR with AF-A-I AVP contains "P3"
200 OK with SDP P-A-S, P-P-S, or A-C header contains "P4"		AAR with AF-A-I AVP contains "P3"

### AVP Generation on Initial INVITE from core

The Oracle Communications Session Border Controller receives the original INVITE. When configured to send an AAR on receipt of the INVITE, the AF-Application-Identifier AVP is populated with the contents of one of the following headers (in order of precedence) if present:

- P-Assured-Service
- P-Preferred-Service
- Accept-Contact

If none of the above three headers are present, the default value of AF-Application-Identifier shall be used.

If the Oracle Communications Session Border Controller is not configured to send an AAR on receipt of an INVITE, the value from the 3 headers will be cached for later AF-Application-Identifier generation.

**AVP Generation on response to initial INVITE from core**

The Oracle Communications Session Border Controller receives the response (1xx or 200 OK) to the original INVITE. When configured to send an AAR on receipt of the response, if none of the three headers (P-Assured-Service, P-Preferred-Service, or Accept-Contact) were present in the original INVITE, the Oracle Communications Session Border Controller populates the AF-Application-Identifier AVP with the contents of one of the following headers (in order of precedence) if present:

- P-Assured-Service (this header is only expected from trusted UEs)
- P-Preferred-Service
- Accept-Contact header is not expected

If none of the three headers are present, the value determined in the INITIAL invite step is used for the AF-Application-Identifier AVP contents.

**AVP generation on reINVITE from core**

The Oracle Communications Session Border Controller can receive re-INVITES from the core. The Oracle Communications Session Border Controller populates the AF-Application-Identifier AVP with one of the following headers (in order of precedence) if present:

- P-Assured-Service (this header is only expected from trusted UEs)
- P-Preferred-Service
- Accept-Contact

If none of the three headers are present, the default AF-Application-Identifier value is used.

Responses to the Re-Invites will be treated in the similar manner.

Example 1:

Received from core	Received from access UE	Sent to PCRF
INVITE with SDP P-A-S, P-P-S, or A-C header contains "P1"		AAR with AF-A-I AVP contains "P1"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P2"	AAR with AF-A-I AVP contains "P1"
re-INVITE with SDP P-A-S, P-P-S, or A-C header contains "P3"		AAR with AF-A-I AVP contains "P3"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P4"	AAR with AF-A-I AVP contains "P3"

Example 2:

Received from core	Received from access UE	Sent to PCRF
INVITE with no SDP		No AAR generated
	200 OK with SDP	AAR with AF-A-I AVP contains "P2"



Received from core	Received from access UE	Sent to PCRF
	P-A-S, P-P-S, or A-C header contains "P2"	
re-INVITE with SDP P-A-S, P-P-S, or A-C header contains "P3"		AAR with AF-A-I AVP contains "P3"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P4"	AAR with AF-A-I AVP contains "P3"

Example 3:

Received from core	Received from access UE	Sent to PCRF
INVITE with no SDP		No AAR generated
	200 OK with SDP No P-A-S, P-P-S, or A-C headers	AAR with AF-A-I AVP contains ext-policy-server hostname.
re-INVITE with SDP P-A-S, P-P-S, or A-C header contains "P3"		AAR with AF-A-I AVP contains "P3"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P4"	AAR with AF-A-I AVP contains "P3"

Example 4:

Received from core	Received from access UE	Sent to PCRF
INVITE with SDP P-A-S, P-P-S, or A-C header contains "P1"		AAR with AF-A-I AVP contains "P1"
	200 OK with no SDP	AAR with AF-A-I AVP contains "P1"
re-INVITE with SDP P-A-S, P-P-S, or A-C header contains "P3"		AAR with AF-A-I AVP contains "P3"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P4"	AAR with AF-A-I AVP contains "P3"

Example 5:

## External Policy Servers

Received from core	Received from access UE	Sent to PCRF
INVITE with no SDP		No AAR generated
	183 with SDP No P-A-S, P-P-S, or A-C headers	AAR with AF-A-I AVP contains ext-policy-server's "hostname"
PRACK with SDP P-A-S, P-P-S, or A-C header contains "P1"		AAR with AF-A-I AVP contains "P1"
	200 OK with no SDP	No AAR generated
	200 OK (INVITE) with SDP P-A-S, P-P-S, or A-C header contains "P4"	AAR with AF-A-I AVP contains "P1"

Example 6:

Received from core	Received from access UE	Sent to PCRF
INVITE with SDP P-A-S, P-P-S, or A-C header contains "P1"		AAR with AF-A-I AVP contains "P1"
	200 OK with SDP P-A-S, P-P-S, or A-C header contains "P2"	AAR with AF-A-I AVP contains "P1"
	re-INVITE with SDP P-A-S, P-P-S, or A-C header contains "P3"	AAR with AF-A-I AVP contains "P1"
200 OK with SDP P-A-S, P-P-S, or A-C header contains "P4"		AAR with AF-A-I AVP contains "P4"

## Diameter: CLF

The Oracle Communications Session Border Controller supports the e2 interface over a Diameter connection acting as a P-CSCF communicating with a CLF. The application ID field must be set to 16777231 to run the e2 reference point.

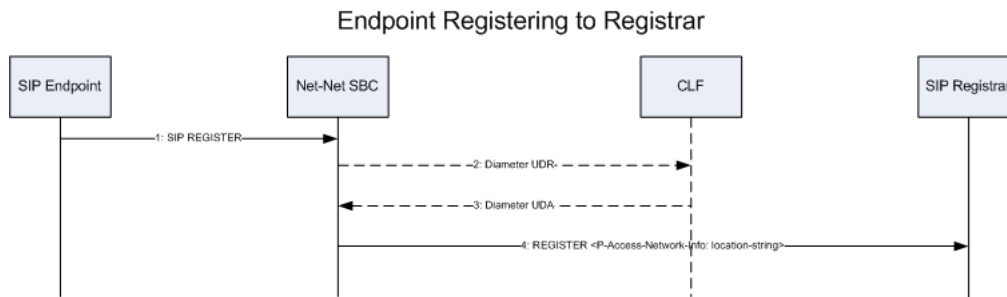
A Connectivity Location Function (CLF) maintains mappings between endpoints with dynamically assigned IP addresses and their physical location. The Oracle Communications Session Border Controller, acting as a P-CSCF, is the intermediary device between a registering endpoint and a CLF. The CLF thus validates and tags a registering endpoint, and the Oracle Communications Session Border Controller applies the CLF's actions.

### CLF Behavior

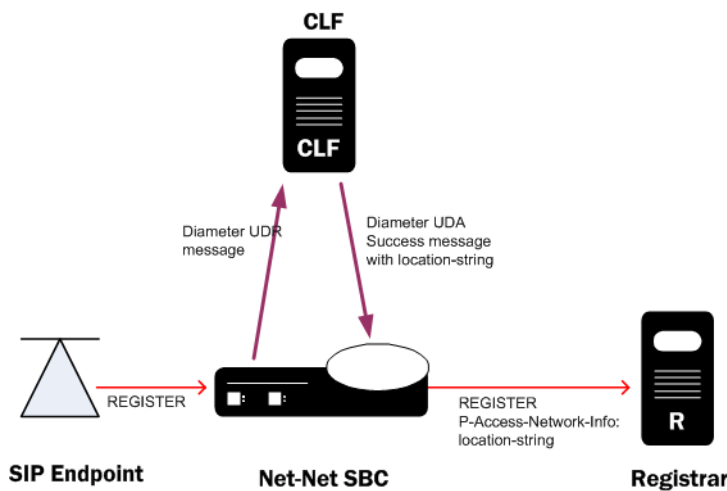
The Oracle Communications Session Border Controller and a CLF only interact with each other when an endpoint registers or re-registers. The Oracle Communications Session Border Controller, acting as the P-CSCF, is the first SIP device that the REGISTER message reaches. Upon receiving the REGISTER message(1), the Oracle

Communications Session Border Controller queries the CLF using the Diameter protocol. The endpoint's (public) IP address and port, and the Oracle Communications Session Border Controller's IP information are sent to the CLF in a Diameter User-Data-Request (UDR) message(2).

The CLF responds to the Oracle Communications Session Border Controller with a Diameter User-Data-Answer (UDA) message(3). If the request is approved, then the CLF also sends a location-string value to be inserted in one of the SIP headers. The Oracle Communications Session Border Controller inserts a P-Access-Network-Info header containing the location-string into the incoming REGISTER message and forwards this message(4) to the SIP registrar/I/S-CSCF.



The Oracle Communications Session Border Controller inserts this P-Access-Network-Info header into all subsequent SIP messages from this endpoint as they are forwarded into the core network. The P-Access-Network-Info header is inserted into all SIP requests and responses except for ACK and CANCEL messages. For all boundaries where SIP messages pass from trusted to untrusted SIP interfaces or session agents, the Oracle Communications Session Border Controller will strip out the P-Access-Network-Info header as expected.



If the CLF responds with a Reject UDA message, the Oracle Communications Session Border Controller rejects the registration, and sends a 503 - Service Unavailable message back to the registering endpoint. In this way, the CLF can be used for admission control.

The Oracle Communications Session Border Controller communicates with the CLF solely for retrieving location information from the CLF, and not for notifying the CLF about an endpoint's registration state or activity. When an endpoint's registration ends, either through a normal expiration, getting rejected by the registrar, or through specific de-registering or error conditions, the Oracle Communications Session Border Controller deletes the locally cached registration location string. The Oracle Communications Session Border Controller does not inform the CLF about any registrations that have been deleted.

### P-Access-Network-Info Header Handling

The P-Access-Network-Info header is created and populated according to the following rules:

1. If the CLF returns an Accept UDA message with a location string, the Oracle Communications Session Border Controller inserts the location string into a P-Access-Network-Info header in the outgoing REGISTER message.

## External Policy Servers

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2. If the CLF returns an Accept UDA message without a location string, the Oracle Communications Session Border Controller inserts the configured default string into a P-Access-Network-Info header in the outgoing REGISTER message.
3. If the CLF returns an Accept UDA message without a location string and no location string is configured on Oracle Communications Session Border Controller, the outgoing REGISTER message is forwarded out of the Oracle Communications Session Border Controller, but no P-Access-Network-Info header is created for the REGISTER message.

### P-CSCF PANI Enhancements

The Oracle Communications Session Border Controller always adds PANI headers to the egress SIP message with the value set to either:

- The location of information received from the remote CLF agent
- The default-location-string

When you enable **allow-pani-for-trusted-only** in the SIP config configuration element, the Oracle Communications Session Border Controller only forwards PANI headers to and from trusted sources or if access-info does not have network-provided in the header. A trusted domain is determined by the trust-level of the corresponding realm or SIP interface.

### CLF Re-registration

The Oracle Communications Session Border Controller will send a new UDR message to the CLF to request a new location string if any of the following events occur:

1. The endpoint's contact address changes.
2. The SIP Register message's Call-ID header changes.
3. The endpoint's public IP Address or UDP port changes.
4. The endpoint connects to a different SIP interface, port, or realm on the Oracle Communications Session Border Controller than it did in the initial REGISTER message.
5. The registration expires in the Oracle Communications Session Border Controller's registration cache.

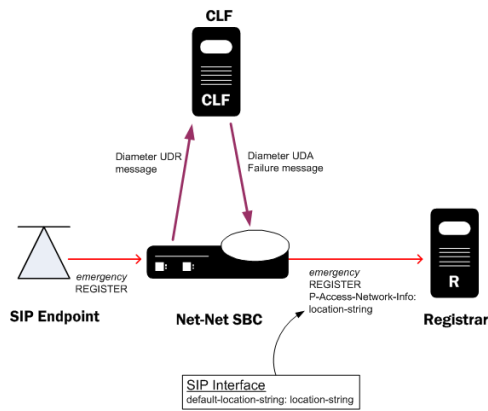
### CLF Failures

If a Diameter connection fails, the Oracle Communications Session Border Controller will continually try to re-establish the connection. Endpoints that are already registered will stay registered unless they timeout or if the registrar rejects their refreshes. When the Diameter connection has not been established, and an endpoint registers on a SIP interface that is configured to use CLF, the Oracle Communications Session Border Controller forwards new REGISTER messages to the registrar using the default location string.

### CLF Emergency Call Handling

The Oracle Communications Session Border Controller allows emergency calls into the network even if the endpoint that places the emergency call is not registered. In the expected fashion, the Oracle Communications Session Border Controller will query the CLF first for an incoming emergency call sourced from an unregistered endpoint. If the CLF response is successful, then the Oracle Communications Session Border Controller will insert the string returned from the CLF into a P-Access-Network-Info header, and insert this header into the emergency call's REGISTER message. If no location string is returned with a successful CLF response, the default location string is inserted into P-Access-Network-Info header.

If the CLF's response is to reject the emergency call, the Oracle Communications Session Border Controller will insert the configured default location string into the P-Access-Network-Info header and forward the emergency call's REGISTER message toward the registrar. For emergency calls where the endpoint has already successfully registered, the call will be routed into the network using the expected methods for emergency call routing.



If the Diameter connection to the CLF is down, emergency calls from un-registered endpoints are still allowed into the network using the default string inserted into the emergency messages.

## CLF Diameter e2 Error Handling

When the Oracle Communications Session Border Controller receives an error, as defined in RFC 3588, in a UDA message from a CLF, it replies to the UA with a 503 Service Unavailable message. There are exceptions for certain values embedded in either the Result-Code AVP or Experimental Result-Code AVP.

### Result-Code AVP

If the CLF sends a Result-Code AVP in a UDA message with a value of `DIAMETER_UNABLE_TO_COMPLY` (5012), the Oracle Communications Session Border Controller forwards the REGISTER message using the default configured location string in the P-Access-Network-Info header. If the configured default string is left blank the Oracle Communications Session Border Controller forwards the SIP REGISTER message without the P-Access-Network-Info header.

### Experimental-Result-Code AVP

If the CLF sends a grouped Experimental Result AVP in a UDA message that is not successful, as per RFC 3588, the Oracle Communications Session Border Controller performs the following actions with respect to receipt of these two result code values:

- `DIAMETER_ERROR_USER_UNKNOWN(5001)`: The Oracle Communications Session Border Controller inserts the configured default-string in for the P-Access-Network-Info header when forwarding the SIP message.
- `DIAMETER_USER_DATA_NOT_AVAILABLE(4100)`: The Oracle Communications Session Border Controller inserts the configured default-string in for the P-Access-Network-Info header when forwarding the SIP message.

For both cases, if the configured default string is left blank the Oracle Communications Session Border Controller forwards the SIP REGISTER message without the P-Access-Network-Info header. For all other failure result codes, the Register is rejected.

### Diameter CLF e2 Error Bypass

When a CLF returns error messages i.e., any other Diameter answer than `DIAMETER_SUCCESS`, you can configure the Oracle Communications Session Border Controller to override the default behavior and still forward REGISTER messages with P-Asserted-Network-Id header containing the **default-location-string** parameter value. If the **default-location-string** parameter value is empty, the REGISTER message is forwarded without the P-Access-Network-Info header. This error bypass feature is configured by setting the **permit-on-reject** parameter to **enabled** for an external policy server configuration element with application-mode set to e2.

# CLF-only AVPs

---

## Globally-Unique-Address AVP

When endpoints registering through the Oracle Communications Session Border Controller reside in nested realms, the Oracle Communications Session Border Controller allows you to set the realm that appears in the Globally-Unique-Address AVP in Diameter UDR messages destined for a CLF.

The **ingress-realm-location** parameter in the external policy server configuration specifies whether to use the realm on which a signaling message arrived, or to use that realm's parent. If you choose to use the parent realm, the Oracle Communications Session Border Controller uses the one associated with the SIP interface on which the REGISTER request arrived.

### e2 Interface

You can configure a value to be sent in the Address-Realm AVP (communicated in the Globally-Unique-Address AVP) for the Diameter e2 interface. This AVP is sent on a per-realm basis in the Location Information Query (UDR) query the Oracle Communications Session Border Controller (as a P-CSCF) sends to the CLF.

The CLF maintains details about IP connectivity access sessions associated with user equipment connected in the network. This CLF supports the standardized Diameter e2 interface with an Application Function; from it, the application function (i.e., the Oracle Communications Session Border Controller in the role of P-CSCF) retrieves location information and other data related to the access session. To do so, the AF sends a UDR containing Global-Unique-Address and Requested Information attributes. Then the CLF returns a Location Information Response (UDA) containing either a success result code with location information or an error result code.

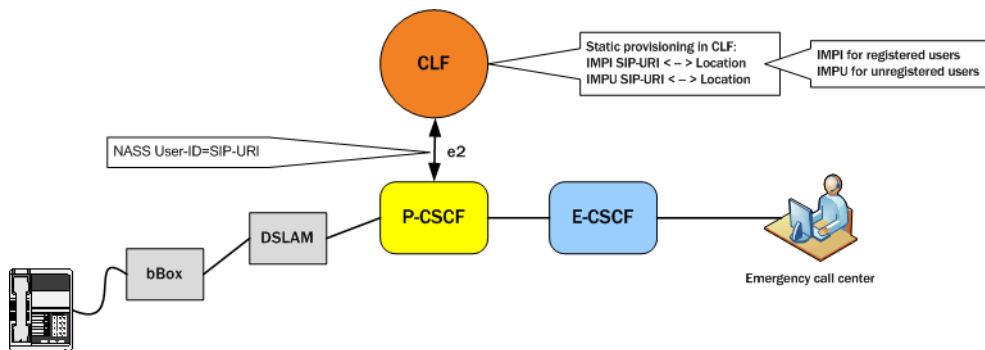
In the UDR's Global-Unique-Address, the Oracle Communications Session Border Controller currently supports the mandatory parameters: the Framed-IP-Address AVP and the Address-Realm AVP. The address realm AVP is the realm address in FQDN form, populated based on the realm on which a REGISTRATION arrived or the SIP interface. With nested realms configured, using a realm for this value can quickly become complicated.

You can configure the value sent in the Address-Realm AVP on a per-realm basis. You set the external policy server's **ingress-realm-location** parameter to the **diam-address-realm** value, pointing the Oracle Communications Session Border Controller to the associated realm from which it will learn Address-Realm AVP information. This access realm (or child realm, the realm on which enter registration came in) has its **diam-e2-address-realm** value set to the value with which to populate the Address-Realm AVP for the outgoing UDR message.

## CLF e2 Interface User-Name AVP Support

In compliance with ETSI ES 283 -35 V1.2.1, the Oracle Communications Session Border Controller's e2 interface can query the CLF using the SIP endpoint's IP address or its NASS User-ID. The system's e2 interface uses the SIP-URI to query the CLF by including the User-Name AVP in the User Data Request (UDR). The CLF can then furnish the P-CSCF (i.e., the Oracle Communications Session Border Controller) with the INSEE code in the Location-Information AVP in its User Data Answer (UDA).

This diagram shows how this capability works. The Oracle Communications Session Border Controller acts as the P-CSCF.



- Registering users—When it receives a registration request, the Oracle Communications Session Border Controller checks the incoming SIP interface to determine CLF use. If CLF use is unnecessary, the Oracle Communications Session Border Controller forwards the registration message to its destination.

When CLF is required, the Oracle Communications Session Border Controller selects the AVPs to send the CLF, including the User-Name AVP before sending it a UDR. A **none** setting for this parameter means the Oracle Communications Session Border Controller does not include the User-Name AVP in any UDRs. The Oracle Communications Session Border Controller adds the User-Name AVP to the UDR if the **user-name-mode** parameter in the external policy server configuration is set to:

- **endpoint-id**—IP address of the registering endpoint is sent as the payload for the User-Name AVP
- **public-id**—SIP-URI portion of the TO header from the register message is sent as the payload for the User-Name AVP
- **auth-user**—Username attribute of the Authorization header from the register is sent as the payload for the User-Name AVP; if there is no authorization header, the Oracle Communications Session Border Controller will not consult the CLF and will forward the registration message
- Unregistered users—When it receives an INVITE request, the Oracle Communications Session Border Controller checks the incoming SIP interface to determine if it should use an external policy server. If it does not need to use an external policy server, the Oracle Communications Session Border Controller forwards the INVITE message to its destination.

When the Oracle Communications Session Border Controller does need to use an external policy server, it also checks to determine if the INVITE is in a registration and if location data in the registration cache is available for that endpoint. These requirements being met, the Oracle Communications Session Border Controller inserts the P-Access-Network-Info header with the location string into the INVITE it forwards to the destination. If these requirements are not met, the Oracle Communications Session Border Controller consults the CLF before forwarding the INVITE. The following describe the impact of the user-name-mode setting in such instance:

- **endpoint-id**—IP address of the endpoint that issued the INVITE is sent as the payload for the User-Name AVP
- **public-id and auth-user**—SIP-URI portion of the INVITE is sent as the payload for the User-Name AVP:

With a P-Asserted-Id header present in the INVITE, the Oracle Communications Session Border Controller uses the first PAID header (if there are multiple PAID headers)

Without a P-Asserted-Id header present in the INVITE's P-Preferred-Identity header, the Oracle Communications Session Border Controller uses the first PPI (if there are multiple PPI headers)

With neither P-Asserted-Id nor P-Preferred-Identity header present, the Oracle Communications Session Border Controller uses the From header

## HA Functionality

The location strings generated by the CLF are replicated on the standby Oracle Communications Session Border Controller in an HA pair. This is required so that a Oracle Communications Session Border Controller in an HA pair can instantly continue processing calls using the previously learned CLF information.

# Configuring Diameter-based CLF

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## SIP Interface Configuration

In the following configuration examples, we assume that your baseline configuration passes SIP traffic, with the Oracle Communications Session Border Controller in the role of an Access Oracle Communications Session Border Controller. In this example, you will configure additions to the ream configuration and the new external policy server configuration.

### SIP Interface Configuration for CLF Support

To configure the SIP interface configuration for CLF support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # sip-interface
ACMEPACKET (sip-interface) #
```

4. Type **select** and the number of the pre-configured sip interface you want to configure for CLF. This should be the ingress SIP interface for

```
ACMEPACKET (sip-interface) # select 1
ACMEPACKET (sip-interface) #
```

5. **ext-policy-svr**—Set this parameter to the same name as the External Policy Server configured that you configured for the CLF server.
6. **default-location-string**—Set this parameter to the default location string you want inserted into a P-Access-Network-Info header for when the CLF server does not return a unique location string.
7. Save your work using the ACLI **done** command.

### External Policy Server Configuration

To configure the external policy server for use with a CLF:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # media-manager
```

3. Type **ext-policy-server** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # ext-policy-server
ACMEPACKET (ext-policy-server) #
```

4. **name**—Set this parameter to an applicable name for this CLF instance of the external policy server. The value of this parameter will be entered in the SIP interface configuration element to reference this CLF.
5. **state**—Set this parameter to **enabled** to enable this CLF. The default value is **enabled**. The valid values are:
  - enabled | disabled
6. **operation-type**—Set this parameter to **admission-control** for the Oracle Communications Session Border Controller to communicate with a CLF. The default value is **disabled**.
7. **protocol**—Set this parameter to **DIAMETER** to connect with a CLF via the DIAMETER protocol.



8. **address**—Set this parameter to the IP address or FQDN of the CLF.
9. **port**—Set this parameter to the port which the CLF uses for Diameter transactions. Port 3868 is the default Diameter port. (The default value is **80**.) The valid range is:
  - Minimum—0
  - Maximum—65535
10. **realm**—Set this parameter to the realm where the CLF exists.
11. **permit-conn-down**—Enable or disable the Oracle Communications Session Border Controller’s ability to permit calls if there is no connection to the external policy server. The default value is **disabled**. The valid values are:
  - enabled | disabled
12. **permit-on-reject**—Change this parameter to enabled if you want the Oracle Communications Session Border Controller to forward the session on at a best-effort. Leave this parameter set to disabled (default), if you want the Oracle Communications Session Border Controller to deny the session on attempts to revert to the previously-requested bandwidth. (The default value is disabled.) Valid values are:
  - enabled | disabled
13. **disconnect-on-timeout**—Disable this parameter to prevent timeouts triggered by Gate-Set or Gate-Delete message sequences between the Oracle Communications Session Border Controller and a policy server from tearing down their connection. Retaining the default (enabled) allows all timeouts to tear down and re-establish the TCP connection. The valid values are:
  - **Default enabled**
  - **enabled | disabled**
14. **product-name**—Enter text string that describes the vendor-assigned name for the CLF. This parameter is required.
15. **application-mode**—Enter the type of interface you want to use. Your choices are: **Rq**, **Rx**, **Gq**, **e2**, and **none**.
16. **application-id**—Enter a numeric application ID that describes the interface used to communicate with the CLF. The default value is zero (**0**). For the e2 CLF reference point, the application id is **16777231**.
17. **framed-ip-addr-encoding**—Enter the format of the Frame-IP-Address (AVP 8) value in Diameter messages. The default value is **octet-string**. The valid values are:
  - **ascii-string**—Example: 192.168.10.1
  - **octet-string**—Example: 0xC0A80A01
18. **dest-realm-format**—Enter the format you want to use for the Destination-Realm AVP. The default value is **user\_with\_realm**. The valid values are:
  - user\_with\_realm | user\_only | realm\_only
19. **ingress-realm-location**—Set this parameter to configure the child realm or its parent for the Address-Realm in the Globally-Unique-Address AVP in Diameter UDR messages that the Oracle Communications Session Border Controller sends to the policy server. There are two choices:
  - **realm-in** (default)—This setting means that the Oracle Communications Session Border Controller will use the same realm on which the REGISTRATION request arrived.
  - **sip-interface**—This setting means that the Oracle Communications Session Border Controller will use the realm associated with the SIP interface on which the REGISTRATION request arrived.
  - **diam-address-realm**—For the e2 interface, this value enables configurable Address-Realm AVPs. This setting points the system to the associated realm from which it will learn Address-Realm AVP information. The default is realm-in.  
  
Remember to set the diam-e2-address-realm parameter in the realm this parameter points to.
20. **user-name-mode**—Determines how the User-Name AVP is constructed. Used primarily with e2 based CLF functionality. There are four choices.

## External Policy Servers

- none—the system does not include the User-Name AVP in any UDRs (default)
  - endpoint-ip—IP address of the registering endpoint is sent as the payload for the User-Name AVP
  - public-id—SIP-URI portion of the TO header from the register message is sent as the payload for the User-Name AVP
  - auth-user—Username attribute of the Authorization header from the register is sent as the payload for the User-Name AVP; if there is no authorization header, the Oracle Communications Session Border Controller will not consult the CLF and will forward the registration message
- 21. domain-name-suffix**—Enter the suffix you want to use for Origin-Realm and Origin-Host AVPs that have a payload string constructed as a domain name Your value can be any string, to which the Oracle Communications Session Border Controller will prepend with a dot if you do not include one. The default value is **.com**.
- 22. gate-spec-mask**—With this parameter, you can configure the Oracle Communications Session Border Controller to use a mask comprised entirely of zeros (0). The default value is 255. This parameter sets the value to use for the COPs pkt-mm-3 interface. This interface maintains a persistent TCP connection to the external policy server, even without responses to requests for bandwidth. This permits calls to traverse the Oracle Communications Session Border Controller even though the external policy server either fails to respond, or rejects the session.
- Default 255
  - Values Min: 0 / Max: 255
- 23. allow-srv-proxy**—Set to enabled in order to include the proxy bit in the header. The presence of the proxy bit allows the Oracle Communications Session Border Controller to tell the external policy server whether it wants the main server to handle the Diameter message, or if it is okay to proxy it to another server on the network (disabled). The default is enabled. The valid values are:
- enabled | disabled
- 24. reserve-incomplete**—Set this parameter to **enabled** if you want the Oracle Communications Session Border Controller to send a message to the CLF that does not include the endpoint's true port number. A value of 0 will be used for the port number. The default value is **enabled**. The valid values are:
- enabled | disabled
- 25. watchdog-ka-timer**—Enter the interval in seconds of Oracle Communications Session Border Controller-initiated watchdog/keep-alive messages. Valid values range between 0 and 65535 seconds.
- 26. trans-expires**—Set the amount of time, in seconds, that the Oracle Communications Session Border Controller waits before performing an expiration if a Diameter base protocol transaction does not occur. The default value is 15 seconds. Valid values range between 1 and 15.
- 27.** Save your work using the ACLI **done** command.

## CLF Debugging

A new argument has been added to the show command for viewing CLF statistics. From the user prompt, type **show ext-clf-svr**.

```
ACMEPACKET# show ext-clf-svr
```

```
14:17:14-114
```

```
EBM Status
```

	Active	-- Period --		----- Lifetime -----		
		High	Total	Total	PerMax	High
Client Trans	0	0	0	0	0	0
Server Trans	0	0	0	0	0	0
Sockets	0	0	0	0	0	0
Connections	0	0	0	0	0	0

	---- Lifetime ----		
	Recent	Total	PerMax
CLF Requests	0	0	0
CLF Admits	0	0	0
CLF Errors	0	0	0
CLF Rejects	0	0	0

CLF Expires	0	0	0
CLFD Errors	0	0	0

You can also refer to the log.ebmd log file located in the /ramdrv/logs/ directory on the Oracle Communications Session Border Controller. This file must be retrieved via FTP or SFTP.

## E2 CLF Configurable Timeout

You can configure a transaction timer value for each external policy server based on the round-trip and response times for each specific policy server. Defining this transaction timer value allows you to avoid situations when policy/DIAMETER servers fail to respond to the Oracle Communications Session Border Controller's requests, ensuring that transactions proceed in the desired amount of time and avoiding undesirable delays in set-up times.

The time you set for the DIAMETER CLF must be from 1 second to 15 seconds. For purposes of backward compatibility, 15 seconds is the default. It is recommended that you set the timer at 6 seconds or greater in accordance with the smallest DIAMETER Watchdog Request/Answer (DWR/DWA) per RFC 3588.

### Activating a Configuration with the E2 CLF Timeout Defined

When you define a value for the E2 CLF timeout and activate your configuration, the change in time will only affect new client transactions. Previously existing client transactions will use the value in effect prior to your having activated the configuration.

### E2 CLF Timeout Configuration

To define the E2 CLF timeout for an external policy server:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **ext-policy-server** and press Enter.

```
ACMEPACKET(media-manager)# ext-policy-server
ACMEPACKET(ext-policy-server)#
```

4. **trans-expiry**—Enter the time (in seconds) for the E2 CLF timeout, a transaction timer value for each external policy server based on the round-trip and response times for each specific policy server. The default for this parameter is 15 seconds; your entry must be between 1 and 15.

5. Save your work.

## Diameter Policy Server High Availability

### External Policy Server HA Cluster

The Oracle Communications Session Border Controller can provide external policy server redundancy through a combination of multiple servers being returned in one FQDN query and maintaining state of these servers.

When multiple IP addresses are returned in a response to a DNS query for Diameter-based external policy servers, the Oracle Communications Session Border Controller assembles the IP addresses into an HA cluster which provides redundancy for Diameter-based applications. This feature is enabled by configuring the external policy server's **address** parameter with an FQDN.

## External Policy Servers

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The number of servers maintained in the HA cluster is configured with the **max connections** parameter. Thus if the **max connections** parameter is set to 3, the Oracle Communications Session Border Controller maintains 1 active external policy server with 2 back up servers.

## Standby Server Prioritization

The Oracle Communications Session Border Controller looks at the priority and weight fields in the DNS response to create the preferred order of the primary, secondary, tertiary, and quaternary Policy Servers. These 4 addresses are known as the top-level PSs in the cluster.

The Oracle Communications Session Border Controller uses the priority and weights according to RFC 2782. However, weights only apply when multiple servers share the same priority value. If the priorities are different, then the Oracle Communications Session Border Controller will not use weights. If the priorities are the same, then weight of the two (or more) contested servers is used to determine which one to use.

## Server States

An HA cluster can contain up to 4 Policy Servers, where the TCP/Diameter connection is established and monitored. Diameter session traffic is only sent to the active policy server in the cluster. The policy servers exist in the following states:

- active-TCP and Diameter connection established. Oracle Communications Session Border Controller using this server for policy decisions. The policy server with the highest priority/weight begins in this state.
- standby- TCP and Diameter connection established. Server in standby mode.
- inactive - Diameter connection not successfully established. Oracle Communications Session Border Controller tries to reconnect to inactive servers.

## HA Cluster Refresh

The Oracle Communications Session Border Controller sends follow-up SRV queries to the DNS server to refresh the list of available policy servers in the cluster in the following instances:

- DNS cache expires after the TTL is exceeded
- a new policy server with FQDN for an address is configured, saved, and activated on the Oracle Communications Session Border Controller
- after an SPU switchover, the newly active SPU performs a DNS query

When the Oracle Communications Session Border Controller re-queries the DNS server for Diameter external policy servers, the cluster is refreshed with the new/changed servers. Policy server priority is also reconfigured based on newly returned priorities and weights. Upon a cluster refresh, the Oracle Communications Session Border Controller:

- closes connections with standby policy servers that are no longer in the cluster
- creates connections with policy servers which are new in the set

If the currently active policy server remains a member of the cluster after a refresh, it remains active even if its priority has changed. If the active-before-the-refresh policy server is not a member of the cluster after a DNS refresh, the Oracle Communications Session Border Controller gracefully closes the connection to this server. The Oracle Communications Session Border Controller then installs the highest priority server as the active policy server.

## DNS Failure

If the Oracle Communications Session Border Controller fails to get a response from the DNS server or does not receive at least one IP address in the SRV RR, it continues to send SRV queries periodically, starting with 5 seconds and doubling the interval for every sequential failure, until it receives a valid response. While waiting for a successful DNS response, the Oracle Communications Session Border Controller uses the existing Diameter servers in the DNS cache.

## Policy Server Failover

The active external policy server fails over to the highest priority standby server when:

- the TCP connection is closed due to a RST or FIN
- the Diameter connection (CER/CEA exchange) could not be established
- a configured number of consecutive Diameter message timeouts occur. This number is configured in the **max timeouts** parameter. The max timeouts parameter refers to all Diameter messages except for the following:
- You can configure Diameter keepalive message time-outs separately from all other Diameter messages by setting **watchdog ka timer** in the external policy server configuration element. This will failover the active policy server based on a timeout value for DWR messages.
- You can configure Diameter STR message time-outs separately from all other Diameter messages by setting the **str-retry=<timeout number>** option in the external policy server configuration element. This will failover the active policy server based on a unique timeout value for STR messages.

If the Oracle Communications Session Border Controller sends an AAR/STR message to the active policy server and then switches to a different policy server, any new Diameter messages related to that session are sent to the same policy server as long as it is not inactive. If that server becomes inactive, messages will be sent to the new policy server. The new policy server however will not recognize the sessionID and discard the request.

## External Policy Server High Availability Configuration

To configure the external bandwidth manager:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # media-manager
```

3. Type **ext-policy-server** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # ext-policy-server
ACMEPACKET (ext-policy-server) #
```

4. **address**—Enter the IP address or FQDN of the external policy server. To use external policy server redundancy, you must enter the address as an FQDN.
5. **port**—If you configure this parameter, it will override the port returned in a DNS reply.
6. **max-connections**—Set the number of servers to maintain in an external policy server cluster. Valid values range from 1 - 20.
7. **srv-selection-strategy**—Leave this parameter at its default, Failover setting.
8. **max-timeouts**—Set the number of request timeouts before the Oracle Communications Session Border Controller sets this external policy server to inactive. You can separately set the number of DWR timeouts that trigger a server to be inactive as an option. Valid values range from 0 - 200.
9. Save and activate your configuration.

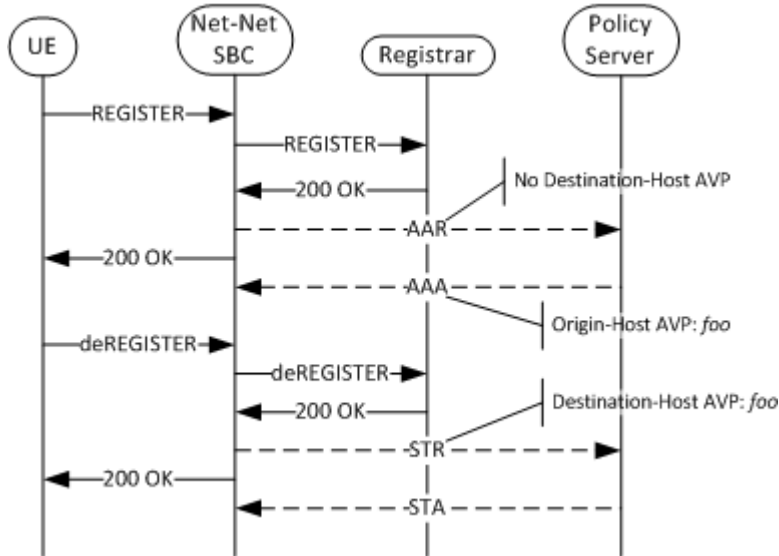
## Diameter Multi-tiered Policy Server Support

When accepting a call in a 2 tiered policy server environment, the Oracle Communications Session Border Controller queries the Tier-1 policy server, running in stateless mode, for AAR-type requests. The Tier-1 PS, identifies an appropriate Tier-2 policy server for every incoming Diameter session. The mapping of session-ID to Tier-2 PS is performed between the Tier-1 and Tier-2 policy servers. This architecture is used to increase overall performance.

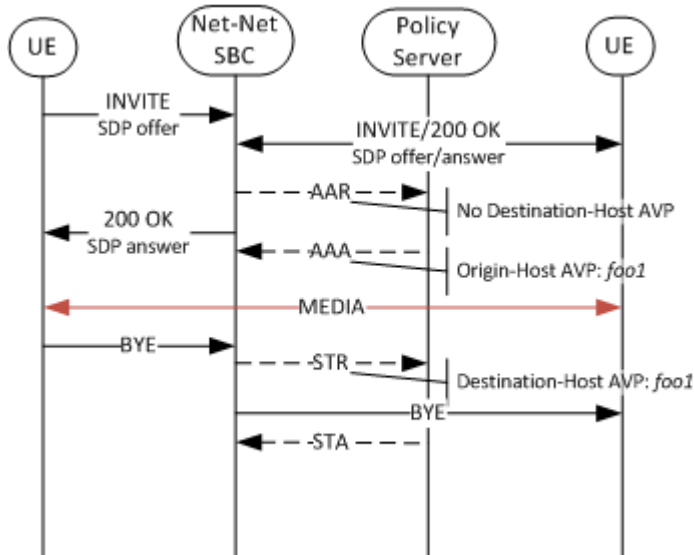
Two-tiered policy support is implemented per user registration and per call. To act in the P-CSCF role with a stateless Tier-1 policy server, the system recalls the Origin-Host AVP value returned in the AAA answer response to the initial Diameter AAR request. The Origin-Host-AVP value is then inserted into the Destination- Host AVP in subsequent Diameter messages (AAR, STR) corresponding to that user registration or call. Set the cache-dest-host parameter to enabled in the ext-policyserver configuration element to enable this feature.

## External Policy Servers

The following call flow is an example two tiered PS support for user registration. While the initial AAR's Destination-Host AVP is empty, the system inserts the value of the received Origin-Host AVP into the registration-terminating STR.



The following call flow is an example of a two tiered PS support for a call. If this call occurs within the scope of the previous registration, the value inserted into the Destination-Host AVP will still be unique per session. Therefore, it is not the value learned from the AAA in the REGISTER, previously illustrated; it is the value learned from the AAA responding to the INVITE setting up the call.



In the case of a reINVITE changing an aspect of the call like the codec use, foo1 is inserted into the AAR's Destination-Host AVP. If the Origin-Host AVP value, as received by the system, changes mid session for a call, all subsequent Destination-Host AVP values are updated with the change.

## Diameter Multi-tiered Policy Server Support

To configure Multi-tiered policy server support:

1. In Superuser mode, type configure terminal and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type media-manager and press Enter to access the media-related configurations.

```
ACMEPACKET(configure)# media-manager
```

3. Type `ext-policy-server` and press Enter.

The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# ext-policy-server
ACMEPACKET(ext-policy-server)#
```

4. `cache-dest-host`—Set this parameter to enabled for the system to participate as a P-CSCF in a multi-tiered policy server environment.
5. Save and activate your configuration.

## Diameter Message Manipulations

The Oracle Communications Session Border Controller can perform manipulations on all grouped and non-grouped AVPs. This is referred to as Diameter Manipulation Rules (DMR). A message manipulation is the ability to search for a predefined string within an AVP and then replace it with another value. This is similar to the Oracle Communications Session Border Controller's header manipulation rules functionality.

A diameter manipulation configuration element is defined by a name parameter. You can optionally add a description field to the diameter manipulation. Within each diameter manipulation you can configure multiple diam manipulation rule subelements. The manipulation rule subelements are the configuration where AVPs are identified, searched, and in which the data is replaced.

The user applies diameter manipulations to external policy servers. These manipulations affect traffic between the Oracle Communications Session Border Controller and the applicable policy server.

### Manipulation Rule

Creating a manipulation rule is divided into three parts, defining the message type and AVP where the manipulation is performed, defining how the search on the AVP is performed, and defining what to replace the found string with.

You must first define the name parameter of the diam manipulation rule configuration element. Optionally you can add a `descr avp` code parameter that is a description of this manipulation rule.

#### Naming Diameter Manipulations

The Oracle Communications Session Border Controller restricts the way you can name a diameter-manipulation rule. Specifically, observe the rules below when naming manipulation elements:

- Character limit - diameter manipulation rule names cannot be longer than 24 characters.
- Numeric characters - diameter manipulation rule names must not start with a numeric character.
- Special characters - Special characters are not supported within diameter manipulation rule names, with the exception of the underscore and hyphen characters.
- Capital letter characters - The Oracle Communications Session Border Controller includes reserved keywords that are named using all-capital letters. To avoid conflicts between keywords and diameter manipulation rules, do not configure diameter manipulation rule names using ll capital letters.

Note that, although diameter-manip-rule and avp-header-rule names have the same character-type restrictions, they do not have a character limit.

#### Message Based Testing

When the Oracle Communications Session Border Controller first receives a message applicable for manipulation, it checks if the message type as **request**, **response**, or **all** as configured in the `msg type` parameter. The Oracle Communications Session Border Controller continues to look at the message command code. Matching values are defined by configuring the `msg cmd` code parameter with a numeric value. You can enter a single value, multiple comma-separated values, or you can leave this parameter blank to indicate all message codes.

### AVP Search Value

After the Oracle Communications Session Border Controller has identified the messages where it can look for an AVP, the avp code must be defined with a numeric AVP value to be searched. Also the AVP data type is defined so Oracle Communications Session Border Controller knows how to correctly parse the AVP once found. This is configured in the avp type parameter with valid values of: octet-string, octet-hex, integer32, unsignedint32, address, utfstring, diameteruri, or enumerated.

The comparison type is defined so that the Oracle Communications Session Border Controller knows the correct way to determine if the match value appears in the avp code. Valid comparison types are:

- Case-sensitive—The comparison-type of both case-sensitive and case-insensitive literally compares the value contained in the match-value against the received value.
- Case-insensitive—The comparison-type of both case-sensitive and case-insensitive literally compares the value contained in the match-value against the received value.
- pattern-rule—the match-value is treated as a regular expression.
- boolean—Used when it is necessary to compare the results of two or several manipulation rules with varying logic (e.g. if (\$rule1 & (\$rule2 | \$rule3))). When the comparison-type is set to boolean, the match-value will be evaluated as a boolean expression.

Finally, the match operation is configured by defining a match value, which is the string to find. The Oracle Communications Session Border Controller evaluates if the match value is found in the avp code AVP. You may also leave the match value empty for the DMR to be applied on the AVP without testing for a match.

### Reserved Keywords

The Oracle Communications Session Border Controller employs certain reserved keywords as a short hand for configuration/message parameters. These keywords are as follows:

**HOSTNAME**—This keyword refers to the agent hostname this rule is being referenced by. If the rule is applied to a realm/interface then the value of the hostname keyword will be an empty string. If the rule is applied to the group, then the hostname for the agent picked will be used.

**ORIGINREALM**—This keyword refers to the Origin-Realm value for the configured realm/interface. If the rule is applied to a Diameter Director Agent, then the origin-realm value is derived from the Diameter Director Interface the agent belongs to.

**ORIGINHOST**—This keyword refers to the Origin-Host value for the configured realm/interface. If the rule is applied to a Diameter Director Agent, then the origin-host value is derived from the Diameter Director Interface the agent belongs to.

### Actions on Found Match Value

When the match-value is found, the Oracle Communications Session Border Controller references the action parameter. This is configured as either **none**, **add**, **delete**, **replace**, **store**, **diameter-manip**, **find-replace-all**, **log** or **group-manip** and indicates the action to perform on the string. If the match-value is not found, the Oracle Communications Session Border Controller continues processing the message without any AVP manipulation. These actions mean the following:

#### **none**

None disables a manipulation rule.

#### **add**

This action inserts the value from the **new value** parameter, creates a new AVP of the specified type and adds it to the list of AVPs at the specified position. The message length and padding are re-computed to account for this newly added AVP.



## delete

This action removes the specified AVP from the list of AVPs being operated on. The message length and padding will be re-computed to account for this deleted AVP.

## replace

This action substitutes the existing value with the **new value** parameter. The message length and padding and AVP length and padding will be re-computed to account for changes. This is mostly applicable to variable length AVP types such as octet-string.

## store

Each manipulation rule has the ability to store the data that was contained in the AVP as a string. This is useful for creating conditional logic to make decisions whether to execute other manipulation rules or to duplicate information within the Diameter message.

Every manipulation rule performs an implicit store operation prior to executing the specified action type. All store operations are based on regular expression patterns configured in the **match value**. The information that is stored in the rule is the resultant of the regular expression applied against the specified string. The **comparison-type** is ignored when the action is set to store as the Oracle Communications Session Border Controller assumes that the **match value** is a regular expression. Conditional logic cannot be used to make a decision whether to perform a store operation or not; storing always occurs. Values stored in a manipulation rule are referred to as user defined variables.

## diameter-manip

When the action is set to **diameter-manip**, the Oracle Communications Session Border Controller executes the diameter-manipulation **name** configured in the **new value**. The diameter-manipulation name in the **new value** must match another diameter-manipulation name exactly (case is sensitive).

diameter-manip action type is primarily to reuse diameter-manipulations that may be common to other use cases. A diameter-manip action should never call back to itself either directly or indirectly through a different diameter-manipulation.

## find-replace-all

The **find-replace-all** action searches the object's string for the regular expression defined in the match-value and replaces every matching occurrence of that expression with the value supplied in the **new value**. If the regular expression contains sub-groups, a specific sub-group can be specified to be replaced by adding the syntax `[[:n:]]` at the end of the expression, where n is the sub-group index (zero-based). When the action is find-replace-all, the comparison-type is ignored and the match-value is always treated as a regular expression.

## group-manip

The **group manip** action is used to manipulate AVPs inside grouped AVPs. For this diameter manipulation, you must set the avp-type to **grouped**.

The **group manip** action is similar to the **diameter manip** action in that the Oracle Communications Session Border Controller executes the diameter-manipulation configured in the new value.

There is an important difference between **group manip** and diameter manip. The **diameter-manip** action is context insensitive, meaning when it jumps from one diameter-manipulation to the next diameter-manipulation, it starts looking for the specified AVP from the top of the message.

The **group manip** action is context sensitive, meaning when the processing jumps from one diameter-manipulation to the next diameter-manipulation, it will look for the specified AVP within the grouped AVP. In short, the **group manip** works at an AVP level. All actions are allowed in the subsequent manipulations that are invoked, with the key difference being that those manipulation rules will be applied to the current grouped AVP in the chain. Thus it is possible to apply manipulation to an AVP at any level in the hierarchy.

Consider the following examples:


## External Policy Servers

In order to replace the experimental-result > experimental-result-code AVP value from 5002 to 3002, a **group manip** can be configured as follows:

```
diam-manipulation
  name                manipExpRslt
  description
  diameter-manipulation-rule
    name              expRslt
    avp-code          297
    descr-avp-code
    avp-type          grouped
    action            group-manip
    comparison-type   case-sensitive
    msg-type          response
    msg-cmd-codes     316,317,318
    match-value
    new-value         expRsltCode
  last-modified-by   admin@console
  last-modified-date 2011-09-13 18:50:33
diam-manipulation
  name                expRsltCode
  description
  diameter-manipulation-rule
    name              expRsltCode
    avp-code          298
    descr-avp-code
    avp-type          unsignedint32
    action            replace
    comparison-type   case-sensitive
    msg-type          response
    msg-cmd-codes     316,317,318
    match-value       5002
    new-value         3002
  last-modified-by   admin@console
  last-modified-date 2011-09-13 18:56:14
```

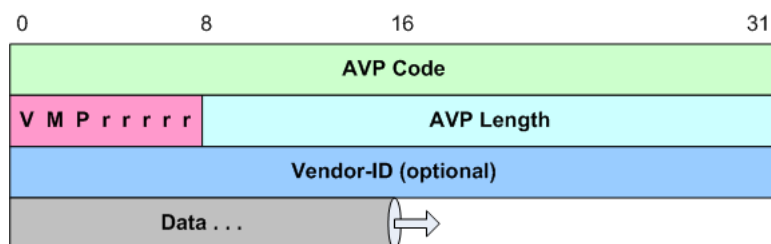
Further, if you want to add a new AVP called AvpD at the following location in the chain of AVPs Message: GrpAvpA > GrpAvpB > GrpAvpC > AvpD, then the manipulation chain would look like this

- diameter-manipulation (name=grpAvpA, action=group-manip, new-value=grpAvpB)
- diameter-manipulation (name=grpAvpB, action=group-manip, new-value=grpAvpC)
- diameter-manipulation (name=grpAvpC, action=group-manip new-value=AvpD)
- diameter-manipulation (name=AvpD action=add new-value="added new value")

 **Note:** using diameter-manip from inside the group-manip chain may have unexpected impact and must be avoided.

## AVP Header Manipulation

In addition to manipulating AVPs, you can manipulate the AVP header itself. After performing AVP DMR, the AVP length and padding is recomputed. This is crucial for scenarios where a vendor-id is added or removed from the header. This functionality is configured in the avp header rules sub element. The following represents a single AVP's header:



## AVP Flag Manipulation

You can manipulate the AVP flags by configuring the **header-type** parameter to **avp-flags**, this invokes operation on the flags byte in the AVP header. AVP flags are 1 byte long, where the first 3 bits represent (1) vendor, (2) must and (3) protected. The last 5 bits are reserved.

The vendor flag is critical to consider here, since it has interdependency with Vendor-Id field in the header shown above. As per RFC 3588, The 'V' bit, known as the Vendor-Specific bit, indicates whether the optional Vendor-ID field is present in the AVP header. When set, the AVP Code belongs to the specific vendor code address space. Please find specific details about the rest of the flags in RFC3588 Section 4.1.

When manipulating AVP flags, a bitwise comparison is performed between the received value and the **match value**. For ease of configuration, the **match value** is configured as a comma-separated enumerated list of vendor, must, and protected. So the **new value** and the **match value** will be used to indicate what bit in the **avp-flag** to operate on. If the **match value** is empty, the configured action is performed without any matching tests. In addition, the new value is configured using the same enumerations. The AVP header rules configuration element appears as follows:

```
avp-header-rules
  name          replaceAvpFlags
  header-type   avp-flags
  action        replace
  match-value   must,protected
  new-value     must
```

According to the example configuration, the Oracle Communications Session Border Controller makes a positive match when only the must and protected bits are set in the received avp-flags. If only the 'M' bit is set, then the match fails, the Oracle Communications Session Border Controller continues to the next header-rule.

When the match is successful (or if the **match value** is left empty), the configured **action** is performed. Consider all following actions applicable to the avp header rules sub element:

- none— no action will be performed
- add—the flags specified in the **new value** are enabled in the header, and state of the existing flags will not be changed.

If **new value** is empty, the add operation will not be performed.

If the **new value**=vendor, and the 'V' bit was not originally set, then the 'V' bit is now be set including a vendor-id of 9148 inserted into AVP. 9148 is the Acme Packet vendor-id assigned by IANA. It is expected that a second header-rule will be used to change this to the desired vendor-id.

- replace—all the received avp-flags will be reset. The values in the **new value** parameter will be set.

If the **new value** is empty, the replace operation will not be performed.

If the **new value**=vendor, and the 'V' bit was not originally set, then the 'V' bit will now be set and also a vendor-id of 9148 (Acme Packet's vendor-id) is added to the AVP header. A second header-rule may be used to change this to the desired vendor-id.

If the **new value** does not contain vendor, and the 'V' bit was originally set, then the 'V' bit will be cleared and the vendor-id will also be set to 0 effectively removing it from the AVP header.

- delete—all flags configured in **new value**, will be deleted from the AVP header

If the **new value** is empty, then no flags are deleted.

If the particular flag is already empty, then it will be skipped. For example, if the **new value**=must and 'M' bit is not set, after applying the DMR the 'M' bit will still be not set.

If the **new value**=vendor, then the 'V' bit will be cleared (if not cleared already) and the vendor-id is set to 0, effectively removing the vendor-id from the avp-header.

### vendor-id Manipulation

You can manipulate the Vendor ID value in the AVP header by configuring the **header-type** parameter to **avp-vendor-id**. This performs the DMR manipulation on the 4-byte vendor-id in the AVP header. AVP vendor id is present in the AVP header only when the ‘V’ bit is set in the flags. This is important because the DMR application relies upon the bit being set to determine where the data payload begins.

The **avp-vendor-id** search invokes an unsigned integer comparison between the received value and the **match-value**. If the **match-value** is empty, the configured action is performed without doing any match.

For the case where **match-value** is non-empty, as in the following example, the DMR engine checks whether the ‘V’ bit is set in the received header. If not, then the vendor id is not present either and the comparison is unsuccessful. If the ‘V’ bit is set, and the match succeeds, the match is successful. (An unsuccessful match has the DMR proceed to the next header-rule.)

```
avp-header-rules
  name      replaceAvpFlags
  header-type  avp-vendor-id
  action    add
  match-value 9148
  new-value  10415
```

When the match is successful (or if the **match value** is left empty), the configured **action** is performed.

Consider all following actions:

- none—no action will be performed
- add—a configured vendor-id value in the **new-value** parameter is added to the AVP header and the ‘V’ bit set to indicate it’s presence. If you prefer to set the ‘V’ bit in an AVP, it is better to do an **avp-vendor-id** action first and then manipulate the rest of the flags.

If the **new-value** is empty, the add operation is not performed.

If a vendor-id already exists in the AVP header, then it is replaced by **new-value**.

- replace—the existing vendor-id value is replaced with the **new-value**.

If the **new-value** is empty, the replace operation is not performed.

If the vendor-id does not exist in the header, then one will be added with the **new-value** and the ‘V’ bit is set to indicate its presence.

- delete—the vendor-id will be removed from the AVP header and ‘V’ bit will be reset to indicate its absence.

If the **new-value** is empty, then the delete operation will not be performed.

If the vendor-id does not exist, then the delete operation is not performed.

### Multi-instance AVP Manipulation

Some AVPs can appear multiple times within a message. To choose a specific occurrence of an AVP, the **avp code** parameter supports indexing logic. By default, the Oracle Communications Session Border Controller operates on all instances of the specified AVP. However, after configuring an **avp-code**, you can specify in square brackets, a specific instance of the AVP on which to operate on. The indexing is zero-based. For example,

```
diameter-manipulation-rule
  name      manip
  avp-code  293 [2]
```

Special characters that refer to non-discrete values are:

- Last occurrence—**avp-code**[^]
- All—**avp-code** [\*]

The last (^) character is used to refer to the last occurring instance of that AVP. Any [^] refers to the first matching header that matches the specified conditional matching criteria. All [\*] is the default behavior, and operates on all

headers of the specified-type. For **group manip** action, the AVP index applies to the instance within that grouped AVP.

## ACLI Instructions

### Diameter Manipulation

To configure a diameter manipulation configuration element:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the media-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type **diameter-manipulation** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# diameter-manipulation
ACMEPACKET(diameter-manipulation)#
```

4. **name**—Enter the name of this Diameter manipulation element.
5. **description**—Enter an optional description for this Diameter manipulation.
6. Type **done** and continue.

### Manipulation Rule

1. Type **diameter-manip-rules** to continue and enter individual policy attributes. The system prompt changes to let you know that you can begin configuring individual parameters.
2. **name**—Enter the name of this manipulation rule. This value will be referenced in when creating a chain of rules.
3. **descr-avp-code**—Enter a description of the AVP code to manipulate.
4. **msg-cmd-code**—Enter the command code number of the message to execute the manipulation on.
5. **msg-type**—Set this to the type of message this manipulation applies to as **request**, **response**, or **all**.
6. **avp-code**—Enter the AVP by code number where this manipulation applies. You can add a multi instance identifier to the end of the avp code value, enclosed in brackets.
7. **avp-type**—Set this to the data type of the content of the match field. Refer to the Diameter standards document for the encodings of individual AVPs. Valid values are:

```
none | octet-string | octet-hex | integer32 | unsignedint32 | address | diameteruri | enumerated | grouped
```

8. **match-value**—Enter the value within the match-field to find and make a positive match on.
9. **action**—Enter either **none**, **add**, **delete**, **store**, **diameter-manip**, **group-manip**, **find-replace-all**, or **replace** as the action to take after making a positive match on the previously entered match-value.
10. **new-value**—Enter the value that should be added or replaced in the old match-value's position.
11. Type **done** and continue.

### AVP Header Manipulation

1. Type **avp-header-rules** to configure AVP header manipulation rules. The system prompt changes to let you know that you can begin configuring individual parameters.
2. **name**—Enter the name of this AVP Header manipulation rule.
3. **header-type**—Set this to either **avp-flag** or **avp-vendor-id** depending on which part of the AVP header you are manipulating.
4. **action**—Enter either **none**, **add**, **delete**, or **replace** as the action to take after making a positive match on the previously entered match-value.
5. **match-value**—Enter the value in the AVP flag field or Vendor ID field to match against.

## External Policy Servers

When matching in the avp flag field, then match-value is interpreted as comma-separated list of enumerated values <vendor,protected,must>. When matching in the Vendor ID field, then match-value is interpreted as 32 bit unsigned integer <1-4294967295>

6. **new-value**—Enter the new value when the match value is found. The resultant new value is entered as the match value is configured.
7. Type **done** to save your work.

### Applying the Manipulation

You can apply a diameter manipulation by name to an external policy server configuration. This element contains the two applicable parameters: diameter-in-manip and diameter-out-manip.

1. To navigate to external policy server configurations from Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # media-manager
```

3. Enter the configuration element where you wish to apply the manipulation.

4. Type **ext-policy-server** and press Enter.

```
ACMEPACKET (media-manager) # ext-policy-server
ACMEPACKET (ext-policy-server) #
```

5. Type **select** and then choose the pre-configured external policy server you want to configure.

```
ACMEPACKET (ext-policy-server) # select
<hostname>:
1: ext-pol-svr-1
2: ext-pol-svr-2
selection: 1
```

You may now add a Diameter manipulation to one or both directions of message flows.

6. **diameter-in-manip**—Enter a name of an existing diameter manipulation to apply as received by the Oracle Communications Session Border Controller on this element.
7. **diameter-out-manip**—Enter a name of an existing diameter manipulation to apply as forwarded from the Oracle Communications Session Border Controller on this element.
8. Type **done** and continue.

## Diameter Manipulation Example - Supported Features AVP

This section shows you a configuration example for diameter manipulation rules. This section shows the configuration for the rule that the Oracle Communications Session Border Controller applied, and sample results of the manipulation. These examples present configurations as an entire list of fields and settings for each ruleset, nested header rules and nested element rules. If a field does not have any operation within the set, the field is shown with the setting at the default or blank.

For this manipulation rule, the Oracle Communications Session Border Controller inserts a Supported Features AVP into every request.

This is a sample of the configuration:

```
diameter-manipulation
  name          diamManip1
  description
  diameter-manip-rule
    name        rule1
    avp-code    628
    descr-avp-code
    avp-type    grouped
    action      add
```

```

msg-type request
msg-cmd-code 265
comparison-type case-sensitive
match-value
new-value
diameter-manip-rule
name rule2
avp-code 628
descr-avp-code
avp-type grouped
action group-manip
msg-type any
msg-cmd-code
comparison-type case-sensitive
match-value
new-value diamManip2

```

This second rule, which defines a new value for the first rule, builds the Feature-List-ID and Feature-List AVPs to be included within the context of the Supported Features group.

```

diameter-manipulation
name diamManip2
description
diameter-manip-rule
name rule1
avp-code 266
descr-avp-code
avp-type unsignedint32
action none
msg-type any
msg-cmd-code
comparison-type case-sensitive
match-value 10
new-value
diameter-manip-rule
name rule2
avp-code 629
descr-avp-code
avp-type unsignedint32
action none
msg-type any
msg-cmd-code
comparison-type case-sensitive
match-value 11
new-value
diameter-manip-rule
name rule3
avp-code 630
descr-avp-code
avp-type unsignedint32
action none
msg-type any
msg-cmd-code
comparison-type case-sensitive
match-value 124
new-value

```

### COPS-based External Policy Servers

---

The Common Open Policy Service (COPS) [RFC 2748] is a protocol supported by the Oracle Communications Session Border Controller to perform and implement Call Admission Control (CAC) based on the policies hosted in an external policy server. While the Oracle Communications Session Border Controller already supports internal CAC policies, they are not as flexible as a Resource and Admission Control Function / Policy Decision Function (RACF/PDF), the generic resource and admission control functional architecture conceived by the ITU-T and the IETF.

The Oracle Communications Session Border Controller COPS model includes a Policy server, functionally called the policy decision point (PDP), and the edge router, functionally called the policy enforcement point (PEP), the Oracle Communications Session Border Controller itself. The PDP and the PEP communicate with each other via the COPS protocol.

One of three licenses is required to use External Policy Server services: External Bandwidth Management-for RACF support.; External CLF Mgmt-for CLF support; External Policy Services-for support of both RACF and CLF.

#### COPS Connection

The COPS session is established over a persistent TCP connection between the PDP and PEP. A COPS Client-Open (OPN) message is sent from the Oracle Communications Session Border Controller to the RACF, which responds with a COPS Client-Accept (CAT) message. A COPS Client-Close (CC) message is sent to either side to gracefully close the persistent connection. This COPS connection is expected to never close, unless an error occurs.

#### COPS Failures

Connection failures are discovered through a keep alive mechanism. Keep alive (KA) messages are periodically sent by the Oracle Communications Session Border Controller to the RACF regardless if any other COPS messages have been exchanged. When a KA message is not received, a connection failure is flagged. If the COPS connection fails, the Oracle Communications Session Border Controller will continually try to re-establish the connection to the PDP. Previously established calls will continue unaffected, but the Oracle Communications Session Border Controller will deny new calls from being established until the COPS connection is restored.

##### Failure Detection

A COPS connection failure is triggered by one of the three following events:

1. COPS KA timeout. The Oracle Communications Session Border Controller flags a COPS KA timeout when it does not receive a response for the KA it sent to the PDP. The PDP flags a COPS KA timeout when it does not receive the KA message within its requested timer time from the Oracle Communications Session Border Controller. At a minimum, when the COPS KA message times out, the TCP socket is closed.
2. Explicit COPS CC. The Oracle Communications Session Border Controller closes a COPS connection if it receives a COPS CC message from the PDP. The PDP closes a COPS connection if it receives a CC message from the Oracle Communications Session Border Controller. After the COPS layer connection is closed, then the TCP socket is closed too.
3. TCP socket termination. If either side receives a TCP FIN or RST, the TCP socket closes as expected. The COPS layer then detects that the socket has been closed before sending any further messages, and thus the COPS connection is closed.

##### Failure Recovery

The Oracle Communications Session Border Controller assumes that the PDP has a mechanism that re-uses the same logical IP Address, restarts itself in a timely manner, or fails over to another PDP. Therefore, no backup PDP IP address is configured on the Oracle Communications Session Border Controller.

The Oracle Communications Session Border Controller will try to re-open the COPS connection to recover from a connection failure. The PDP is never the device to initiate a connection. The Oracle Communications Session Border Controller increases its retry interval after successive reconnect failures. Once the retry interval has grown to every five minutes, the Oracle Communications Session Border Controller continues to retry to open the COPS connection at the five minute interval.



## COPS PS Connection Down

You can configure whether or not you want the Oracle Communications Session Border Controller to reject or allow new calls to be established despite the failure of a policy server (PS) connection.

You enable this feature in the external policy server configuration using a new parameter. When you enable the feature, the Oracle Communications Session Border Controller allows new SIP calls to be established even though the connection to the PS has failed. In this case, the PS will not respond and will not be aware of the established sessions. When you disable this feature, the Oracle Communications Session Border Controller behaves as it did in prior releases by responding to a connection failure with a 503 Service Unavailable.

## HA Support

The Oracle Communications Session Border Controller's high availability (HA) capabilities have been extended to support COPS. When one Oracle Communications Session Border Controller in an HA configuration goes down, the MAC addresses are reassigned to a healthy Oracle Communications Session Border Controller. IP addresses "follow" the MAC addresses to provide a seamless switchover between HA nodes.

After an HA failover, the COPS connection on the primary Oracle Communications Session Border Controller is either gracefully torn down, or times out depending on behavior of the PDP. The backup Oracle Communications Session Border Controller attempts to create a new COPS connection with the PDP. The OPN message uses the same PEPID and Client Type as in the previous pre-failover session.

## Application Types

The Oracle Communications Session Border Controller supports the following COPS-based methods for interfacing with a RACF:

- PKT-MM3 (PacketCable™ Specification Multimedia Specification PKT-SP-MM-I03-051221) (client type: 0x800A)
- Acme Packet proprietary (client type: 0x7926)

The Oracle Communications Session Border Controller supports the following COPS-based methods for interfacing with a CLF:

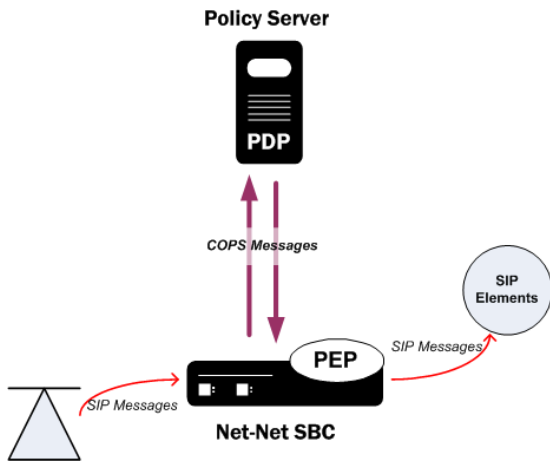
- Oracle proprietary (client type: 0x7929)

## COPS: RACF

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CAC is performed according to the following typical scenario. When the Oracle Communications Session Border Controller receives a SIP INVITE, it sends a COPS request (REQ) message to the PDP. The REQ message includes the call ID, the SIP client's IP address, the Oracle Communications Session Border Controller's IP address and port number of the ingress interface for the call, and SDP based bandwidth requirements. The PDP responds with a COPS Decision (DEC) message with either the Install or Remove command. An Install command directs the Oracle Communications Session Border Controller to forward the INVITE to the next SIP device. A Remove command directs the Oracle Communications Session Border Controller send a SIP 503 Service Unavailable message sent back to the UA and reject the call.

## External Policy Servers



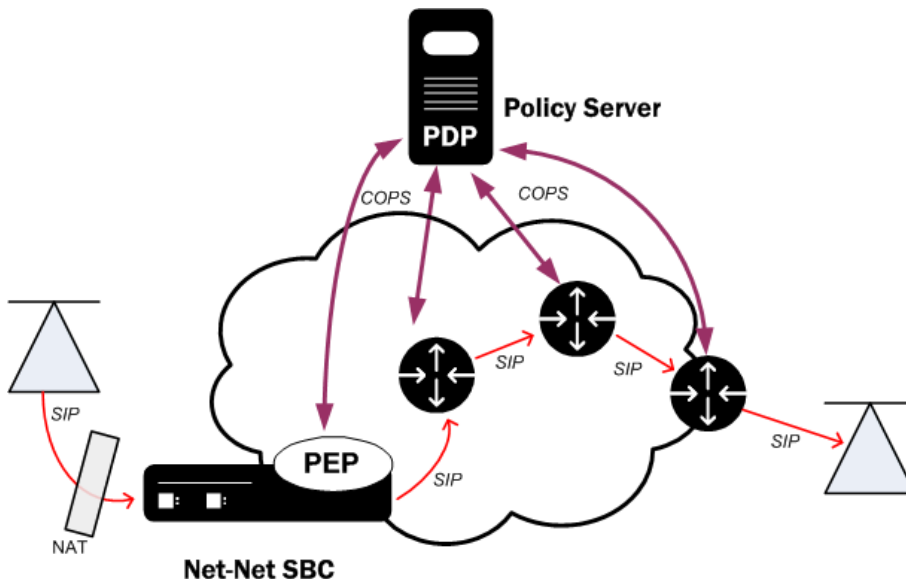
The Oracle Communications Session Border Controller can be configured so that both sides of a call, based on realm, are subject to COPS bandwidth enforcement. Each flow is treated as a unique call/event, because from a media and signaling perspective, they are distinct. As the Oracle Communications Session Border Controller functions as one side of a call, its IP address is inserted into the REQ message regardless of whether it is the calling or called party. This allows for the COPS install or remove decision to be made before the Oracle Communications Session Border Controller receives the 200 OK response, and before ringing the far-end phone. Only one external policy server can be used within a single realm.

When a call ends, either with the expected SIP BYE or CANCEL methods, or due to other error conditions, the Oracle Communications Session Border Controller will delete the reservation on the PDP by sending a COPS delete request state (DRQ) message to the PDP. All ended calls must be deleted from the PDP in order to accurately track used and available bandwidth.

## Implementation Features

As the Oracle Communications Session Border Controller proxies and forwards calls, caller media information is known before the INVITE reaches the callee. The PEP can request a specific amount of bandwidth for a call, and the PDP can reserve this amount of bandwidth for a call before the called phone rings. A call's required bandwidth can also be reserved by network devices along the path from the caller to callee if the QoS admission criteria is pushed to PEPs such as routers, along this path to the callee.

The RACF can apply its hosted policies for calls originating at SIP UAs located behind NATs. This is a standard part of the Oracle Communications Session Border Controller's ability to provide seamless HNT.



---

## Bandwidth Negotiation

Because the decision whether to admit or reject a call is made before the INVITE is forwarded to the called party, some information is not available to the PDP at the initial request. The final IP Address, UDP port number, that transport the RTP flow, and the codec used are not known by the Oracle Communications Session Border Controller until the called party responds with its own SDP information (either in the 180 or 200 response).

The Oracle Communications Session Border Controller sends a request to the PDP requesting as much bandwidth as the codec with the highest bandwidth in the SDP message requires. If the call is admitted, and when the called party returns finalized SDP information, the Session Director will modify the original reservation with the chosen codec's bandwidth requirements. This ensures the PDP has current and accurate information with which to make policy decisions.

---

## COPS pkt-mm-3 Policy Control

You can configure the Oracle Communications Session Border Controller's COPS pkt-mm-3 interface to maintain a persistent TCP connection to the external policy server, despite there being no responses to requests for bandwidth. This permits calls to traverse the Oracle Communications Session Border Controller even though the external policy server either fails to respond or rejects the session. Without this functionality configured, the Oracle Communications Session Border Controller waits for the external policy server's authorization decision or the request to the external policy server times out, resulting in COPS pkt-mm-3 interface error responses and time-outs.

While these type of error responses and time-outs might be handled in some networks, for others they cause unacceptable call failures and call denials. To avoid this situation, you can select the granularity at which your network admits calls through a best-effort pipe and guarantee the Oracle Communications Session Border Controller's TCP connection to the external policy server remains uncompromised. The external policy server offers three parameters allowing you to enable this capability: **permit-on-reject**, **disconnect-on-timeout**, and **gate-spec-mask**.

## Relationship to the TCP Connection

The TCP connection between the Oracle Communications Session Border Controller and the external policy server plays an important role because second-tier networks can experience issues between the policy server and the cable modem termination system (CMTS). Since one policy server can be connected to multiple others, one CMTS's failure can cascade and cause multiple failures for other CMTSs and policy servers.

Enables the **disconnect-on-timeout** parameter allows the Oracle Communications Session Border Controller to maintain its TCP connection to the external policy server regardless of upstream issues between PSs and CMTSs. When you disable this setting, the Oracle Communications Session Border Controller can send Gate-Set and Gate-Delete messages to in response the PS's timeouts and guard against impact to the TCP connection between the Oracle Communications Session Border Controller and the PS.

## COPS Gate-Set Timeout

Without this capability enabled, the Oracle Communications Session Border Controller views the situation as a rejection when its COPS Decision Message times out at the PS. As such, the Oracle Communications Session Border Controller denies the session or attempts to revert to the previously requested bandwidth from either a **Gate-Set** add or **Gate-Delete** modify message. The **permit-on-reject** parameter, however, enables the system to forward the session on at a best-effort.

Enabling both the **permit-on-reject** and the **disconnect-on-timeout** parameters prompts yet another Oracle Communications Session Border Controller behavior. Consider the situation where a session is in the processing of being established on the Oracle Communications Session Border Controller, and it carries COPS Gate-Set messages to the PS. Typically in this scenario, one voice call has two COPS decision messages: one for the upstream audio, and one for the downstream audio. If any of the Gate-Set message do not receive a response, the Oracle Communications Session Border Controller forwards on the session under as a best effort. And if a non-final Gate-Set message in the session does not receive a response, any remaining Gate-Set messages for the session are not forwarded to the PS; the whole session is forwarded on as a best effort. In other words, if the first Oracle

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Communications Session Border Controller Gate-Set message out of two times out, the second for the session is not sent.

At every opportunity, the Oracle Communications Session Border Controller tried to elevate the session to a QoS session from its best-effort status. When you enable the external policy server configuration's **reserve-bandwidth** parameter, the Oracle Communications Session Border Controller checks twice for bandwidth: once upon INVITE, and a second time upon the 200 OK. Alternatively, when Oracle Communications Session Border Controller receives a ReINVITE, it checks for bandwidth then, too. At the time of these second checks, a call's best-effort status can improve to that of a QoS session.

### When a Gate-Set Times Out

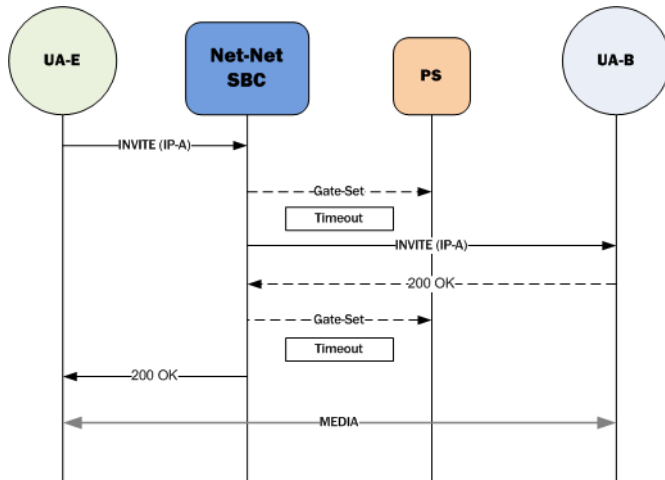
If the Oracle Communications Session Border Controller labels a session best-effort, it does so because the Gate-Set message it has sent to the PS have timed out. The Oracle Communications Session Border Controller does not send a Gate-Delete in this situation because it has not received a valid Gate-Id for that session from the PS.

It is also possible that when a Gate-Set message times out, the CMTS and PS might have allocated the session after the Oracle Communications Session Border Controller's timeout. One of two errors scenarios result from this occurrence, neither of which is of significance. However, you should be aware of them:

- If a Gate-Set message times out on the INVITE side of the session, the subsequent Gate-Set message triggered by the 200 OK looks like a request to allocate the same resources. If the CMTS and PS are able to successfully install the flow outside the Oracle Communications Session Border Controller's eight-second request time-out, issues can ensue.
- Assuming the Gate-Set message time out but are nonetheless installed after the Oracle Communications Session Border Controller's request time-out, the Oracle Communications Session Border Controller does not send Gate-Delete messages for the flows because it believes they are unallocated on the PS. Because the Oracle Communications Session Border Controller has no Gate-Id data, it remains unaware the sessions are installed on the PS and CMTS. The PS and CMTS might then be left with stranded gates.

This diagram shows a call flow where the Gate-Set times out. It assumes an access-to-core configuration, where the external bandwidth management configuration shows:

- **disconn-on-timeout**=disabled
- **permit-conn-down**=enabled
- 



### COPS Decision Gate-Set Message Rejected

The COPS pkt-mm-3 interface reply as it does when this feature is not enabled (reply with an error to the signaling application) when:


- The PS responds to the Gate-Set message with an error, either denying or rejecting the flow

- The **permit-on-reject** parameter is disabled (its default setting)

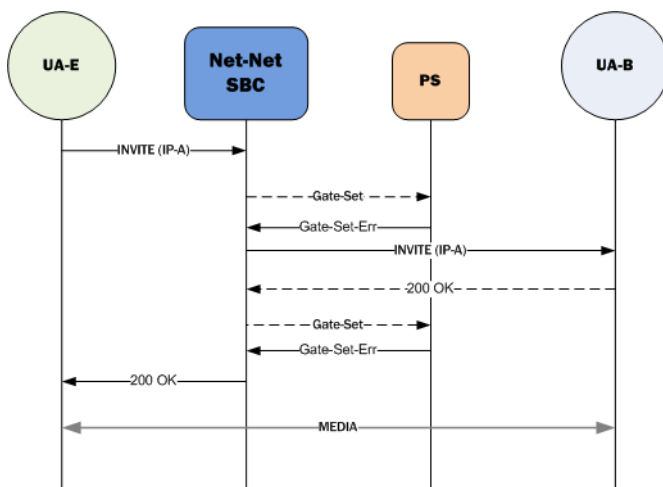
This again results in the situation where a session is in the processing of being established on the Oracle Communications Session Border Controller, and it carries COPS Gate-Set messages to the PS. Typically in this scenario, one voice call has two COPS decision messages: one for the upstream audio, and one for the downstream audio. If any of the Gate-Set message do not receive a response, the Oracle Communications Session Border Controller forwards on the session under as a best effort when the **permit-on-reject** parameter is enabled.

If a non-final Gate-Set message for a session elicits an error or deny response, the remaining Gate-Set message(s) for that session are not sent to the PS. The Oracle Communications Session Border Controller forwards the entire session as a best effort. This means that if the Oracle Communications Session Border Controller sends two Gate-Set messages and the first meets an error or deny response, then the second one is not sent. Still, the Oracle Communications Session Border Controller attempts to elevate the session’s status from best-effort to QoS given the multiple requests for bandwidth it might issue.

Note

 **Note:** The Oracle Communications Session Border Controller does not send Gate-Delete message for session that have been deemed best-effort. This is because the Gate-Set messages the Oracle Communications Session Border Controller sent to the PS have timed out, and the Oracle Communications Session Border Controller does not have the PS’s Gate-Id data for the session.

This diagram shows a call flow where Gate-Set messages are denied. It assumes an access-to-core configuration, where the external bandwidth management configuration shows a **permit-on-reject set to enabled**.



## About the Gate-Spec Mask

Vendors of policy servers and those who incorporate PSs into their networks must allow the CMTS to overwrite IP packets with ToS bytes because the value of the Oracle Communications Session Border Controller’s DSCP/TOS mask is 0xFF. The CMTS uses the following equation to determine whether or not a ToS byte should be overwritten:

$$\text{new-ip-tos} = ((\text{orig-ip-tos} \text{ AND } \text{tos-and-mask}) \text{ OR } (\text{tos-or-mask}))$$

Using the **gate-spec-mask** parameter, you can configure the Oracle Communications Session Border Controller to use a mask comprised entirely of zeros (0s). Doing so allows the PS to reset the byte before calculations using the media policy configuration’s DCSP field can take place.

## COPS Debugging

A new argument has been added to the show command for viewing COPS and CAC statistics. From the user prompt, type **show ext-band-mgr**.

```
ACMEPACKET# show ext-band-mgr
10:11:38-194
```

## External Policy Servers

EBM Status	-- Period --			----- Lifetime -----		
	Active	High	Total	Total	PerMax	High
Client Trans	0	0	0	0	0	0
Server Trans	0	0	0	0	0	0
Sockets	1	1	1	1	1	1
Connections	0	0	0	0	0	0
		---- Lifetime ----				
	Recent	Total	PerMax			
Reserve	0	0	0			
Modify	0	0	0			
Commit	0	0	0			
Remove	0	0	0			
EBM Requests	0	0	0			
EBM Installs	0	0	0			
EBM Errors	0	0	0			
EBM Rejects	0	0	0			
EBM Expires	0	0	0			
EBMD Errors	0	0	0			

You can also refer to the log.ebmd log file located in the /ramdrv/logs/ directory on the Oracle Communications Session Border Controller . This file must be retrieved via FTP or SFTP.

## COPS-based RACF Configuration

In the following configuration examples, we assume that your baseline configuration passes SIP traffic. In this example, you will configure additions to the realm configuration and the new external bandwidth manager configuration. You must also configure media profiles to accept bandwidth policing parameters.

### Realm Configuration

To configure the realm configuration for COPS support in a CAC scenario:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # realm-config  
ACMEPACKET (realm-config) #
```

4. Type **select** and the number of the pre-configured sip interface you want to configure.

```
ACMEPACKET (realm-config) # select 1  
ACMEPACKET (realm-config) #
```

5. **mm-in-realm**—Set this parameter to **enabled** so that calls from devices in the same realm have their media flow through the Oracle Communications Session Border Controller to be subject to COPS CAC. The default value is **disabled**. The valid values are:
  - enabled | disabled
6. **mm-in-network**—Set this parameter to **enabled** so that the Oracle Communications Session Border Controller will steer all media traveling between two endpoints located in different realms, but within the same network. If this field is set to **disabled**, then each endpoint will send its media directly to the other endpoint located in a different realm, but within the same network. The default value is **enabled**. The valid values are:
  - enabled | disabled
7. **ext-bw-manager**—Enter the name of the external bandwidth manager configuration instance to be used for external CAC for this Realm.

8. Save your work using the CLI **done** command.

## External Bandwidth Manager Configuration

To configure the external bandwidth manager:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET(configure)# media-manager
```

3. Type **ext-policy-server** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(media-manager)# ext-policy-server
ACMEPACKET(ext-policy-server)#
```

4. **name**—Enter the name for this external bandwidth manager instance. This parameter is used to identify the PDP used for that will be used in each Realm configuration.
5. **state**—Set state to **enabled** to enable this external policy server.
6. **operation-type**—Enter **bandwidth-mgmt** for this external policy server configuration element to perform bandwidth management and communicate with a RACF. This sets the COPS client type to **0x7926**. If another vendor's Policy Server is supported, it will be a different protocol value.
7. **protocol**—Enter **COPS** to support COPS-based CAC. The **A-COPS** protocol implicitly sets the Oracle Communications Session Border Controller to use **0x4AC0** as the COPS client type. The default value is **C-SOAP**.
8. **address**—Enter the IP Address or FQDN of the external COPS-based policy server.
9. **port**—Enter the port number the COPS connection connects to on the PDP. The default value is **80**. (The standard port for COPS is 3288.) The valid range is:
  - Minimum—0
  - Maximum—65535
10. **realm**—Enter the name of the Realm in which this Oracle Communications Session Border Controller defines the external policy server. This is NOT necessarily the Realm that the Oracle Communications Session Border Controller performs admission requests for.
11. **application-mode**—Enter the type of interface you want to use. Your choices are: **Rq, Rx, Gq, e2**, and **none**. Set this to **none** or **pkt-mm3**.
12. **application-id**—Enter a numeric application ID that describes the interface used to communicate with the RACF. The default value is zero (0). The valid range is:
  - Minimum—0
  - Maximum—999999999
13. **permit-conn-down**—Enter enabled for the Oracle Communications Session Border Controller to establish new SIP sessions despite PS connection failure. The default value is **disabled**. The valid values are:
  - enabled | disabled
14. **reserve-incomplete**—Set this parameter to **enabled** when communicating with a PDP via COPS. The parameter allows the Oracle Communications Session Border Controller to make admission requests before learning all the details of the flows and devices (e.g., not knowing the final UDP port numbers for the RTP media streams until after the RTP has begun). The default value is **enabled**. The valid values are:
  - enabled | disabled
15. **permit-on-reject**—Set this parameter to **enabled** if you want the Oracle Communications Session Border Controller to forward the session on a best-effort basis. Leave this parameter set to **disabled**, its default, if you want the system deny the session or attempts to revert to the previously requested bandwidth.

## External Policy Servers

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- 16. disconnect-on-timeout**—Leave this parameter set to enabled, its default, so the Oracle Communications Session Border Controller can maintain its TCP connection to the external policy server regardless of upstream issues between PSs and CMTSS. When you disable this setting, the Oracle Communications Session Border Controller can send Gate-Set and Gate-Delete messages to in response the PS's timeouts and guard against impact to the TCP connection between the Oracle Communications Session Border Controller and the PS.
- 17. gate-spec-mask**—This parameter sets the value to use for Gate-Spec mask for the COPS pkt-mm-3 interface. The default is `.255`. The minimum value is `0`, and the maximum is `255`.
- 18.** Save your work using the ACLI **done** command.

## Media Profile Configuration

To configure the media profile configuration for COPS support in a CAC scenario:

Values for the following parameters can be found in the PacketCable™ Audio/Video Codecs Specification PKT-SP-CODEC-I06-050812 document.

- In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

- Type **session-router** and press Enter to access the session router path.

```
ACMEPACKET (configure) # session-router
```

- Type **media-profile** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # media-profile  
ACMEPACKET (media-profile) #
```

- Type **select** and the number of the pre-configured media profile you want to configure.

```
ACMEPACKET (media-profile) # select 1  
ACMEPACKET (media-profile) #
```

- peak-rate-limit**—Enter the r, P value:

- r**—bucket rate
- p**—peak rate

- max-burst-size**—Enter the b, m, M value:

- b**—Token bucket size
- m**—Minimum policed unit
- M**—Maximum datagram size

- Save your work using the ACLI **done** command.

## COPS: CLF

---

A Connectivity Location Function (CLF) maintains mappings between endpoints with dynamically assigned IP addresses and their physical location. The Oracle Communications Session Border Controller, acting as a P-CSCF, is the intermediary device between a registering endpoint and a CLF. The CLF thus validates and tags a registering endpoint, and the Oracle Communications Session Border Controller applies the CLF's actions. The Oracle Communications Session Border Controller and the CLF maintain a connection with each other using the COPS protocol.

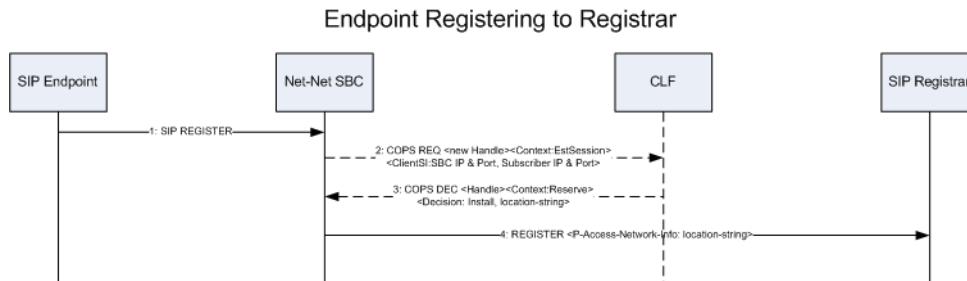
## CLF Behavior

The Oracle Communications Session Border Controller and a CLF only interact when an endpoint registers or re-registers. The Oracle Communications Session Border Controller, acting as the P-CSCF, is the first SIP device that the REGISTER message reaches. Upon receiving the REGISTER message(1), the Oracle Communications Session



Border Controller queries the CLF using the COPS protocol. The endpoint's (public) IP address and port, and the Oracle Communications Session Border Controller's IP information are sent to the CLF in a COPS REQ message(2).

The CLF responds to the Oracle Communications Session Border Controller with an Approve or Reject COPS DEC message(3). If the request is approved, then the CLF also sends a location-string value to be inserted in one of the SIP headers. The Oracle Communications Session Border Controller inserts a P-Access-Network-Info header containing the location-string into the incoming REGISTER message and forwards this message(4) to the SIP registrar/I/S-CSCF.



The Oracle Communications Session Border Controller will insert this P-Access-Network-Info header into all subsequent SIP messages from this endpoint as they are forwarded into the core network. The P-Access-Network-Info header is inserted into all SIP requests and responses except for ACK and CANCEL messages. For all boundaries where SIP messages pass from trusted to untrusted SIP interfaces or session agents, the Oracle Communications Session Border Controller will strip out the P-Access-Network-Info header as expected.

If the CLF responds with a Reject DEC message, the Oracle Communications Session Border Controller rejects the registration, and sends a 503 - Service Unavailable message back to the registering endpoint. In this way, the CLF can be used for admission control.

The Oracle Communications Session Border Controller communicates with the CLF solely for retrieving location information from the CLF, and not for notifying the CLF about an endpoint's registration state or activity. When an endpoint's registration ends, either through a normal expiration, getting rejected by the registrar, or through specific de-registering or error conditions, the Oracle Communications Session Border Controller deletes the locally cached registration location string. The Oracle Communications Session Border Controller does not update the CLF about any registrations that have been deleted.

### P-Access-Network-Info Header Handling

The P-Access-Network-Info header is created and populated according to the following rules:

- If the CLF returns an Accept DEC message and a location string, the Oracle Communications Session Border Controller inserts the location string into a P-Access-Network-Info header in the outgoing REGISTER message.
- If the CLF returns an Accept DEC message without a location string, the Oracle Communications Session Border Controller inserts the configured default string into a P-Access-Network-Info header in the outgoing REGISTER message.
- If the CLF returns an Accept DEC message without a location string and no location string is configured on Oracle Communications Session Border Controller, the outgoing REGISTER message is forwarded out of the Oracle Communications Session Border Controller, but no P-Access-Network-Info header is created for the REGISTER message.

### CLF Re-registration

The Oracle Communications Session Border Controller will send a new REQ message to the CLF to request a new location string if any of the following events occur:

- The endpoint's contact address changes.
- The SIP Register message's Call-ID header changes.
- The endpoint's public IP Address or UDP port changes.
- The endpoint connects to a different SIP interface, port, or realm on the Oracle Communications Session Border Controller than it did in the initial REGISTER message.

## External Policy Servers

- The registration expires in the Oracle Communications Session Border Controller 's registration cache.

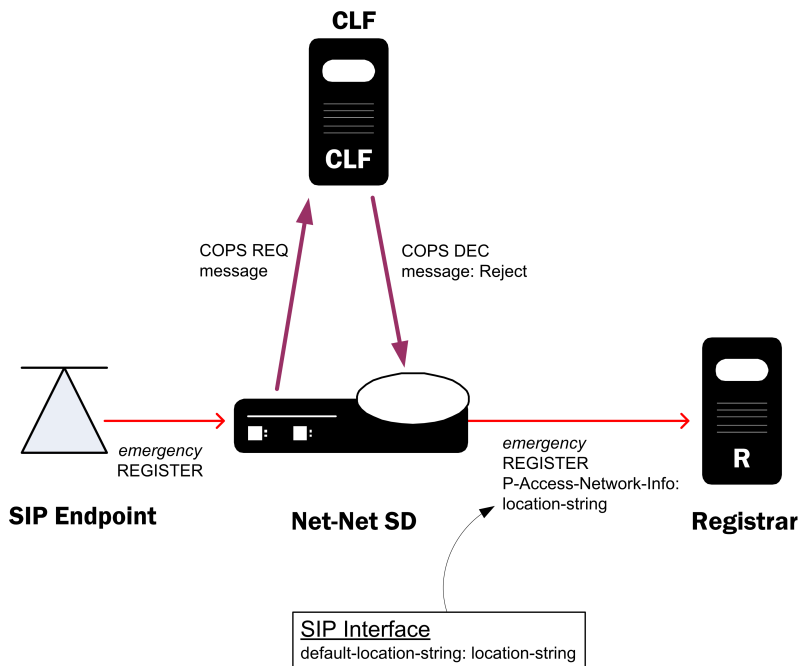
## CLF Failures

If a COPS connection fails, the Oracle Communications Session Border Controller will continually try to re-establish the connection. Endpoints that are already registered will stay registered unless they timeout or if the registrar rejects their refreshes. When the COPS connection has not been established, and an endpoint registers on a SIP interface that is configured to use CLF, the Oracle Communications Session Border Controller forwards new REGISTER messages to the registrar using the default location string.

## CLF Emergency Call Handling

The Oracle Communications Session Border Controller allows emergency calls into the network even if the endpoint that places the emergency call is not registered. In the expected fashion, the Oracle Communications Session Border Controller will query the CLF first for an incoming emergency call sourced from an unregistered endpoint. If the CLF response is successful, then the Oracle Communications Session Border Controller will insert the string returned from the CLF into a P-Access-Network-Info header, and insert this header into the emergency call's REGISTER message. If no location string is returned with a successful CLF response, the default location string is inserted into P-Access-Network-Info header.

If the CLF's response is to reject the emergency call, the Oracle Communications Session Border Controller will insert the configured default location string into the P-Access-Network-Info header and forward the emergency call's REGISTER message toward the registrar. For emergency calls where the endpoint has already successfully registered, the call will be routed into the network using the expected methods for emergency call routing.



If the COPS connection to the CLF is down, emergency calls from un-registered endpoints are still allowed into the network using the default string inserted into the emergency messages.

## HA Functionality

The location strings generated by the CLF are replicated on the standby Oracle Communications Session Border Controller in an HA pair. This is required so that a Oracle Communications Session Border Controller in an HA pair can instantly continue processing calls using the previously learned CLF information.

## CLF Debugging

A new argument has been added to the show command for viewing CLF statistics. From the user prompt, type **show** <space> **ext-clf-svr** <return>.

```
ACMEPACKET# show ext-clf-svr
14:17:14-114
EBM Status
Active      -- Period -- ----- Lifetime -----
Client Trans 0         0         0         0         0         0
Server Trans 0         0         0         0         0         0
Sockets      0         0         0         0         0         0
Connections  0         0         0         0         0         0

----- Lifetime -----
Recent      Total      PerMax
CLF Requests 0          0          0
CLF Admits   0          0          0
CLF Errors   0          0          0
CLF Rejects  0          0          0
CLF Expires  0          0          0
CLFD Errors  0          0          0
```

You can also refer to the log.ebmd log file located in the /ramdrv/logs/ directory on the Oracle Communications Session Border Controller . This file must be retrieved via FTP or SFTP.

## COPS-based CLF Configuration

In the following configuration examples, we assume that your baseline configuration passes SIP traffic, with the Oracle Communications Session Border Controller in the role of an Access Oracle Communications Session Border Controller . In this example, you will configure additions to the ream configuration and the new external policy server configuration.

### SIP Interface Configuration

To configure the SIP interface configuration for CLF support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-related configurations.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # sip-interface
ACMEPACKET (sip-interface) #
```

4. Type **select** and the number of the pre-configured sip interface you want to configure for CLF. This should be the ingress SIP interface for

```
ACMEPACKET (sip-interface) # select 1
ACMEPACKET (sip-interface) #
```

5. **ext-policy-svr**—Set this parameter to the same name as the External Policy Server configured that you configured for the CLF server.
6. **default-location-string**—Set this parameter to the default location string you want inserted into a P-Access-Network-Info header for when the CLF server does not return a unique location string.
7. Save your work using the ACLI **done** command.

To configure the external policy server for use with a CLF:

8. In Superuser mode, type **configure terminal** and press Enter.

## External Policy Servers

```
ACMEPACKET# configure terminal
```

9. Type **media-manager** and press Enter to access the media-related configurations.

```
ACMEPACKET (configure) # media-manager
```

10. Type **ext-policy-server** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (media-manager) # ext-policy-server  
ACMEPACKET (ext-policy-server) #
```

11. **name**—Set this parameter to an applicable name for this CLF instance of the external policy server. The value of this parameter will be entered in the SIP interface configuration element to reference this CLF.
12. **state**—Set this parameter to **enabled** to enable this CLF. The default value is **enabled**. The valid values are:
- enabled | disabled
13. **operation-type**—Set this parameter to **admission-control** for the Oracle Communications Session Border Controller to communicate with a CLF. The default value is **disabled**.
14. **protocol**—Set this parameter to **COPS** to connect with a CLF via the COPS protocol. The default value is **C-SOAP**. The valid values are:
- **COPS**—Standard COPS implementation. COPS client type is 0x7929 for CLF, and 0x7926 for PDP/RACF usage as defined in the operation-type parameter.
  - **A-COPS**—Vendor specific protocol. COPS client type is 0x4AC0 for admission-control operation-type.
15. **address**—Set this parameter to the IP address or FQDN of the CLF.
16. **port**—Set this parameter to the port which the CLF uses for COPS transactions. The standard port for COPS is **3288**. The default value is **80**. The valid range is:
- Minimum—0
  - Maximum—65535
17. **realm**—Set this parameter to the realm in which the CLF exists.
18. **num-connections**—Set this parameter to the number of connections the Oracle Communications Session Border Controller will create with the CLF. The default value is **1**. The valid range is:
- Minimum—0
  - Maximum—65535
19. **reserve-incomplete**—Set this parameter to **enabled** if you want the Oracle Communications Session Border Controller to send a COPS REQ message to the CLF that does not include the endpoint's true port number. A value of 0 will be used for the port number. The default value is **enabled**. The valid values are:
- enabled | disabled
20. Save your work using the ACLI **done** command.

## Application Type / Interface Matrix Reference

### Bandwidth Management Applications

Standards Reference Point	SBC as	External Policy Server as	Protocol	Interface
TISPAN	P-CSCF	A-RACF	Diameter	Rq
3GPP (R7)	P-CSCF	PCRF	Diameter	Rx
3GPP (R6)	P-CSCF (AF)	PDF	Diameter	Gq

## External Policy Servers

Standards Reference Point	SBC as	External Policy Server as	Protocol	Interface
Packet Cable	P-CSCF (AF)	PDF	COPS	PKT-MM3
		RACF	ACME COPS	

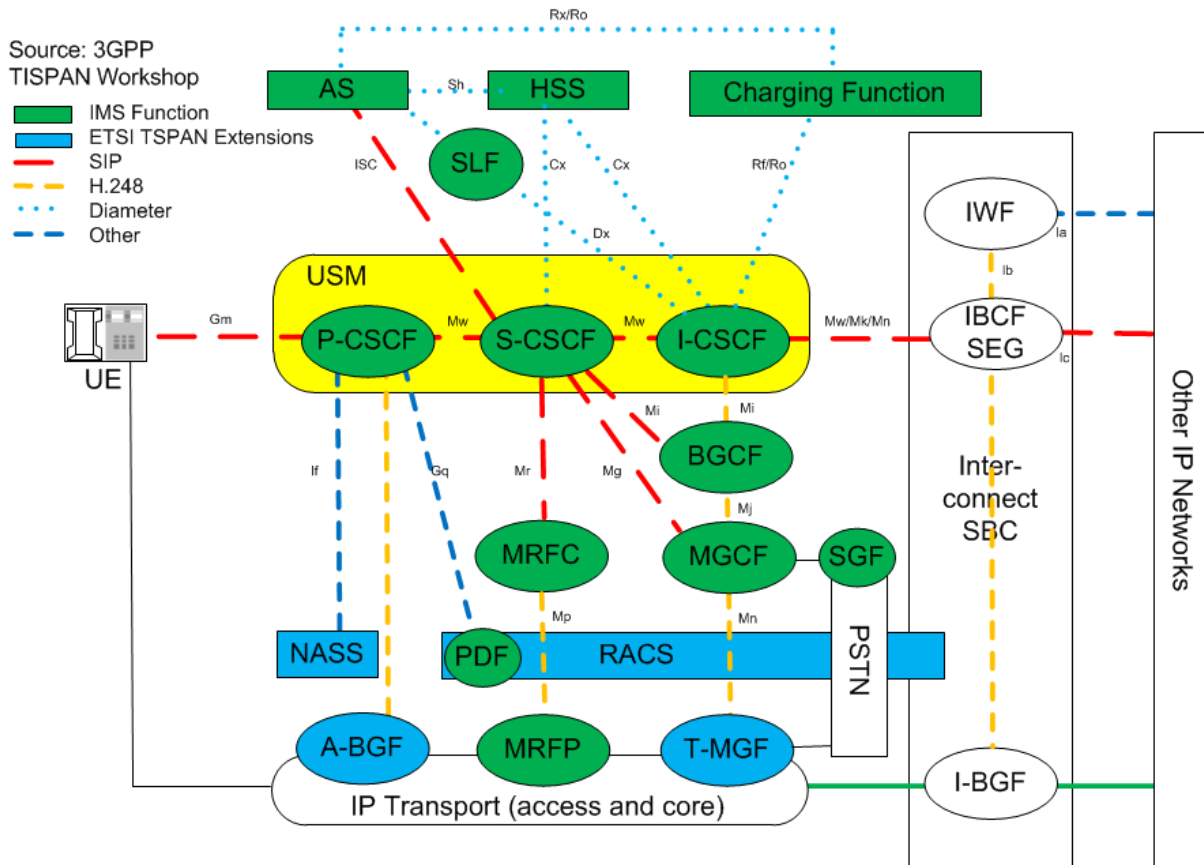
## Emergency Location Services

Standards Reference Point	SBC as	External Policy Server as	Protocol	Interface
	P-CSCF	CLF	Diameter	e2
		CLF	ACME COPS	



# IMS Support

The ETSI TISPAN NGN defines several subsystems that make up the NGN architecture. The model for the target NGN architecture is depicted below. The Oracle Communications Session Border Controller is an integrated session control, policy enforcement and media management solution that incorporates functional components of the IP multimedia subsystem (IMS), the Resource and Admission Control Subsystem (RACS) and functions necessary for connecting with other IP networks/domains. The functions of the Oracle Communications Session Border Controller within the NGN architecture are divided into the interconnect border functions and the access border functions. The diagram below depicts the mapping of these functions across IMS architecture.



### Oracle Communications Session Border Controller Access Border Functions

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- Proxy CSCF (P-CSCF)
- Access/Core Border Gateway Function (A/C-BGF)
- RACF AF and SPDF functions

### Oracle Communications Session Border Controller Interconnect Border Functions

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- Interconnect Border Control Function (I-BCF)
- Interworking Function (IWF)
- Interconnect Border Gateway Function (I-BGF)

### IMS Access Border Functions

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The Oracle Communications Session Border Controller is deployed as the access point between the core IMS network and UEs to deliver the functions defined in the TISpan architecture as the P-CSCF, and A-BGF. These two functions can not be separated. The Oracle Communications Session Border Controller performs the following functions as the Access Oracle Communications Session Border Controller:

#### P-CSCF Functions

The Oracle Communications Session Border Controller performs the following functions in the role of P-CSCF:

- Forwards SIP REGISTER messages and maintains a cached mapping of the user info and the UE's Address of Record (AoR), including the far-end NAT address in the case of hosted NAT traversal (HNT).
- Forwards SIP messages to a S-CSCF based on service route discovery procedures.
- Performs local emergency session handling—Local routing policy is used by the Oracle Communications Session Border Controller to identify emergency sessions and provide unique routing (e.g. can route to a dedicated S-CSCF function for emergency session handling).
- Operates as a UA (B2BUA) for generating independent SIP transactions for security purposes and handling of abnormal conditions.
- Offers current session timers which are used to monitor for media faults and abandoned calls.
- Generation of CDRs—The Oracle Communications Session Border Controller generates real-time accounting records via RADIUS.
- Authorization of bearer resources and QoS management—With integrated BGF capabilities, the Oracle Communications Session Border Controller allocates bearer resources (NAPT flows) and applies QoS policies (including packet marking) based on local policies and/or policies acquired via interaction with the A-RACF (PDF).
- Interaction with the A-RACF (PDF) for session-based policy enforcement and admission control—The Oracle Communications Session Border Controller PDF interface options include COPS and SOAP/XML.
- Traffic Policing—Traffic is policed at the session and media/transport layer. At the signaling layer, the Oracle Communications Session Border Controller polices at a number of levels including:
  - Capacity—Total number of concurrent calls to/from each realm
  - Session set-up rate—Maximum rate of call attempts to/from each signaling element
  - Signaling message rate—Each endpoint's signaling message rate is monitored and policed
  - Signaling bandwidth—each endpoint's signaling bandwidth is policed individually



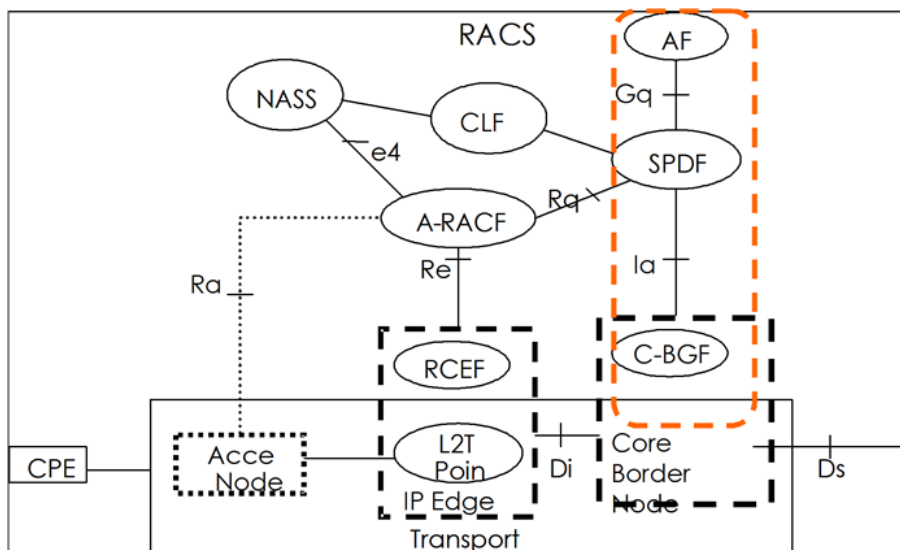
## A-BGF Functions

The Oracle Communications Session Border Controller performs the following IMS BGF functions:

- Opening and closing gates/packet filtering—The Oracle Communications Session Border Controller opens and closes gates (media pinholes) on a session-by-session basis. Packet filtering rules include full source and destination IP address and port number.
- Per-session DiffServ or ToS marking—Media flows destined for the IMS core network can be explicitly marked using ToS or DiffServ. Media packets can be marked by VPN, by codec (voice, video) or by E.164 phone number prefix.
- NAT-PT and topology hiding—The Oracle Communications Session Border Controller provides NAT for all media flows associated with a session on a per session-basis. Double NATing, NATing both source and destination sides, is utilized to fully hide topology in each direction for RTP and RTCP. Local IP addresses and port resources are dynamically allocated from steering pools provisioned on the Oracle Communications Session Border Controller.
- Hosted NAT traversal—The Oracle Communications Session Border Controller supports HNT function that allows media flow traversal through the CPE firewall/NAT without upgrading the CPE equipment. The system interacts with the endpoints to dynamically establish and maintain bindings in the CPE firewall/NAT that allow the signaled communications to pass through. The Oracle Communications Session Border Controller's registration management and media relay functions make CPE-based NATs transparent to the service delivery elements.
- Traffic Policing—Traffic is policed at the session and media/transport layer. At the signaling layer, the Oracle Communications Session Border Controller polices at a number of levels including:
  - Policing of Media (e.g. RTP & RTCP) traffic on a per-flow basis—CBR policing is applied to each flow based on negotiated offered and negotiated media codecs.

## Resource and Admission Control (RACS) Functions

The figure below illustrates the mapping of Oracle Communications Session Border Controller functions to the RACS functional model. In this model, the Oracle Communications Session Border Controller incorporates the Application Function (in the case of IMS this is the P-CSCF function), the SPDF (Service Policy Decision Function) and the Core Border Gateway function.



The Oracle Communications Session Border Controller, acting as the SPDF, interfaces with the PDF (A-RACF policy decision function) for resource authorization and admission control on a call-by-call basis. COPS is the supported PDF interface.

### IMS Interconnect Border Functions

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The Oracle Communications Session Border Controller is deployed at IP interconnect points between service providers to deliver the functions defined in the TISpan architecture as the I-BCF, IWF and I-BGF. The Oracle Communications Session Border Controller performs the following functions as the Interconnect border Oracle Communications Session Border Controller:

#### Interworking Function (IWF)

- Interworking SIP profiles and other protocols (e.g. H.323)

#### Interconnect Border Control Function (I-BCF)

- Interaction with I-BGF (including NAT and firewall functions)
- Insertion of the IWF when appropriate
- Topology hiding—screening of signalling information

#### Interconnect-Border Gateway Function (I-BGF)

- Gate opening/closing
- NAT and packet filtering
- Packet marking
- Resource allocation and bandwidth reservation
- Security and topology hiding
- Session admission control, resource and traffic management
- Upstream/downstream flow policing
- Quality monitoring and reporting
- Usage metering - CDR generation
- Lawful Intercept

### IMS Path and Service Route Header Support

---

The Oracle Communications Session Border Controller supports the Path header and the Service-Route header used in the registration phase of a SIP transaction. The Oracle Communications Session Border Controller will learn the route vectors from the SIP URIs contained in these headers in order to preload SIP headers with the correct route vectors in subsequent SIP message exchanges between the UA and the S-CSCF across the Oracle Communications Session Border Controller. This is how the Oracle Communications Session Border Controller supports RFC 3608 and RFC 3327.

#### Path Header

When a UE registers to an S-CSCF, the Oracle Communications Session Border Controller adds the Path header in the REGISTER message as it is proxied to the S-CSCF. The Path header includes the SIP URIs that form the route vector which describes how the UE reaches the Oracle Communications Session Border Controller, through a specific series of proxies. This route vector is saved in the Oracle Communications Session Border Controller's registration entry for the UE, routing all subsequent SIP messages from the S-CSCF to the UE. As the Path header is sent to the S-CSCF, the Oracle Communications Session Border Controller, as P-CSCF, inserts the SIP URI of itself as the top entry in the Path header.

The Path header only appears in SIP messages exchanged during the registration process.

If the REGISTER request already contains a Path header, the Oracle Communications Session Border Controller stores the contents of the Path header(s) for this endpoint for routing back to the endpoint in subsequent messages.

---

## Service Route Header

When a UE registers through the Oracle Communications Session Border Controller to the registrar, the registrar returns a Service-Route header in a 200 OK message in response to the REGISTER message to the UE. This header contains the route vector that directs traffic through a specific sequence of proxies used to reach the S-CSCF. The Service-Route header only appears during the SIP registration process.

The P-CSCF ( Oracle Communications Session Border Controller) will now store the URIs listed in the Service-Route header(s) in the registration entry of the UE for use in routing subsequent traffic to the S-CSCF. The Oracle Communications Session Border Controller inserts this sequence of proxies into and outgoing message's Route headers; this is called a pre-loaded route. This route is only applicable for the traffic flowing between the originating UE and the contacted S-CSCF.

When receiving subsequent requests from the UE, the Oracle Communications Session Border Controller looks at the UE's registration entry for a service route, and will insert the route vector as appropriate Route headers. If the service route is not found in the registration entry, the routing is performed in the usual fashion.

As an exception, you may wish for the Oracle Communications Session Border Controller to not use the Service-Route header to route subsequent Register requests.



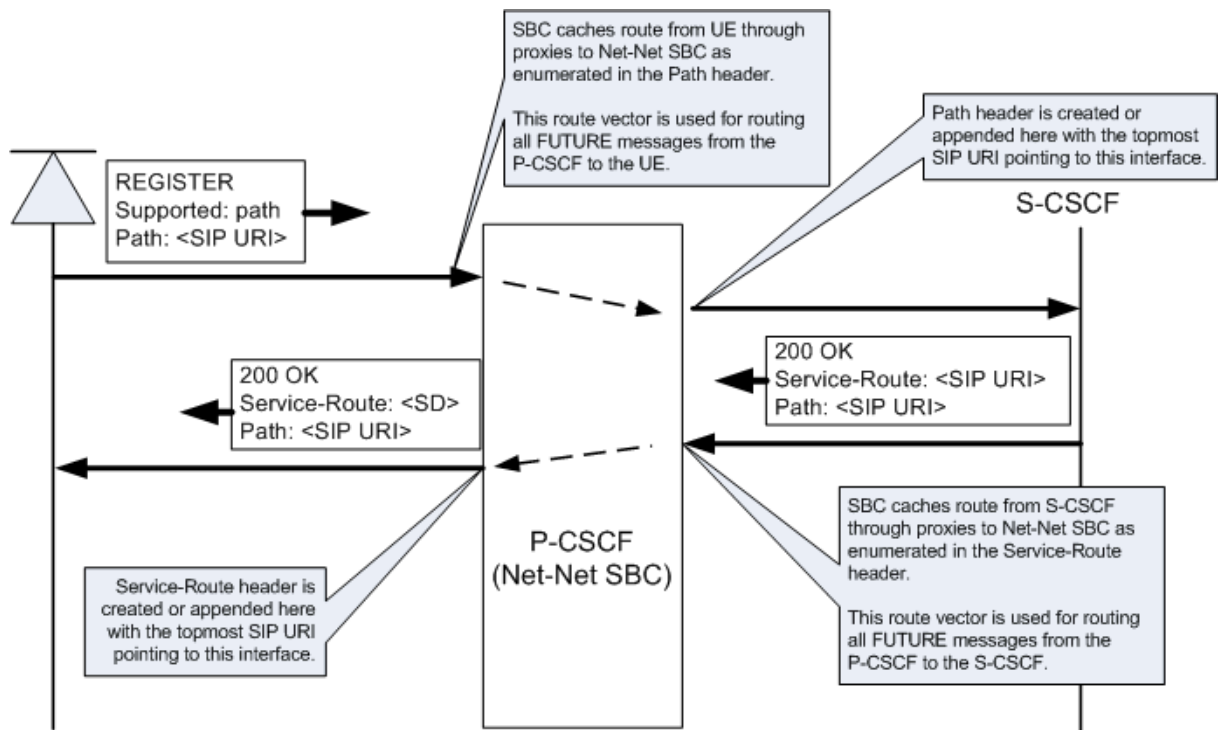
**Note:** In the configuration section the way to disable Service-Route header routing.

The manner in which passing or stripping of Service-Route headers sent from the S-CSCF is done is determined by local configuration on the Oracle Communications Session Border Controller. There is no verification of configured local policy against the route included in the stored service route. The Service-Route header, as created by the Oracle Communications Session Border Controller, and exiting back to the UE, contains a SIP URI pointing to itself as the topmost entry. This is used so that other proxies can learn the route back to the Oracle Communications Session Border Controller.

## Summary

If a request originates at the UE, the routes enumerated in the Service-Route header are used to route the request to the S-CSCF. If a request is meant to terminate at a UE, the routes enumerated in the Path header are used to route the response to the UE. Service-Route routes take priority over configured local policy.

Path headers received in a 200 OK response from the registrar are transmitted to the UE unchanged. If you want them stripped as the SIP message leaves the Oracle Communications Session Border Controller, you can use the SIP Header Manipulation function.



## IMS Path and Service Route Headers Configuration

IMS and all related functions must be enabled on both the access-side and core-side SIP interfaces. Only IMS features discussed up to this point are enabled by the following procedure.

To enable RFC 3608 and RFC 3327 support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. Type **select** and the number of the pre-configured sip interface you want to configure.

```
ACMEPACKET(sip-interface)# select 1
```

5. **sip-ims-feature**— Enable IMS functionality on this SIP interface. The default value is **disabled**. Valid values are:

- enabled | disabled

```
ACMEPACKET(sip-interface)# sip-ims-feature enabled
```

This completes enabling IMS for a given SIP interface.

**If you wish to disable subsequent routing of Register messages via the Service-Route header, type `route-register-no-service-route` and press Enter.**

6. Save your work using the ACLI **done** command.

## Wildcard Public User Identity (PUI)

The Oracle Communications Session Border Controller supports wildcard Public User Identities (PUI). This capability is most often used to streamline processing of REGISTER and INVITE messages between a PBX and an IMS core.

The HSS (Home Subscriber Server, an IMS network-wide database) is pre-provisioned with the extension numbers associated with the PBX's base telephone number. When the PBX registers its own base telephone number with the S-CSCF, that server downloads a wildcarded PUI -- a regular expression that describes all extension numbers associated with the registering PBX.

The S-CSCF returns the wildcarded PUI in the 200 OK response to the REGISTER message. Upon receiving the 200 OK, the SBC (acting in the P-CSCF role) uses the wildcarded PUI to construct a registration cache entry to implicitly treat subsequent INVITEs from PBXs extensions that match the wildcard as registered endpoints.

A wildcarded PUI consists of a delimited regular expression located either in the user info portion of the SIP URI or in the telephone-subscriber portion of the Tel URI. The regular expression takes the form of an Extended Regular Expressions (ERE) as defined in chapter 9 of The Single UNIX Specification (IEEE 1003.1-2004 Part 1). The exclamation mark (!) serves as the delimiter.

For example, the following PUIs will match to the wildcard ERE "sip:chatlist!\*!@example.com".

```
sip:chatlist1@example.com
sip:chatlist2@example.com
sip:chatlist42@example.com
sip:chatlistAbC@example.com
sip:chatlist!1@example.com
```

In addition to the ability of the P-CSCF to recognize and parse a wildcarded PUI, The P-CSCF must also support the PATH and P-Profile-Key SIP headers.

### Path Header

The Path header is defined in RFC 3327, Session Initiation Protocol (SIP) Extension Header Field for Registering Non-Adjacent Contacts. The Path header enables the accumulation and transfer of a list of proxies between a SIP UA and a REGISTER. Its operation is very similar to the Record-Route header in dialog initiating request. Within the context of support for wildcarded PUIs, the Path header is used to ensure that REGISTER responses containing an wildcard ERE transit the P-CSCF, thus enabling the implicit registration of PBX extensions.

### P-Profile-Key Header

The P-Profile-Key header is defined in RFC 5002, The Session Initiation Protocol (SIP) P-Profile-Key Private Header (P-Header). If the identity of the server user of the request was via a wildcarded PUI, that wildcarded PUI is included in the P-Profile-Key header in the outgoing request to the core. This Header will be included for requests like INVITE, SUBSCRIBE, and so on that match the registered wildcarded PUI. The header will occur in all subsequent requests to the core as well.

For this header to be present, the request needs to be going from access to core and the ims-feature must be enabled on both the access and core SIP interfaces.

### Registration Event Package

If subscription to the Registration Event package (defined in RFC 3680, A Session Initiation Protocol (SIP) Event Package for Registrations) is enabled, the P-CSCF receives asynchronous NOTIFYs from the core reporting registration state changes. Each NOTIFY provides a </reg-info> XML element that reports state changes. A sample XML report is shown below.

```
<registration aor="sip:user1_public1@home1.net" id="as10" state="active">
  <contact id="86" state="active" event="created">
    <uri>sip:[5555::aaa:bbb:ccc:ddd]</uri>
  </contact>
```

```
</registration>

<registration aor="sip:ep_user1@home1.net" id="as11" state="active">
    <contact id="86" state="active" event="created">
        <uri>sip:[5555::aaa:bbb:ccc:ddd]</uri>
    </contact>

    <ere:wildcardedIdentity>sip:ep_user!.*!@home1.net</
ere:wildcardedIdentity>

</registration>
</reginfo>
```

If a `</wildcardedIdentity>` sub-element is included in the `<registration>` element (as in the second `</registration>`), the wildcarded PUI is taken from the `</wildcardedIdentity>` sub-element, and that wildcarded PUI replaces any previously stored wildcarded PUIs in the registration.

In the absence of `</wildcardedIdentity>` sub-element(as in the first registration), the PUI is taken from the ‘aor’ attribute of the registration element, and the registration cache is left unchanged.

## Message Flows

A basic registration exchange with support for PUIs enabled.

From UA to P-CSCF

```
REGISTER sip:192.168.1.232:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.231:5060;branch=z9hG4bK-13794-1-0
From: <sip:7818888@hxu.com;tag=1
To: sut <sip:7818888@hxu.com>
Call-ID: 1-13794@192.168.1.231
CSeq: 1 REGISTER
Contact: <sip:7818888@192.168.1.231:5060>;expires=3600
Content-Length: 0
```

From P-CSCF to Registrar

If `reg-cache-route` is enabled on egress sip-interface, P-CSCF provides Path support with a value of P-CSCF-contact, so INVITE from core will have route header with value of this path per RFC 3327.

```
REGISTER sip:192.168.200.231:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.200.232:5060;branch=z9hG4bK0g6tnu200o10tfgl0641.1
From: <sip:7818888@hxu.com;tag=1
To: sut <sip:7818888@hxu.com>
Call-ID: 1-13794@192.168.1.231
CSeq: 1 REGISTER
Contact: <sip:7818888-rrbgth3ot667c@192.168.200.232:5060;transport=udp>;expires=3600
Content-Length: 0
Supported: path
Path: <sip:7818888-rrbgth3ot667c@192.168.200.232:5060; lr>
Max-Forwards: 70
```

From Registrar to P-CSCF

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.200.232:5060;branch=z9hG4bK0g6tnu200o10tfgl0641.1
Call-ID: 1-13794@192.168.1.231
CSeq: 1 REGISTER
```

```
Contact: <sip:7818888-rrbgth3ot667c@192.168.200.232:5060;transport=udp>;expires=3600
P-Associated-URI: <sip:781!.*!@hxu.com> -- wildcard regex
Content-Length: 0
```

#### From P-CSCF to UA

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.231:5060;branch=z9hG4bK-13794-1-0
From: <sip:7818888@hxu.com;tag=1
To: sut <sip:7818888@hxu.com>;tag=1
Call-ID: 1-13794@192.168.1.231
CSeq: 1 REGISTER
Contact: <sip:7818888@192.168.1.231:5060>;expires=3600
P-Associated-URI: <sip:781!.*!@hxu.com>
Content-Length: 0
```

#### Incoming Request INVITE from core

For the incoming request from the core to the P-CSCF, if reg-cache-route is enabled on ingress SIP Interface, P-CSCF checks the registered P-CSCF-contact-URI in the top Route P-CSCF-contact-URI, instead of with request-URI.

```
INVITE sip:7816666@192.168.200.232:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.200.231:5060;branch=z9hG4bK-13800-1-0
From: sipp <sip:3054252165@192.168.200.231:5060;user=phone>;tag=1
To: sut <sip:7816666@hxu.com>
Call-ID: 1-13800@192.168.200.231
Supported: 100rel,timer,resource-priority,replaces
CSeq: 1 INVITE
Contact: sip:7816666@192.168.200.231:5060
Max-Forwards: 70
Route: <sip:7818888-rrbgth3ot667c@192.168.200.232:5060;lr>
Content-Type: application/sdp
Content-Length: 141
```

```
v=0^M
o=user1 53655765 2353687637 IN IP4 192.168.200.231
s=-
c=IN IP4 192.168.200.231
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

#### outgoing Request INVITE

For outgoing Request from access side, the Register Cache entry will be found by reg-via-key, when options 'reg-via-key' and 'reg-via-match' are set, P-CSCF will have wildcarded PAU checking for allow-anonymous (or registered) verification if option wildcard-puid-match is set.

```
INVITE sip:service@192.168.1.232:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.231:5060;branch=z9hG4bK-11911-1-0
From: sipp <sip:7816666@hxu.com>;tag=1
To: sut <sip:service@192.168.1.232:5060>
Call-ID: 1-11911@192.168.1.231
Supported: 100rel,timer,resource-priority,replaces
CSeq: 1 INVITE
Contact: sip:781666@192.168.1.231:5060
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 137
```

```
v=0
o=user1 53655765 2353687637 IN IP4 192.168.1.231
s=-
c=IN IP4 192.168.1.231
t=0 0
```

```
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

### HA Configurations

Support for wildcarded PUIs requires checkpointing two new data points that need to be replicated in HA configurations.

- the P-Profile-Key in the sip session so that subsequent requests in a session can take the cached value and include that value in egress requests to core
- Upon SUB-NOT-200OK, the active P-CSCF replicates the subscribe session to the standby. This information is required to replicate the new PAU list of the associated SIP User if the </wildcardedIdentity> element arrives in the NOTIFY XML and has wildcarded entities.

### IMS Support for Private Header Extensions for 3GPP

---

As part of its RFC 3455 support, the Oracle Communications Session Border Controller supports the following headers in its IMS implementation:

- P-Associated-URI
- P-Asserted-Identity
- P-Called-Party-ID
- P-Charging-Function-Address
- P-Visited-Network-ID

The procedure to enable IMS support is explained in the previous section. IMS and all related functions must be enabled on both the access-side and core-side SIP interfaces.

#### P-Associated-URI Header

In the SIP registration process, the registrar often returns a set of associated URIs for a registering AoR. When the Oracle Communications Session Border Controller receives the list of associated URIs, it stores them in the registration entry for the registering endpoint. The service provider allocates one or more associated URIs per user for his or her own usage. After an endpoint successfully registers, the P-Associated-URI header returned in a 200 OK message informs the UE of all URIs associated with the AoR.

When the Oracle Communications Session Border Controller receives a request from a UE, the URI in the From header is matched against the registration cache for that endpoint. If the registering endpoint matches an associated-URI already in the registration table, the Service-Route associated with this endpoint is used to create the route for originating transactions associated with the endpoint to the S-CSCF.

The inclusion or exclusion of the P-Associated-URI header is not dependent on the trust level of an ingress or egress realm.

#### P-Asserted-Identity Header

The Oracle Communications Session Border Controller inserts a P-Asserted-Identity header into any initial request for a dialog or standalone transaction sourced by the UE.

The inclusion or exclusion of the P-Asserted-Identity header is dependent on the trust level of an egress realm.

#### P-Asserted-Identity Header Handling

1. The Oracle Communications Session Border Controller inserts a P-Asserted-Identity header into all messages other than the REGISTER message.
2. When the P-Preferred-Identity header is present in an INVITE sourced by the UE, and the SIP URI contained in this header is also present in the UE's associated URI list, then this SIP URI is inserted in the P-Asserted-Identity header as the SIP message enters the core network.



3. When the P-Asserted-Identity header is present in an INVITE sourced by the UE, and the SIP URI contained in this header is also present in the UE's associated URI list, then the original P-Asserted-Identity header and SIP URI is passed unchanged into the core network.
4. When the From header is present in an INVITE sourced by the UE, and the SIP URI contained in this header appears in the UE's Associated URI list, then this SIP URI is inserted into the P-Asserted-Identity header as the SIP message enters the core network.
5. When the P-Asserted-Identity header is present in an INVITE sourced by the UE, and the SIP URI contained in this header is not present in the Associated URI list, the Oracle Communications Session Border Controller acts like no P-Asserted-Identity was received from the UE.
6. When no P-Asserted-Identity can be derived from an INVITE sourced by the UE, the P-Asserted-Identity is based on the first URI in the Associated URI list.
7. The P-Asserted-Identity header will be removed from SIP messages sent and received from a UE if either the ingress or egress side is untrusted and the UE's Privacy header's contents is id.
8. If no P-Associated-URI exists for a registered endpoint, the Oracle Communications Session Border Controller will use the configured default P-Asserted-Identity found on the sourcing session agent. This feature works with both SIP and H.323 session agents.
9. If the session agent that originates a message does not include a P-Asserted-Identity header or the request is not originated from the session agent, and the P-CSCF has not received P-Associated-URI list from the registrar for a particular user, no P-Asserted-Identity will be created.
10. The P-Preferred-Identity header will never be passed to the S-CSCF.

If the above steps fail to insert a P-Asserted-Identity header, you can manually configure a value to be inserted into a P-Asserted-Identity header. The sip-ims-feature parameter must still be enabled to use the P-Asserted-Identity header override.

## P-Asserted-Identity Header Configuration

P-Asserted-Identity header handling is enabled with the sip-ims-feature as described in the previous section. A P-Asserted-Identity header can be manually configured for a session agent if the automatic logic, explained earlier in this section, fails.

To configure the P-Asserted-Identity header for a session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-level configuration elements.

```
ACMEPACKET (configure)# session-router
ACMEPACKET (session-router)#
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router)# session-agent
ACMEPACKET (session-agent)#
```

4. Type **select** and the number of the pre-configured session agent you want to configure.

```
ACMEPACKET (session-agent)# select 1
```

5. Type **p-asserted-id** <space> <URI to use when no P-Asserted-ID has been created> and press Enter. This completes the configuration.

```
ACMEPACKET (session-agent)# p-asserted-id sip:name@acmepacket.com
```

6. Save your work using the ACLI **done** command.

## P-Called-Party-ID Header

The Oracle Communications Session Border Controller transparently passes the P-Called-Party-ID header between the S-CSCF and a UA.

## IMS Charging Headers

The Oracle Communications Session Border Controller supports IMS Charging Headers. These headers include P-Charging-Vector and the P-Charging-Function-Address. IMS charging header support is configured separately from other IMS functions in order to support a variety of customer needs. Charging header information is now recorded in the CDR records.

A charging vector is defined as a collection of charging information. The charging vector may be filled in during the establishment of a dialog or standalone transaction outside a dialog. The information inside the charging vector may be filled in by multiple network entities (including SIP proxies) and retrieved by multiple network entities. There are three types of correlation information to be transferred: the IMS Charging Identity (ICID) value, the address of the SIP proxy that creates the ICID value, and the Inter Operator Identifiers (IOI).

Charging headers are inserted, deleted, or ignored for request messages. They are forwarded through the Oracle Communications Session Border Controller unmodified when embedded in response messages. If you wish to modify the charging headers in a response message, you must use the Oracle Communications Session Border Controller's header manipulation feature as a general solution.

### P-Charging-Vector

You can configure the Oracle Communications Session Border Controller to process the P-Charging Vector header in three different ways.

- If a P-Charging-vector header is present in an incoming SIP request, the Oracle Communications Session Border Controller can pass the header untouched, as part of the full SIP message that is forwarded out of an egress interface.
- If a P-Charging-vector header is present in an incoming SIP request, the Oracle Communications Session Border Controller can delete the header and forward the full SIP message out of an egress interface.
- If an incoming SIP request does not contain a P-charging-vector header, the Oracle Communications Session Border Controller can create and insert the header and forward the full SIP message out of an egress interface. Likewise, if an incoming SIP request contains an existing P-Charging-Vector header, the Oracle Communications Session Border Controller can overwrite this header with the values generated internally.

The P-Charging-Vector header is composed of four parameters: `icid-value`, `icid-gen-addr`, `orig-ioi`, `term-ioi`. See RFC 3455, Section 4.6 for more information.

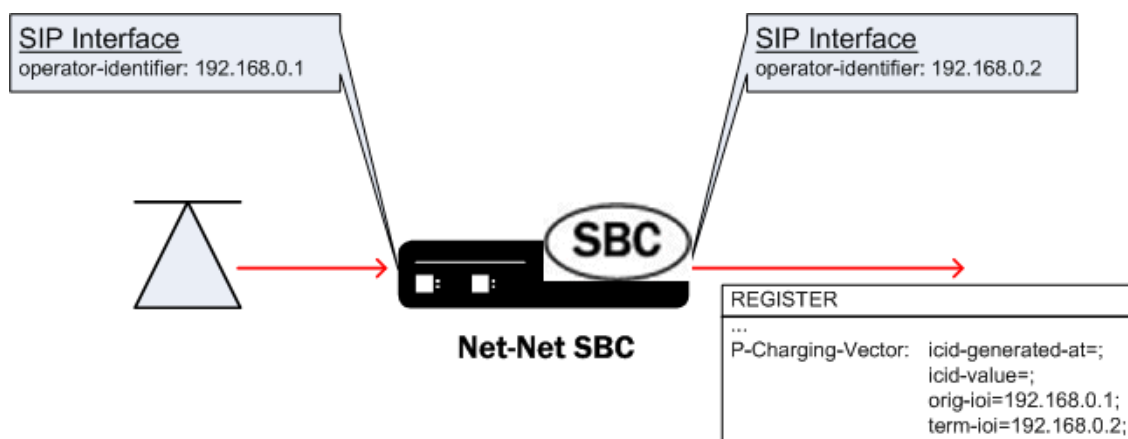
1. The Oracle Communications Session Border Controller constructs the `icid-value` in the following format:  
`string2@string1` where:

String 1 is the IP address of the egress SIP interface.

String 2 is a unique string value created by the Oracle Communications Session Border Controller and based on the realm, local IP port, time, and a sequence number.

2. The `icid-gen-addr` parameter's value is the IP address of the egress SIP interface. This value is generated by the Oracle Communications Session Border Controller.
3. The `orig-ioi` parameter's value is set manually using the operator-identifier field located in the SIP interface configuration element.
4. The `term-ioi` parameter's value is set manually using the operator-identifier field located in the SIP interface configuration element.

You configure charging vector handling on the Oracle Communications Session Border Controller interface that receives the SIP request by turning on the switches that enable charging vector processing on the ingress interface for the call. Based on the direction of the call, the Oracle Communications Session Border Controller will insert the operator-identifier configuration parameter into the `orig-ioi` and the `term-ioi` parameters. The `orig-ioi` parameter takes the value of the operator-identifier configuration parameter of the SIP interface that receives the SIP request. The `term-ioi` parameter takes the value of the operator-identifier configuration parameter of the SIP interface that sends the SIP request to its next hop.



### P-Charging-Vector Header Example

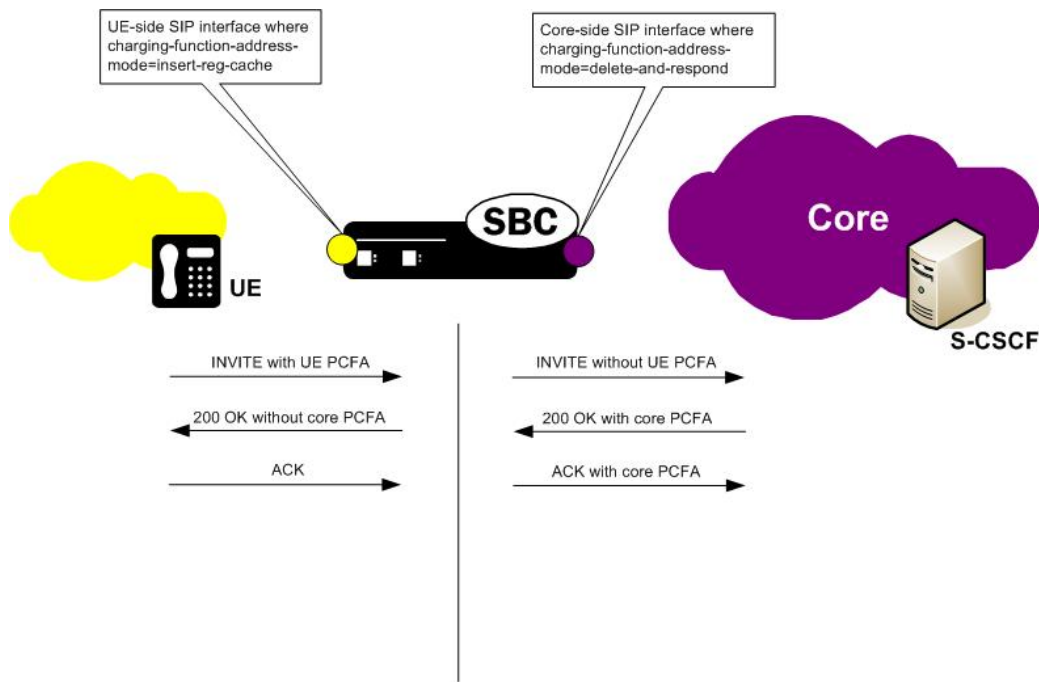
```
P-Charging-Vector: icid-
value=late6g46n1823s8719ck3ps6gbt46m5d3bci3po5hhdg3n86g1csio47g9c43@192.168.0.
2;
icid-generated-at=192.168.0.2;
orig-ioi=192.168.0.1;
term-ioi=192.168.0.2;
```

### P-Charging-Function-Address

The P-Charging-Function-Address header is composed of two configurable parameters: ccf, ecf. You can configure the Oracle Communications Session Border Controller to process the P-Charging-Function-Address header in four different ways.

- If a P-Charging-Function-Address header is present in an incoming SIP request, the Oracle Communications Session Border Controller can be set to pass the header, untouched, as the full SIP request is forwarded out of an egress interface.
- If a P-Charging-Function-Address header is present in an incoming SIP request, the Oracle Communications Session Border Controller can be set to delete the header and forward the SIP request out of an egress interface.
- If an incoming SIP request does not contain a P-Charging-Function-Address header, the Oracle Communications Session Border Controller can be set to create and insert the header and forward the SIP message out of an egress interface.
- If an incoming SIP request contains a P-Charging-Function-Address header, and the Oracle Communications Session Border Controller is set to insert a configured P-Charging-Function-Address header, the new parameters will be appended before the existing parameters in the header. The Oracle Communications Session Border Controller will then forward the SIP request out of an egress interface.

In addition, the Oracle Communications Session Border Controller can also be configured to perform insertion and caching for the PCFA header in dialog-creating or stand-alone messages. The following diagram illustrates how this works:



For this scenario, there are two main functions, PCFA insertion and PCFA caching:

- PCFA insertion—Using the insert-reg-cache and delete-and-respond configuration values, the Oracle Communications Session Border Controller adds the PCFA to all SIP requests and to the response on the S-CPCF facing the SIP interface. However, only dialog-creating and standalone requests, and responses to each of those, update the Oracle Communications Session Border Controller and accounting information. Such requests do not have a To tag, and responses do not appear in established dialogs. The Oracle Communications Session Border Controller inserts the PCFA into provisional (1XX) and success (2XX) responses, with the exception of the 100 Trying response.


You can use SIP header manipulation rules (HMR) to remove any unwanted headers.

- PCFA caching—When you use either of the insert-reg-cache and delete-and-respond configuration values, the Oracle Communications Session Border Controller uses the latest cached copy of a PCFA header to insert into requests and responses. The Oracle Communications Session Border Controller does not cache any PCFA headers it receives on SIP interfaces using the none, insert, or insert-reg-cache modes because this type of SIP interface faces the UE making its replacement headers ones from the core.

Though there can be various sources for the latest cached copy, the PCFA header received as part of a dialog-creating or standalone request has highest precedence. This PCFA header is then stored as the latest cached value for that dialog. That is, for each specific dialog, the Oracle Communications Session Border Controller the PCFA is cached separately so it can add the most specific PCFA to the message—and is added to any message for the dialog.

When there is no cache PCFA for a specific dialog, the Oracle Communications Session Border Controller uses the registration cache entry as the latest cached copy. And when there is no entry in the registration, the PCFA uses the **ccf-address** and **ecf-address** values from the SIP interface.

The latest cached copy or the **ccf-address** is the value reported in the RADIUS VSA Acme-Session-Charging-Function-Address; this VSA is used for both of the new modes.

 **Note:** Only the **ccf-address** is reported in RADIUS records; the **ecf-address** is not.

## P-Charging-Function-Address Header Example

```
P-Charging-Function-Address: ccf=192.168.0.20 ; ecf=192.168.0.21;
```

## RADIUS Accounting of Charging Headers

When the Oracle Communications Session Border Controller creates either the P-Charging-Vector header or the P-Charging-Function-Address header, it inserts an entry in the RADIUS record to record the charging header data.

For a P-Charging-Vector header, the icid-value is saved to the P-Charging-Vector attribute in the radius record. If the Oracle Communications Session Border Controller does not create a P-Charging-Vector header, but it receives a SIP message that already has the P-Charging-Vector header with an icid-value, the existing icid-value is written to the RADIUS record.

For a P-Charging-Function-Address header, the first CCF value is saved to the P-Charging-Function-Address attribute. When the Oracle Communications Session Border Controller creates the P-Charging-Function-Address, the CCF value it inserts into the header is saved to the radius record. If the Oracle Communications Session Border Controller does not create a P-Charging-Function-Address header, but it receives a SIP message that already has the P-Charging-Function-Address with a CCF value, the existing CCF value is written to the RADIUS record.

Name	Value	Value Type
Acme-Session-Charging-Vector	54	string
Acme-Session-Charging-Function-Address	55	string

## P-Charging-Vector Processing for SIP Interfaces Configuration

P-Charging-Vector header handling is enabled in the SIP interface.

To configure P-Charging-Vector processing in a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter.


```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

If you are adding support to an existing SIP interface, then you will need to select the interface you want to edit using the ACLI **select** command.

```
ACMEPACKET(sip-interface)# select
```

4. **charging-vector-mode**—Sets the Oracle Communications Session Border Controller to create the P-Charging-Vector header. The default value is **pass**. The valid values are:

- **none**—Pass the P-Charging-Vector header received in an incoming SIP message untouched as the message is forwarded out of the Oracle Communications Session Border Controller, but does not include icid-value in accounting records.
- **pass**—Pass the P-Charging-Vector header received in an incoming SIP message untouched as the message is forwarded out of the Oracle Communications Session Border Controller, includes icid-value in accounting records.
- **delete**—Delete the P-Charging-Vector header received in an incoming SIP message before it is forwarded out of the Oracle Communications Session Border Controller.
- **insert**—Insert the P-Charging-Vector-Mode header in an incoming SIP message that does not contain the Charging-Vector header. If the incoming message contains the Charging-Vector header, the Oracle Communications Session Border Controller will overwrite the Charging-Vector-Mode header with its values. This option always uses the **ccf-address** and **ecf-address** static values.

- **delete-and-respond**—To be configured on the SIP interface facing the S-CPCF, configures the Oracle Communications Session Border Controller to strip out the latest cached PCFA from the core side. The Oracle Communications Session Border Controller then remembers this PCFA and uses it in communications sent to the core.
5. **operator-identifier**—Set the operator identifier value to be inserted into a P-Charging-Vector header. The direction of the call determines whether this value is inserted into the orig-ioi or the term-ioi parameter in the P-Charging-Vector header. This string **MUST** begin with an alphabetical character.
  6. **charging-function-address-mode**—Set the charging function address mode you want to use. The default value is **pass**. The valid values are:
    - **none**—Pass the P-Charging-Function-Address header received in an incoming SIP message untouched as the message is forwarded out of the Oracle Communications Session Border Controller , but does not include icid-value in accounting records.
    - **pass**—Pass the P-Charging-Function-Address header received in an incoming SIP message untouched as the message is forwarded out of the Oracle Communications Session Border Controller , includes icid-value in accounting records.
    - **delete**—Delete the P-Charging-Function-Address header received in an incoming SIP message before it is forwarded out of the Oracle Communications Session Border Controller .
    - **insert**—Insert the P-Charging-Function-Address header in an incoming SIP message that does not contain the P-Charging-Function-Address header. If the incoming message contains the P-Charging-Function-Address header, the Oracle Communications Session Border Controller will overwrite the P-Charging-Function-Address header with its values. This option always uses the **ccf-address** and **ecf-address** static values.
    - **insert-reg-cache**—To be configured on the SIP interface facing the UE, configures the Oracle Communications Session Border Controller to replace the PCFA with the most recently cached value rather than the **ccf-address** and **ecf-address** you set to be static in your configuration. The cached values come from one of the following that the Oracle Communications Session Border Controller has received most recently: request, response, registration, or local configuration.
    - **delete-and-respond**—To be configured on the SIP interface facing the S-CPCF, configures the Oracle Communications Session Border Controller to strip out the latest cached PCFA from the core side. The Oracle Communications Session Border Controller then remembers this PCFA and uses it in communications sent to the core.
-  **Note:** The default settings for this parameter and for **charging-vector-mode** are pass for new SIP interface configurations. If you are upgrading and there are pre-existing SIP interfaces in your configuration, the defaults become none.
7. **ccf-address**—Set the CCF address value that will be inserted into the P-Charging-Function-Address header.
  8. **ecf-address**—Set the ECF address value that will be inserted into the P-Charging-Function-Address header.
  9. Save your work using the ACLI **done** command.

## Charging Correlation

IMS is based on a distributed architecture that can result in multiple network entities becoming involved in providing access and services. Operators require the ability to charge for the access and services as they provide. This necessitates coordination among the network entities (for example, SIP proxies), which includes correlating charging records generated from different entities that are supporting the same session.

Charging correlation is supported in Release SC-X7.1.2 and later.

During initial provisioning of session information, the Oracle Communications Session Border Controller, functioning as a P-CSCF, indicates its support for charging correlation by subscribing to access network charging information from the PCRF (Policy Charging Rule Function). In a typical exchange, the subscription is requested with an AA-Request (AA-R) command that contains the Specific-Action (Type 513) and AF-ChargingIdentifier (Type

505) AVPs. The Specific-Action AVP is set to 1, (CHARGING\_CORRELATION\_EXCHANGE), requesting that the PCRF provide new and updated charging information as such information becomes available.

The PCRF responds with an AA-Answer (AA-A) that usually contains one or more associated Access-Network-Charging-Identifier (Type 502) and an Access-Network-Charging-Address (Type 501) AVPs. The Access-Network-Charging-Address AVP contains the IP address of a network entity that is charging for session services. The Access-Network-Charging-Identifier associated with the entity identified by the Access-Network-Charging-Address AVP. This identifier is applied to all traffic/media flows within the current session.

The AA-A can contain an IP-CAN-Type (Type 1027) AVP that indicates the type of Connectivity Access Network and an associated RAT-Type AVP that provides more specific information as to the access technology. For IMS charging correlation, the IP-CAN-Type AVP can contain one or two values: 0 (3GPP), or 5 (3GPP-EPS).

In some cases, the PCRF may not have the requested charging information at the time of the initial AA-R. When information becomes available, the PCRF sends the data to the P-CSCF via a Re-Auth-Request (RAR) command, which is then acknowledged by the P-CSCF with a Re-Auth-Answer (RAA) command.

Upon receiving charging information from the PCRF, the P-CSCF saves the data in the context of the associated SIP session. Saved data is used to populate the P-Charging-Vector SIP header.

## Charging Correlation Configuration

To enable support for charging correlation:

1. Navigate to ext-policy-server configuration mode.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# ext-policy-server
ACMEPACKET(ext-policy-server)#
```

2. **specific-action-subscription**—Enables the system to subscribe to IMS action-specific notification services. By default, no subscriptions are enabled. Available subscriptions are:

- charging-correlation-exchange
- loss-of-bearer
- recovery-of-bearer
- release-of-bearer
- out-of-credit
- successful-resource-allocation
- failed-resource-allocation

```
ACMEPACKET(ext-policy-server)# specific-action-subscription changing-
correlation-exchange
```

adds a single subscription

```
ACMEPACKET(ext-policy-server)# specific-action-subscription (changing-
correlation-exchange loss-of-bearer recovery-of-bearer)
```

adds multiple subscriptions

3. Use done, exit, and verify-config to complete configuration.

## P-Early-Media SIP Header Support

Version S-CZ7.2.0 provides support for the SIP P-Early-Media that can be used in SIP INVITES, PRACKS, and UPDATES to request and authorize the use of early media. It offers an alternative to policy-based early media support.

The P-Early-Media SIP header is defined in RFC 5009, Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media. RFC 5009 defines the use of the P-Early-Media header within SIP messages in certain SIP networks to authorize the cut-through of backward (server-to-client) and/or forward (client-to-server) early media when permitted by the early media policies of the networks involved. The P-Early-Media header field is intended for use in a SIP network, such as a 3GPP IMS that has the following characteristics: its early media policy prohibits the exchange of early media between end users; it is interconnected with other SIP networks

that have unknown, untrusted, or different policies regarding early media; and it has the capability to gate (enable/disable) the flow of early media to/from user equipment.

### P-Early-Media SIP Header

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A SIP network containing both PSTN gateways and SIP end devices, for example, can maintain such an early media policy by gating "off" any early media with a SIP end device acting as UAS, gating "on" early media with a SIP end device acting as UAC, and gating "on" early media at each PSTN gateway. This is what we have been doing for years with the SIP early media suppression feature, which allows determining who can send early media and in what direction.

Unfortunately, in SIP interconnection scenarios there is no means of assuring that the interconnected network is implementing a compatible early media policy, thus allowing the exchange of user data within early media under some circumstances. For example, if a network "A" allows all early media with user equipment as UAC and an interconnected network "B" allows all early media with user equipment as UAS, any session established between user equipment as UAC in "A" and user equipment as UAS in "B" will allow bidirectional user data exchange as early media.

The P-Early-Media header is used for the purpose of requesting and authorizing requests for backward and/or forward early media. It's sent from UAS to UAC to indicate authorization for early media.

### P-Early-Media-Header Usage

The syntax of the P-Early-Media header field is as follows.

```
P-Early-Media = "P-Early-Media" HCOLON[ em-param *(COMMA em-param) ]
               em-param = "sendrecv" / "sendonly" / "recvonly" / "inactive" /
               "gated" /
               "supported" / token
```

The P-Early-Media header is used for requesting and authorizing requests for backward and/or forward early media. The P-Early-Media header field in an INVITE request contains the "supported" parameter. If P-CSCF is part of the trusted domain, then it must decide whether to insert or delete the P-Early-Media header field before forwarding the INVITE. The P-CSCF upon receiving the P-Early-Media header field in a message towards the UAC needs to verify that the early media request comes from an authorized source. If a P-Early-Media header field arrives from either an untrusted source, a source not allowed to send backward early media, or a source not allowed to receive forward early media, then it may remove the P-Early-Media header field or alter the direction parameter(s) of the P-Early-Media header field before forwarding the message, based on local policy.

The P-Early-Media header field with the "supported" parameter in an INVITE request indicates that the P-CSCF on the path recognizes the header field. The P-Early-Media header field includes one or more direction parameters where each has one of the values: "sendrecv", "sendonly", "recvonly", or "inactive", following the convention used for SDP stream directionality. Each parameter applies, in order, to the media lines in the corresponding SDP messages establishing session media. The parameter value "sendrecv" indicates a request for authorization of early media associated with the corresponding media line, both from the UAS towards the UAC and from the UAC towards the UAS. The value "sendonly" indicates request for authorization of early media from the sender to the receiver and not in the other direction. The value "recvonly" indicates a request for authorization of early media from the receiver, and not in the other direction. The value "inactive" indicates either a request that no early media associated with the corresponding media line be authorized, or a request for revocation of authorization of previously authorized early media. Each parameter applies, in order, to the media lines in the corresponding SDP lines. Unrecognized parameters are discarded and non-direction parameters are ignored. If there are more direction parameters than media lines, the excess are silently discarded. If there are fewer direction parameters than media lines, the value of the last direction



parameter applies to all remaining media lines. The P-Early-Media header field in any message within a dialog towards the sender of the INVITE request can also include the non-direction parameter "gated" to indicate that a network entity on the path towards the UAS is already gating the early media, according to the direction parameter(s).

As defined in 3GPP TS 24.229, IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 both the P-CSCF and IBCF may add, remove, or modify, the P-Early-Media header field within forwarded SIP requests and responses according to procedures in RFC 5009.

The P-CSCF can use the P-Early-Media header field for the gate control procedures, as described in 3GPP TS 29.214. Prior to Version S-CZ7.2.0, this capability was based on policy configuration, not examination of the P-Early-Media header.

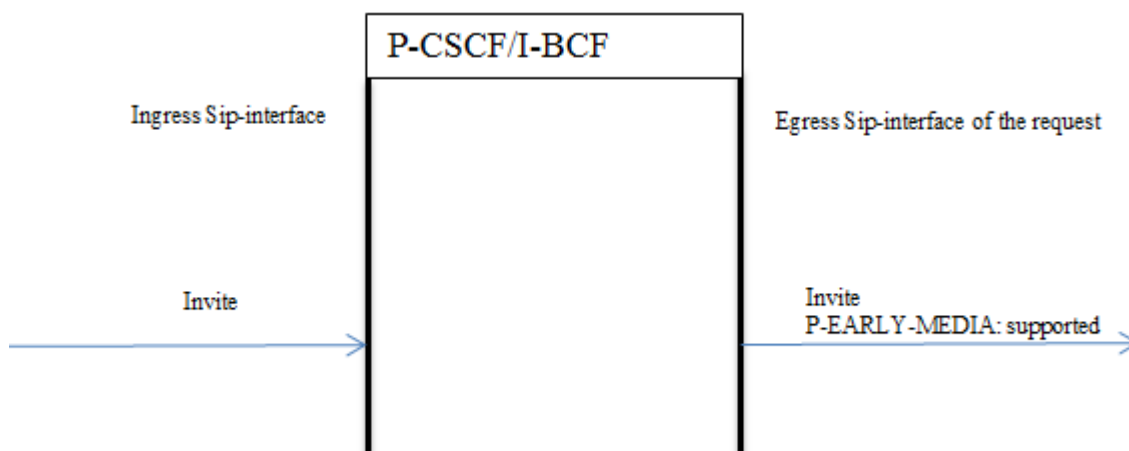
In the current implementation, if the configuration option "early-media-allow" is set to none, the Oracle Communications Session Border Controller will send the Flow-Status AVP in any AAR request set to disable until the final response.

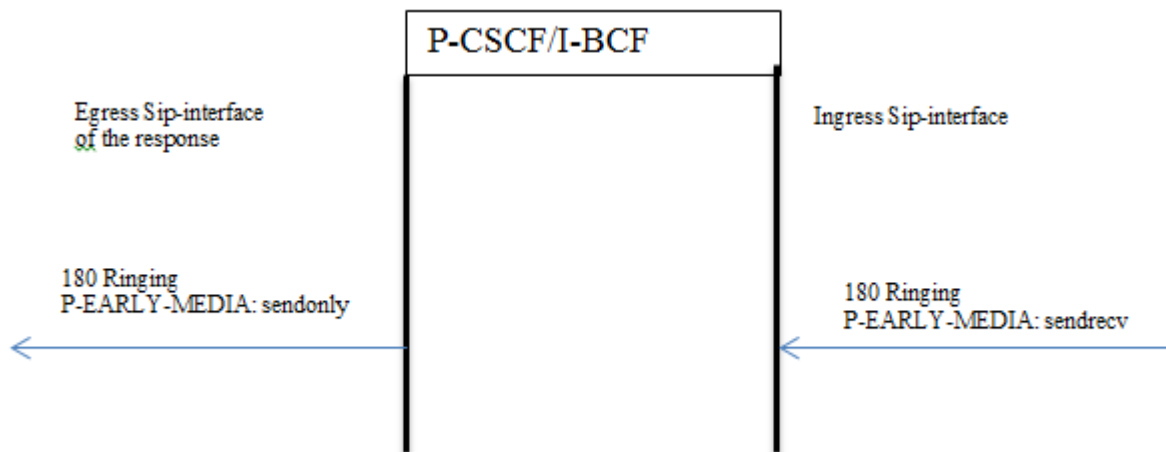
## Functional Design

Acceptance of and authorization for early media is accomplished with two new ACLI parameters -- **p-early-media-header** and **p-early-media-direction**, which are added to SIP interface configuration in Version S-CZ7.2.0 and later releases.

The **p-early-media-header** parameter will enable the feature when the value is set to either "add" or "modify". The **p-early-media-header** and **p-early-media-direction** should be configured on egress interface of the incoming message. The values for parameter **p-early-media-direction** are "sendrecv, sendonly, recvonly, inactive". It is a list and each configured value corresponds to the m-line in the SDP. If the number of configured values is more than the number of m-lines in the SDP, the excess configured values are ignored. If the number of configured value is less than the number of m-lines in the SDP, the last configured value is used for all the m-lines.

The following illustrations show the ingress and egress sip-interface configuration.





P-Early-Media headers in Re-Invites are ignored. If the SDP contains Content-Disposition: early-session the P-Early-Media header is ignored.

Endpoint is considered trusted or untrusted based on the configuration on the ingress sip-interface of the P-CSCF. Sip-interface has the configuration parameter “trust-mode”. If the “trust-mode” is set to “none” then nobody is trusted in sip-interface. By default the value is “all”. Possible values are <all, agents-only, realm-prefix, registered, none>.

For multiple dialogs due to forking, P-CSCF will identify the media associated with a dialog, and then setup early media flow for the selected media. The configuration elements restricted-latching in realm-config, and latching in media-router should be enabled.

The following 3 tables detail early media implementation details.

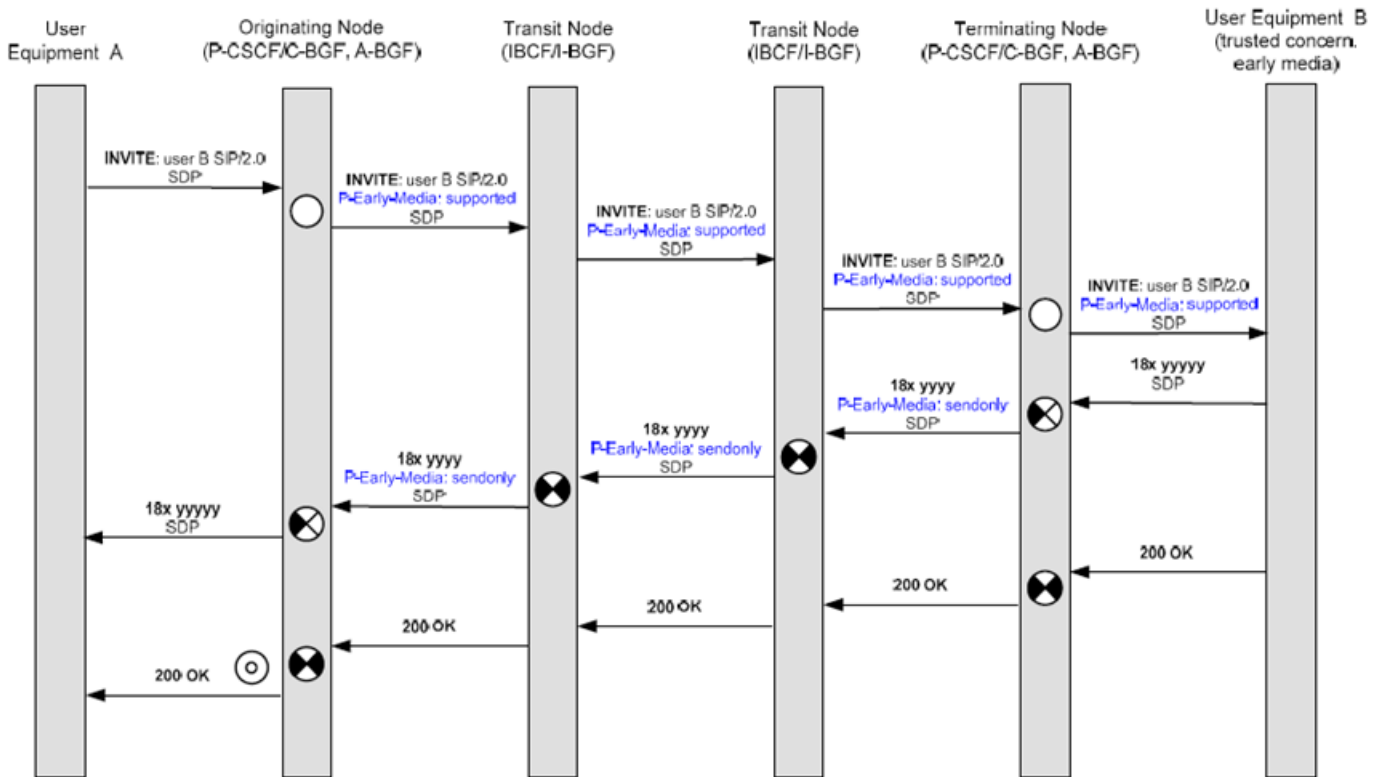
**P-Early-Media Trusted/Untrusted to Trusted**

This table illustrates the P-CSCF case when messages are received from trusted/untrusted endpoints and forwarded to trusted endpoints.

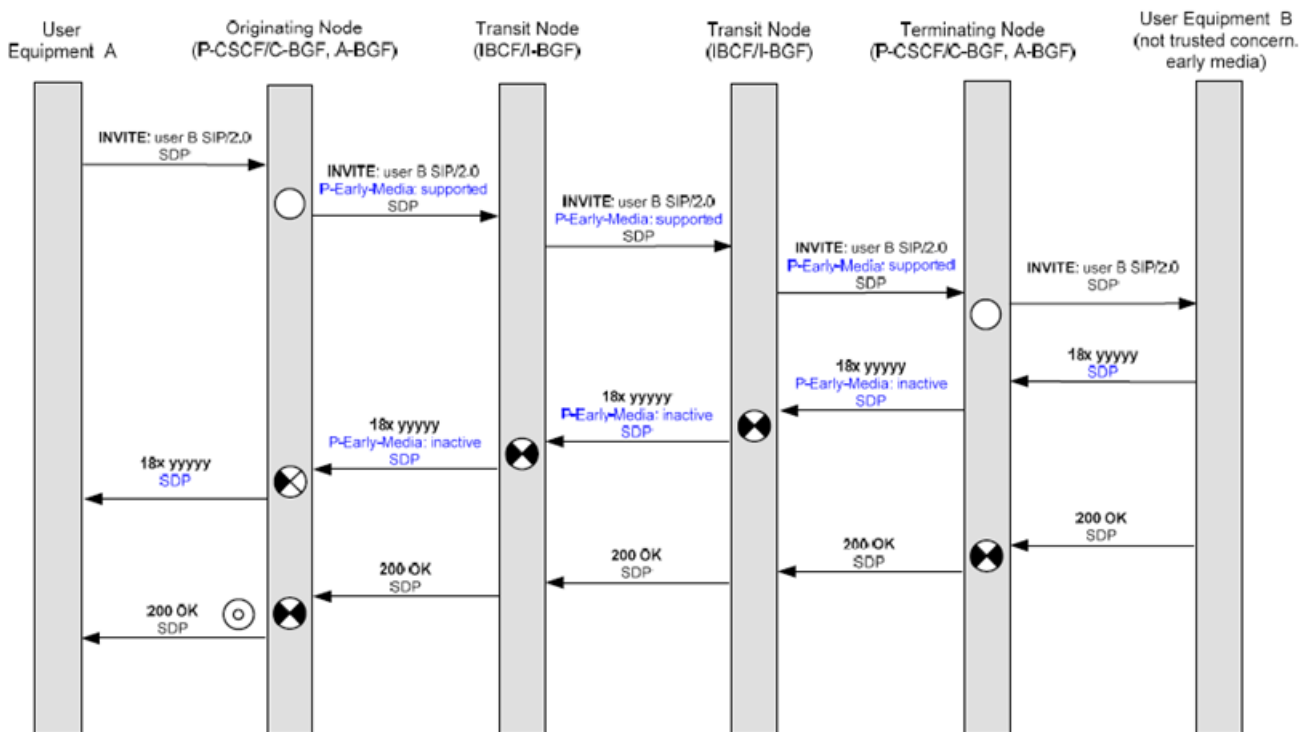
Message -->	INVITE (no PEM)	INVITE (with PEM)	18x (no PEM)	18x (with PEM)	UPDATE/ PRACK (no PEM)	UPDATE/ PRACK (with PEM)	INVITE/18x /UPDATE/ PRACK
Parameters configured on  egress interface		with one or more direction parameters where each has one of the values: "sendrecv", "sendonly", "recvonly", or"inactive"		with one or more direction parameters where each has one of the values: "sendrecv", "sendonly", "recvonly", or"inactive"		with one or more direction parameters where each has one of the values: "sendrecv", "sendonly", "recvonly", or"inactive"	with unrecognized PEM value
p-early-media-header-“add” p-early-media-direction - “sendonly	Add header with "supported" value. Setup flows based on configured value.	Add "supported" value if not present. Setup flows based on the value in the header	Add header with configured value and "gated" value.	Add the "gated" value. Setup the flows based on the value in the header.	Add header with configured value and "gated" value. Setup flows based on	Add the "gated" value . Setup the flows based on the value in the header	Discard the header

					configured value.		
p-early-media-header-"modify" p-early-media-direction -"sendonly"	Pass the message transparently.	Add "supported" value if not present. Setup flows based on the value in the header.	Pass the message transparently.	Modify header with configured value and add "gated" value. Setup flows based on configured value.	Pass the message transparently.	Modify header with configured value and add "gated" values. Setup flows based on configured value.	Discard the header
p-early-media-header-"add" No p-early-media-direction	Add header with "supported" value. Setup flows based on default value.	Add "supported" value if not present. Setup flows based on the value in the header.	Add header with default value and "gated" value. Setup flows based on default value.	Pass the message transparently.	Add header with default value and "gated" value. Setup flows based on default value.	Pass the message transparently.	Discard the header
p-early-media-header-"modify" No p-early-media-direction	Pass the message transparently.	Add "supported" value if not present. Setup flows based on the value in the header.	Pass the message transparently.	Modify header with default value and "gated" values. Setup flows based on configured value.	Pass the message transparently.	Modify header with configured value and add "gated" values. Setup flows based on configured value.	Discard the header

This illustration shows P-Early-Media set-up with a trusted endpoint.



This illustration shows P-Early-Media set-up with an untrusted endpoint.



### P-Early-Media Untrusted to Trusted

This table illustrates the P-CSCF case when messages are received from untrusted endpoints and forwarded to trusted endpoints.

Message -->	INVITE (no PEM)	INVITE (with PEM)	18x (no PEM)	18x (with PEM)	UPDATE/PRACK (no PEM)	UPDATE/PRACK (with PEM)	INVITE/18x/UPDATE/PRACK
Parameters configured on  egress interface		with one or more direction parameters where each has one of the values: "sendrecv", "sendonly", "recvonly", or "inactive"		with one or more direction parameters where each has one of the values: "sendrecv", "sendonly", "recvonly", or "inactive"		with one or more direction parameters where each has one of the values: "sendrecv", "sendonly", "recvonly", or "inactive"	with unrecognized PEM value
p-early-media-header-"add" p-early-media-direction - "sendonly"	Add header with "supported" value. Setup flows based on configured value.	Discard the header.	Add header with configured value and "gated" value. Setup the flows based on configured value.	Discard the header.	Add header with configured value and "gated" value. Setup flows based on configured value.	Discard the header.	Discard the header .
p-early-media-header-"modify" p-early-media-direction -"sendonly"	Pass the message transparently.	Discard the header.	Pass the message transparently.	Discard the header.	Pass the message transparently.	Discard the header.	Discard the header.
p-early-media-header-"add" No p-early-media-direction	Add header with "supported" value. Setup flows based on default value.	Discard the header.	Add header with default value and "gated" value. Setup flows based on default value.	Discard the header.	Add header with default value and "gated" value. Setup flows based on default value.	Discard the header.	Discard the header.
p-early-media-header-"modify" No p-early-media-direction	Pass the message transparently.	Discard the header.	Pass the message transparently.	Discard the header.	Pass the message transparently.	Discard the header.	Discard the header.

### P-Early-Media Trusted to Untrusted

This table illustrates the P-CSCF case when messages are received from trusted endpoints and forwarded to untrusted endpoints.

Message -->	INVITE (no PEM)	INVITE (with PEM)	18x/UPDATE/PRACK (no PEM)	18x (with PEM)	UPDATE/PRACK (no PEM)	INVITE/18x/UPDATE/PRACK
Parameters configured on egress interface						
p-early-media-header- "add"  p-early-media-direction -"sendonly"	Pass the message transparently.	Delete the header from the message.	Pass the message transparently	Delete the header from the message.	Delete the header from the message.	N/A
p-early-media-header-"modify"  p-early-media-direction -"sendonly"	Pass the message transparently.	Delete the header from the message.	Pass the message transparently.	Delete the header from the message.	Delete the header from the message.	N/A
p-early-media-header-"add"  No early-media-direction	Pass the message transparently.	Delete the header from the message.	Pass the message transparently	Delete the header from the message.	Delete the header from the message.	N/A
p-early-media-header-"modify"  No early-media-direction	Pass the message transparently.	Delete the header from the message.	Pass the message transparently.	Delete the header from the message.	Delete the header from the message.	N/A

**P-Early-Media ACLI Configuration**

Use the following procedure to configure P-Early-Media SIP header support.

1. Access the **sip-interface** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

2. Select the **sip-interface** object to edit.

```
ACMEPACKET(sip-interface)# select
<RealmID>:
1: realm01 172.172.30.31:5060

selection: 1
ACMEPACKET(sip-interface)#
```

3. Use the **p-early-media-header** parameter to enable P-Early-Media SIP header support. This parameter is disabled by default.
  - **disabled**—(the default value) disables support
  - **add**—enables support and allows the Oracle Communications Session Border Controller to add the P-Early-Media header to SIP messages.
  - **modify**—enables support and allows the Oracle Communications Session Border Controller to modify or strip the P-Early-Media header in SIP messages.

```
ACMEPACKET(sip-interface)# p-early-media-header add
ACMEPACKET(sip-interface)#
```

4. Use the **p-early-media-direction** parameter to specify the supported directionalities.

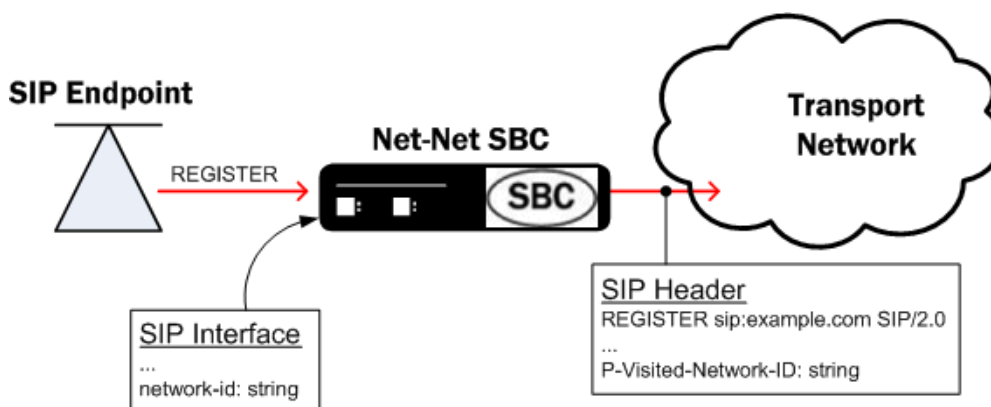
- **sendrecv**—send and accept early media
- **sendonly**—send early media
- **recvonly**—receive early media
- **inactive**—reject/cancel early media

```
ACMEPACKET(sip-interface)# p-early-media-direction sendrecv,sendrecv
ACMEPACKET(sip-interface)#
```

5. Type **done** to save your configuration.

## P-Visited-Network-ID Header

The Oracle Communications Session Border Controller's IMS support also includes the insertion of a P-Visited-Network-ID header into SIP messages when applicable. When a UE sends a dialog-initiating request (e.g., REGISTER or INVITE message) or a standalone request outside of a dialog (e.g., OPTIONS) to the P-CSCF, the Oracle Communications Session Border Controller inserts the P-Visited-Network-ID header into the SIP message as it enters into the destination network.



The P-Visited-Network ID header will be stripped from SIP messages forwarded into untrusted networks as expected. The content of a P-Visited-Network-ID header is a text string that identifies the originating UE's home network. This string is user-configurable.

## P-Visited-Network-ID Header Handling for SIP Interfaces Configuration

P-Visited-Network-ID header handling is enabled with the sip-ims-feature as described earlier. The actual P-Visited-Network-ID string must be configured on the access-side SIP interface. The Oracle Communications Session Border Controller must consider the egress device trusted or it does not add that the P-Visited-Network-ID header to the forwarded request.

To configure the P-Visited-Network-ID string in a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session-level configuration elements.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. Type **select** and the number of the pre-configured sip interface you want to configure.

```
ACMEPACKET(sip-interface) # select 1
```

5. Type **network-id** <space> <network ID string> and press Enter. This completes the configuration of the P-Visited-Network-ID string for a given SIP interface.

```
ACMEPACKET(sip-interface) # network-id examplenetworkid
```

6. Save your work using the ACLI **done** command.

## Second P-Asserted-Identity Header for Emergency Calls

The Oracle Communications Session Border Controller can add a second P-Asserted-Identity header when forwarding an emergency message into the network.

When the UE registers with an S-CSCF, the S-CSCF returns a set of associated URIs, the implicit registration set (IRS,) for the AoR in the 200 OK response. The Oracle Communications Session Border Controller caches the IRS. The user identities that comprise the cached IRS are used for validation later.

As the Oracle Communications Session Border Controller receive a UE's INVITE, the value in the P-Preferred-Identity header is validated against the public user identities in the cached IRS. If the inbound P-Preferred-Identity matches any items in the IRS, the Oracle Communications Session Border Controller inserts that value into a P-Asserted-Identity header in the egress message. The P-Preferred-Identity header is stripped from the outbound message.

Inbound INVITE Contains:	Validate Against Implicit Registration Set	Outbound Invite Created With: (all P-Preferred-Identity headers are removed)
P-Preferred-Identity: value-1	value-1 present	P-Asserted-Identity: value-1

The Oracle Communications Session Border Controller can create a second P-Asserted-Identity header by configuring the add-second-pai option in the SIP config. Once a combination of SIP URIs and/or TEL URIs in the inbound P-Preferred-Identity header are validated against the IRS, the Oracle Communications Session Border Controller forwards the emergency INVITEs with two P-Asserted-Identity headers to the E-CSCF. The behavior is based upon the URI type received in the P-Preferred-Identity header and whether those values validate against the IRS list.

Inbound INVITE Contains:	Validate Against Implicit Registration Set	Outbound Invite Created With: (all P-Preferred-Identity headers are removed)
P-Preferred-Identity: SIP-URI-1	SIP-URI-1 present TEL-URI-1 present (TEL-URI-n present)	P-Asserted-Identity: SIP-URI-1 P-Asserted-Identity: TEL-URI-1
P-Preferred-Identity: TEL-URI-1	TEL-URI-1 present SIP-URI-1 present (SIP-URI-n present)	P-Asserted-Identity: TEL-URI-1 P-Asserted-Identity: SIP-URI-1
NO P-Preferred-Identity:	(SIP   TEL) URI -1 present (SIP   TEL) URI -n present	P-Asserted-Identity: 1st public identity from IRS
P-Preferred-Identity: SIP-URI-1 P-Preferred-Identity: TEL-URI-1	TEL-URI-1 present SIP-URI-1 present	P-Asserted-ID: SIP-URI-1 P-Asserted-ID: TEL-URI-1



Inbound INVITE Contains:	Validate Against Implicit Registration Set	Outbound Invite Created With: (all P-Preferred-Identity headers are removed)
P-Preferred-Identity: SIP-URI-1 P-Preferred-Identity: SIP-URI-2	SIP-URI-1 present SIP-URI-2 present TEL-URI-n present	P-Asserted-Identity: SIP-URI-1 P-Asserted-Identity: 1st TEL-URI from IRS
P-Preferred-Identity: TEL-URI-1 P-Preferred-Identity: TEL-URI-2	TEL-URI-1 present TEL-URI-2 present SIP-URI-n present	P-Asserted-Identity: TEL-URI-1 P-Asserted-Identity: 1st SIP-URI from IRS

If the INVITE does not include a P-Preferred-Identity header and does not include a P-Asserted-Identity header, or the value in the original P-Preferred-Identity or P-Asserted-Identity header is not contained in the IRS, or the URI from the From: header is not the in the IRS, then default public user identity is inserted into a P-Asserted-Identity header in the egress message. (The default public user identity is the first on the list of URIs in the P-Associated-URI header.)

If the strict-3gpp-pai-compliance option is configured in the outbound SIP interface, the first P-Asserted-Identity header also includes the display name.

### Two Incoming P-Asserted-Identity Headers

When the inbound INVITE contains 2 P-Asserted-Identity headers, the Oracle Communications Session Border Controller ensures that both outbound P-Asserted-Identity headers contain public user identities from the IRS according to the following:

Inbound Invite Contains	Validate Against Implicit Registration Set:	Outbound Invite Created With:
P-Asserted-Identity: SIP-URI-1 P-Asserted-Identity: TEL-URI-1	SIP-URI-1 and TEL-URI-1 present	P-Asserted-Identity: SIP-URI-1 or TEL-URI-1 P-Asserted-Identity: 1st public identity from IRS other than value in 1st P-Asserted-Identity
P-Asserted-Identity: SIP-URI-1 P-Asserted-Identity: SIP-URI-2	SIP-URI-1 or SIP-URI-2 present TEL-URI-1 present	P-Asserted-Identity: SIP-URI-1 P-Asserted-Identity: TEL-URI-1
P-Asserted-Identity: TEL-URI-1 P-Asserted-Identity: TEL-URI-2	TEL-URI-1 or TEL-URI-2 present SIP-URI-1 present	P-Asserted-Identity: TEL-URI-1 or TEL-URI-2 P-Asserted-Identity: SIP-URI-1

1. Navigate to the sip-config configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

2. Type select to begin configuring this object.

```
ACMEPACKET(sip-config)# select
ACMEPACKET(sip-config)#
```

3. options—Configure the add-second-pai option:

## IMS Support

```
ACMEPACKET(sip-config)# options +add-second-pai
ACMEPACKET(sip-config)#
```

4. Save and activate your configuration.

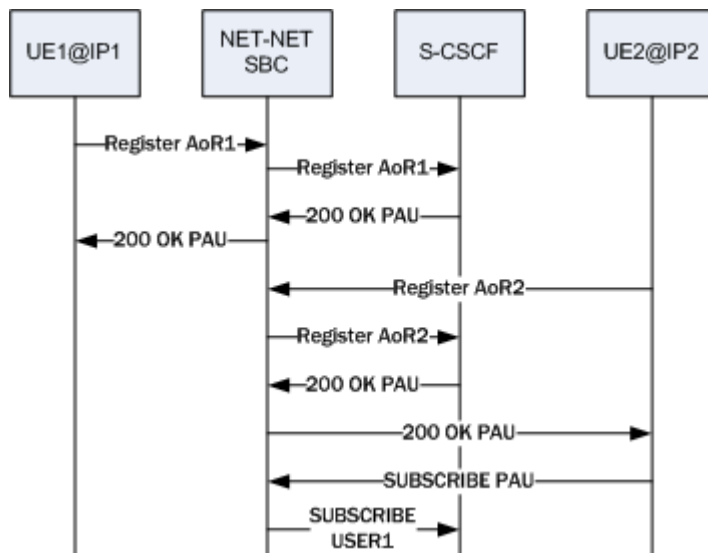
## Temporary Public User Identities and Multi-SIM Scenarios

The Oracle Communications Session Border Controller's SIP interface supports multiple registered users for the same P-Asserted-Uri (PAU), a useful ability for multi-SIM scenarios. The call flow for this type of scenario differs depending on whether or not you configure the SIP interface facing the UE with the **reg-via-key** and **reg-via-match** options.

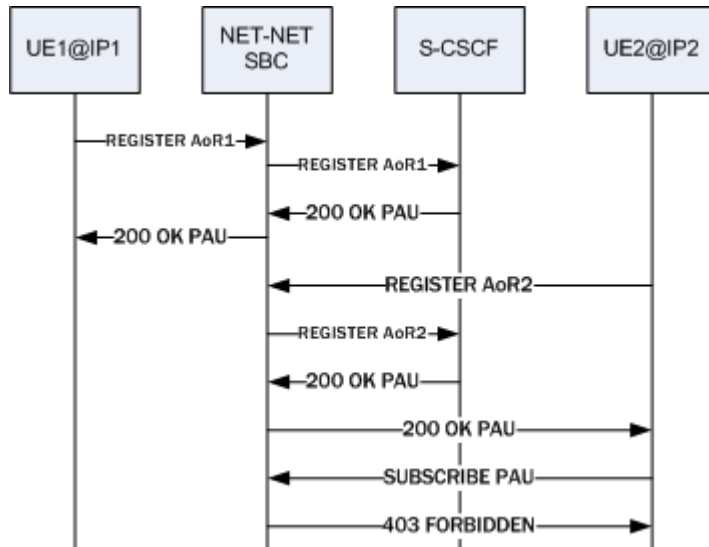
In a multi-SIM scenario, the UE derives a temporary IMS public identity (IMPU); that UE then registers with the Oracle Communications Session Border Controller using the IMPU as the address of record (AoR) from a unique IP. The S-CSCF returns a PAU in the 200 OK, which the Oracle Communications Session Border Controller caches. The UE then derives another IMPU; it registers with the Oracle Communications Session Border Controller using that IMPU as the AoR from another unique IP. The S-CSCF again returns the same PAU in the 200OK.

### Old Behavior

Before the introduction of this change, the Oracle Communications Session Border Controller associated the PAU only with the first IMPU request. The Oracle Communications Session Border Controller considered any request made from that PAU to be a request from the first user, regardless of the request's originating IP.

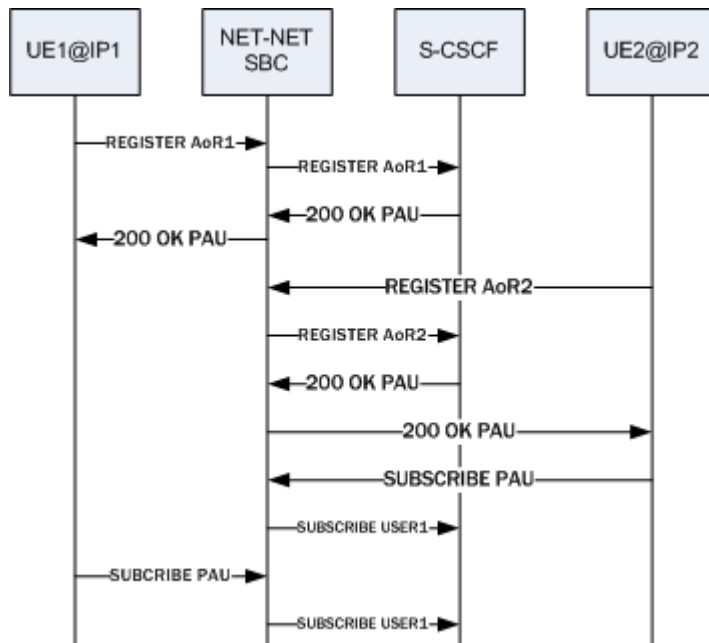


When the SIP interface facing the UE had the **reg-via-key** and **reg-via-match** options configured, the Oracle Communications Session Border Controller rejected the request from the second user with the PAU as the From or the PPI. Because it only associates the PAU with the first user, the Oracle Communications Session Border Controller issued a 403 message.

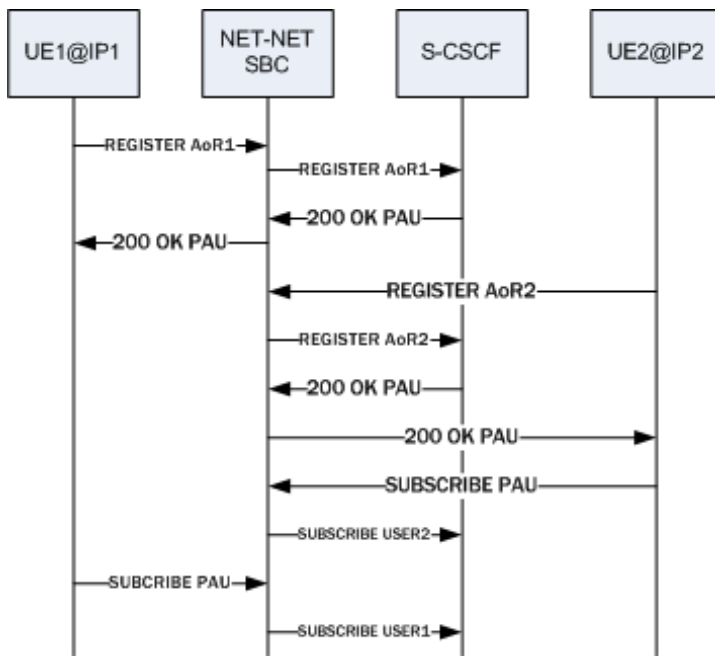


### New Behavior

Your Oracle Communications Session Border Controller now associates the PAU with both the first and second IMPU. The Oracle Communications Session Border Controller considers any request made from that PAU to be a request from the user at the top of its registration cache table irrespective of where the request originated (the IP).



When the SIP interface facing the UE has the **reg-via-key** and **reg-via-match** options configured, and the request from the user with the PAU as the From or with PPI, the Oracle Communications Session Border Controller matches to the proper user based on the source IP.



### Configuring SIP Interface with reg-via-key and reg-via-match

If you do not want to use the call scenario associated with the SIP interface options in the New Behavior section, you do not need to make any change to your configuration.

If you want your call scenarios to resemble the one associated with the SIP interface options in the New Behavior section, then you need to configure **reg-via-key** and **reg-via-match** options on the SIP interface facing the UE.

To configure a SIP interface with the reg-via-key and reg-via-match options:

1. In Superuser mode, type configure terminal and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type session-router and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type sip-interface and press Enter.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-interface)#
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the ACLI **select** command) the configuration that you want to edit.

4. **options**—Set the options parameter by typing options, a Space, and then the option name. Then press Enter.

```
ACMEPACKET(sip-interface)# options +reg-via-key
ACMEPACKET(sip-interface)# options +reg-via-match
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

5. Save your work.

## Surrogate Registration

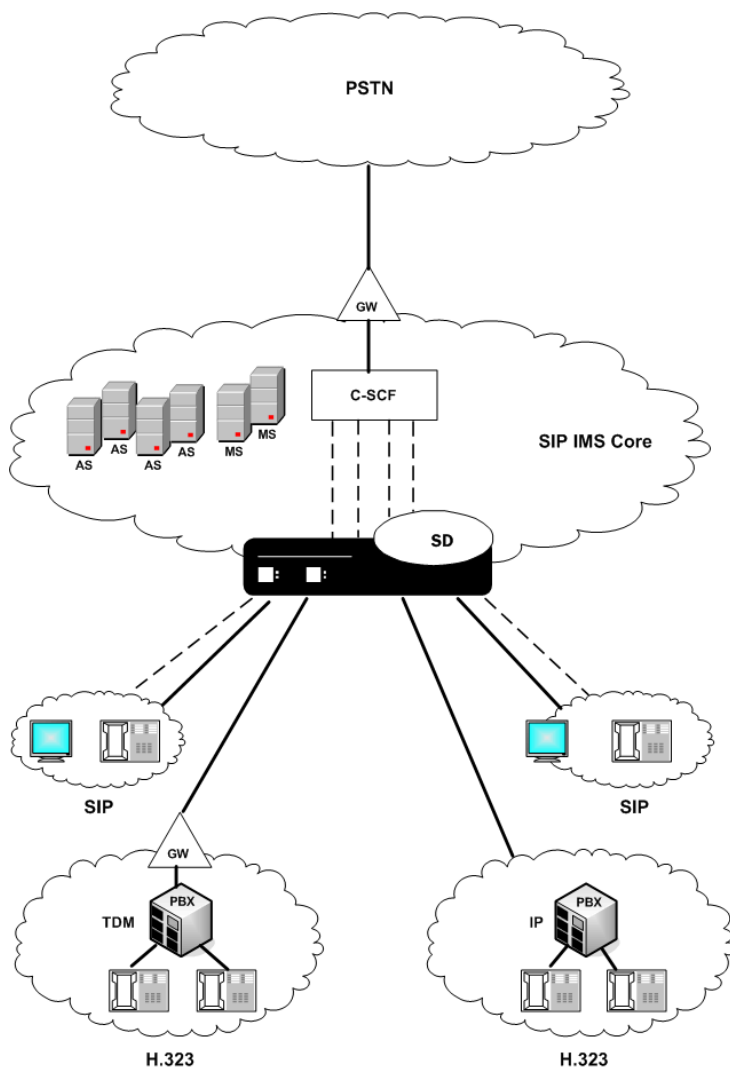
The Oracle Communications Session Border Controller surrogate registration feature lets the Oracle Communications Session Border Controller explicitly register on behalf of a Internet Protocol Private Branch Exchange (IP-PBX). After you configure a surrogate agent, the Oracle Communications Session Border Controller periodically generates a REGISTER request and authenticates itself using a locally configured username and password, with the Oracle Communications Session Border Controller as the contact address. Surrogate registration also manages the routing of class from the IP-PBX to the core and from the core to the IP-PBX.

## Integrating with IMS

With surrogate registration, the Oracle Communications Session Border Controller lets IP-PBXes integrate with the IP Multimedia Subsystem (IMS) architecture. The IP-PBX registers itself as if it were user equipment (UE), which triggers the implicit registration of all phone numbers associated with the IP-PBX.

Implicit registration means the explicit registration of one address of record (AoR) triggers the implicit registration of all the other AoRs associated with that UE. The implicitly registered AoRs are passed back to the UE as P-Associated-URIs in the registration's 200 (OK).

IMS assumes that each SIP endpoint can register itself with its Serving-CSCF (S-CSCF). However, phones can be connected to SIP Integrated Access Devices (IADs) or SIP or H.323 IP-PBXes. The Oracle Communications Session Border Controller performs SIP registration on behalf of the IP-PBX and IADs.



The Oracle Communications Session Border Controller registers on behalf of the IP-PBXes and then stores the associated URIs returned by the Serving Call Session Control Function (S-CSCF). The calls from the phones behind the IP-PBX can be routed based on the cache entry the Oracle Communications Session Border Controller creates after it receives each phone's associated URI. Calls are routed using the service route, local policy or any other routing mechanism based on the associated session agent or session agent group. The Oracle Communications Session Border Controller also supports multiple registrations on behalf of a IP-PBX because the IP-PBX can support thousands of phones, but the registrar might only be able to send 10 to 20 associated URIs in response to a single registration.

The Oracle Communications Session Border Controller replaces the Contact URI for requests from the IP-PBX to the core to match the registered value. For calls from the IMS core to the IP-PBX, the Oracle Communications Session Border Controller replaces the Request-URI username with P-Called-Party-ID/To-URI username. The IMS cores sends INVITES for the phones behind the IP-PBX with the registered Contact URI as the Request-URI instead of the AoR of the phones. The IP-PBX needs to see the phone's AoR in the Request-URI.

### Registration

The Oracle Communications Session Border Controller uses the configuration information of the surrogate agent that corresponds to a specific IP-PBX. After the surrogate agents are loaded, the Oracle Communications Session Border Controller starts sending the REGISTER requests on their behalf. (You can configure how many requests are sent.)

The SIP surrogate agent supports the ability to cache authorization or proxy-authorization header values from a REGISTER 401, 407, and 200 OK messages and reuse it in subsequent requests, such as in an INVITE. This allows the Oracle Communications Session Delivery Manager to support authorization of subsequent requests in cases such as, when a customer PBX doesn't support registration and authentication. It also supports the generation of authorization/proxy-authorization header if subsequent requests get challenged with a 401/407 response.

If the Oracle Communications Session Border Controller receives 401 or 407 responses to REGISTER, requests, it will use the Message Digest algorithm 5 (MD5) digest authentication to generate the authentication information. You need to specify the password. The Oracle Communications Session Border Controller also supports authentication challenge responses with the quality of protection code set to auth (qop=auth), by supporting the client nonce (cnonce) and nonce count parameters.

The Oracle Communications Session Border Controller creates a registration cache entry for each of the AoRs for which it is sending the REGISTER requests. When the Oracle Communications Session Border Controller receives the associated URIs, it checks whether the customer host parameter is configured. If it is configured, the Oracle Communications Session Border Controller changes the host in the received Associated-URI to the customer host. If it is not configured, the Oracle Communications Session Border Controller does not change the Associated-URI. It makes the registration cache entries that correspond to each of the Associated-URIs. The From header in the INVITE for calls coming from the IP-PBX should have one of the Associated-URIs (URI for a specific phone). If the Oracle Communications Session Border Controller receives a Service-Route in the 200 (OK) response, it stores that as well.

The Oracle Communications Session Border Controller uses the expire value configured for the REGISTER requests. When it receives a different expire value in the 200 OK response to the registration, it stores the value and continues sending the REGISTER requests once half the expiry time has elapsed.

REGISTER requests are routed to the registrar based on the configuration. The Oracle Communications Session Border Controller can use the local policy, registrar host and the SIP configuration's registrar port for routing.

If the Oracle Communications Session Border Controller is generating more than one register on behalf of the IP-PBX, the user part of the AoR is incremented by 1 and the register contact-user parameter will also be incremented by 1. For example, if you configure the register-user parameter as caller, the Oracle Communications Session Border Controller uses caller, caller1, caller2 and so on as the AoR user.

### Routing Calls from the IMS Core

The calls coming from the core will have the Oracle Communications Session Border Controller's Contact-URI (which is sent in the REGISTER request) as the Request-URI. The Oracle Communications Session Border Controller looks for a registration entry that corresponds to this URI. After finding the registration entry and the corresponding surrogate agent, the Oracle Communications Session Border Controller looks for the routing

mechanism it should use to route this INVITE to the IP-PBX. It uses the customer-next-hop configuration parameter to determine if it routes this call to the session agent, the session agent group, or directly to a particular IP address.

## SIP

If the customer-next-hop parameter points to a SIP session agent or the SIP session agent group, the Oracle Communications Session Border Controller creates a Route header using the session agent and modifies the Request-URI. It changes the user portion of the Request-URI to either the user portion of the P-Called-Party-ID header, if present, or to the user portion of the To header. The Oracle Communications Session Border Controller also changes the host portion of the Request-URI to the hostname configured in the customer-host configuration parameter. It makes the change because the domain name on the core side can be different than the domain name on the access IP-PBX side. The Oracle Communications Session Border Controller then uses the added Route header to properly route the call.

## H.323

If the session agent or the session agent group configured for the customer-next-hop parameter references an H.323 device, the Oracle Communications Session Border Controller sends the INVITE to its interworking task. If a session agent group is being used, the parameter containing the session agent group name is added to the Request-URI. The host portion of the Request-URI will point to the interworking IP address and the port is changed to 1720.

If a session agent is used, the Oracle Communications Session Border Controller uses it to route the call properly to the interworking task to take care of the H.323 call setup.



**Note:** Even though the customer-next-hop field allows specification of a SAG or FQDN, the functionality will only support these values if they resolve to a single IP address. Multiple IP addresses, via SAG, NAPTR, SRV, or DNS record lookup, are not allowed.

## Routing Calls from the IP-PBX

The Oracle Communications Session Border Controller looks for a match in the registration cache based on the From header or the P-Preferred-Identity header. The header should contain the user portion of one of the Associated-URIs that it received from the registrar in the 200 (OK) responses to REGISTER requests. It should also have the same hostname that is configured in the customer-host parameter. If that parameter is not configured, then the hostname should be same as the one configured for the register-host parameter.

With an H.323 IP-PBX, when the Oracle Communications Session Border Controller receives an INVITE from the interworking task it looks to see if the call is coming from a session agent. If it is, it looks to see if this session agent has a surrogate agent configured. If it does, the Oracle Communications Session Border Controller changes the host portion of the From header to match the registration entry stored in the registration cache.

After the corresponding registration Service-Router entry is found, the Oracle Communications Session Border Controller uses the Service-Route for this endpoint to route the call, if it exists. If no Service-Route exists but the SIP interface's route-to-registrar parameter is enabled, the Oracle Communications Session Border Controller tries to route this to the registrar. You can configure the surrogate agent to override the SIP interface's route-to-register setting. If the surrogate agent's route-to-register parameter is set to disable, it takes precedence over the SIP interface's setting. The Oracle Communications Session Border Controller will not try to route the call to the registrar.

## Configuring Surrogate Registration

You can configure surrogate registration using the ACLI. You need to configure a surrogate agent for each IP-PBX proxy for which the Oracle Communications Session Border Controller will be registering. Those parameters that are optional are marked, the rest are mandatory.

To configure the surrogate agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type surrogate-agent and press Enter. The prompt changes to indicate you can configure individual parameters.

```
ACMEPACKET (session-router) # surrogate-agent  
ACMEPACKET (surrogate-agent) #
```

From this point, you can configure surrogate agent parameters. To view all surrogate agent configuration parameters, enter a ? at the system prompt.

4. **register-host**—Enter the registrar’s hostname to be used in the Request-URI of the REGISTER request. This name is also used as the host portion of the AoR To and From headers.
5. **register-user**— Enter the user portion of the AoR (Address of Record).
6. **state**—Set the state of the surrogate agent to indicate whether the surrogate agent is used by the application. The default value is **enabled**. The valid values are:
  - enabled | disabled
7. **realm-id**— Enter the name of realm where the surrogate agent resides (where the IP-PBX proxy resides). There is no default.
8. **description**—Optional. Enter a description of this surrogate agent.
9. **customer-host**—Optional. Enter the domain or IP address of the IP-PBX, which is used to determine whether it is different than the one used by the registrar.
10. **customer-next-hop**—Enter the next hop to this surrogate agent:
  - session agent group:

```
SAG: <session agent group name>
```
  - session agent:  
<hostname> or <IPV4>
  - specific IP address:  
<IPV4> or <IPV4: port>
11. **register-contact-host**—Enter the hostname to be used in the Contact-URI sent in the REGISTER request. This should always point to the Oracle Communications Session Border Controller. If specifying a IP address, use the egress interface’s address. If there is a SIP NAT on the registrar’s side, use the home address in the SIP NAT.
12. **register-contact-user**—Enter the user part of the Contact-URI that the Oracle Communications Session Border Controller generates.
13. **password**—If you are configuring the auth-user parameter, you need to enter the password used in case the registrar sends the 401 or 407 response to the REGISTER request.
14. **register-expires**—Enter the expires in seconds to be used in the REGISTER requests. The default value is **600,000** (1 week). The valid range is:
  - Minimum—0
  - Maximum—999999999
15. **replace-contact**—This specifies whether the Oracle Communications Session Border Controller needs to replace the Contact in the requests coming from the surrogate agent. If this is enabled, Contact will be replaced with the Contact-URI the Oracle Communications Session Border Controller sent in the REGISTER request. The default value is **disabled**. The valid values are:
  - enabled | disabled
16. **route-to-registrar**—This indicates whether requests coming from the surrogate agent should be routed to the registrar if they are not explicitly addressed to the Oracle Communications Session Border Controller. The default value is **enabled**. The valid values are:
  - enabled | disabled
17. **aor-count**—Enter the number of registrations to do on behalf of this IP-PBX. If you enter a value greater than **1**, the Oracle Communications Session Border Controller increments the register-user and the register-contact-



user values by that number. For example, if this count is 3 and register-user is john then users for three different register messages will be john, john1, john2. It does the same for the register-contact-user values. The default value is 1. The valid range is:

- Minimum—0
- Maximum—999999999

**18. options**—Optional. Enter non-standard options or features.

**19. auth-user**—Enter the authentication user name you want to use for the surrogate agent. This name is used when the Oracle Communications Session Border Controller receives a 401 or 407 response to the REGISTER request and has to send the REGISTER request again with the Authorization or Proxy-Authorization header. The name you enter here is used in the Digest username parameter. If you do not enter a name, the Oracle Communications Session Border Controller uses the value of the register-user parameter.

**20. Save and activate your configuration.**

### Example

The following example shows the surrogate agent configuration.

```
surrogate-agent
register-host      acmepacket.com
register-user     234567
state            enabled
realm-id         public
description
customer-host    acmepacket.com
customer-next-hop 111.222.333.444
register-contact-host 111.222.5.678
register-contact-user eng
password
register-expires  600000
replace-contact  disabled
route-to-registrar enabled
aor-count        1
options
auth-user
last-modified-date 2006-05-04 16:01:35
```

## SIP Surrogate Registration Enhancements

For IMS-E networks, enhancements to the Oracle Communications Session Border Controller's SIP surrogate registration capabilities enable it to register a series of endpoints on behalf of a set of devices that are unable to register themselves. In addition, the Oracle Communications Session Border Controller retries failed registrations, prevents authentication loops, and sends an SNMP trap for failed retransmissions. The automatic incrementing of register-user and register-contact-user values are also now more flexible.

### Without Enhancements

Without the enhancements configured, the Oracle Communications Session Border Controller's surrogate agent performs a series of registrations based on count when the system boots or when its configuration changes. It only attempts to register each user once. Although the surrogate agent uses the same retry mechanism used for SIP client transactions, it does not attempt further if it receives a failure response until the entry expires. When it receives 401, 403, or 407 responses to requests that include authentication, the surrogate agent's automatic incrementing mechanism appends a number to the end of each registered username. Always starting at one, this number cannot appear in any other position in the username.

### With Enhancements

With the enhancements configured, the Oracle Communications Session Border Controller supports:

- Registration retry—You can configure the surrogate agent to retry registration when after a failure, timeout, or transport error. You can set how many times the Oracle Communications Session Border Controller will attempt to

register each user; a setting of zero means retries are unlimited. You can also define the number of seconds to wait before initiating a retry. The Oracle Communications Session Border Controller tracks each registration retry count and timers, and sends an SNMP trap when it reaches the maximum number of retries, which signifies failed registration.

- Authentication loop prevention—Authentication loops can occur in previous releases when the Oracle Communications Session Border Controller resends a registration request with authentication in response to 401, 403, or 407 responses (indicating, for example, that there might be a password error). Using the new enhancements, the Oracle Communications Session Border Controller only allows permits the retransmission of one request. It now considers further 401, 403, or 407 responses to be errors and initiates the retry mechanism.
- Automatic increment enhancements—Now, the automatic increment works with the caret (^) in the register-user and register-contact-user fields. These carets define where the automatically generated incrementing number is inserted in the username. You can also use multiple carets to define leading zeroes to insert; for example, the entry user^^^^ will become user0001. You can also define the starting integer for the incrementing registrations. For example, setting the AoR count to 20, the count start to 5, and using the value user^^^^ for register-user and register-contact-user results in the incremented user registrations user0005 through user0025.

### Configuring the Retry Mechanism

To set the retry mechanism:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **surrogate-agent** and press Enter.

```
ACMEPACKET(session-router)# surrogate-agent
ACMEPACKET(surrogate-agent)#
```

4. **max-register-attempts**—Using a value from 0 (meaning registration attempts are unlimited) to 10, enter the number of times you want to attempt registration. The default value is 3. The valid range is:

- Minimum—0
- Maximum—10

5. **register-retry-time**—Enter the amount of time in seconds, between 30 and 3600 seconds, you want the Oracle Communications Session Border Controller to wait before reattempting registration. The default value is 300. The valid range is:

- Minimum—10
- Maximum—3600

6. Save and activate your configuration.

### Configuring the Count Start

To set the value where automatic incrementing will start:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
```

3. Type **surrogate-agent** and press Enter.

```
ACMEPACKET(session-router)# surrogate-agent
ACMEPACKET(surrogate-agent)#
```

4. **count-start**—Change this parameter from its default of 1 if you want the automatic increment count to start at any other number when the Oracle Communications Session Border Controller performs multiple registrations. The default value is 1. The valid range is:
- Minimum—0
  - Maximum—999999999

## SIP-IMS Surrogate Registration Proxy Authorization Header for Non-Register Requests

The Oracle Communications Session Delivery Manager's IMS functionality helps customers who use SIP IP PBX or SIP gateways that can only peer with carriers connected to IMS via a P-CSCF. As part of this function, the Oracle Communications Session Delivery Manager provides for generating a Proxy-Authorization or Authorization header for REGISTER requests that are challenged. This feature extends the Oracle Communications Session Delivery Manager's capabilities by also allowing you to configure the generation of Proxy-Authorization and Authorization headers for non-REGISTER requests.

When you configured it to do so, the Oracle Communications Session Delivery Manager caches Proxy-Authorization or Authorization headers from the most recent (last-sent) messages in the following exchange: REGISTER--407 Proxy Authentication Required--REGISTER--200. Then the system uses these values in the subsequent requests. The following methods are supported:

- INVITE
- ACK
- BYE
- CANCEL
- UPDATE
- INFO
- PRACK
- OPTIONS


The Oracle Communications Session Delivery Manager updates the following parameters when it generates the header:

- **nonce-count**—Incremented for every new request the Oracle Communications Session Delivery Manager receives
- **response**—Contains the digest-request, newly generated using the cnonce, nonce, and other fields as input

In addition, the system supports the nonce text parameter in the Authentication-Info header. And for surrogate registration, it recognizes the Authentication-Info header in 200 OK responses received from the UAS and updates its cached nonce value accordingly; in this case, the system resets the nonce count to 1 for the subsequent request.

## SIP-IMS Surrogate Registration Proxy Authorization Header Configuration

You configure the SIP-IMS surrogate registration proxy authorization header for non-register requests by setting the **options** parameter in the surrogate agent configuration. You set two types of options: **auth-methods** and **auth-info**.

 **Note:** If authentication of any SIP requests other than REGISTER is required, then the surrogate-agent option **auth-methods** MUST be configured. Supported methods are INVITE, ACK, BYE, CANCEL, UPDATE, INFO, PRACK, OPTIONS. For example:

```
ACMEPACKET (surrogate-agent) # options +auth-methods=' INVITE,OPTIONS'
```

To enable SIP-IMS Surrogate Registration Proxy Authorization Header for Non-Register Requests:

1. In Superuser mode, type configure terminal and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **surrogate-agent** and press Enter.

```
ACMEPACKET (session-router) # surrogate-agent
ACMEPACKET (surrogate-agent) #
```

If you are adding support for this feature to a pre-existing configuration, then you must select (using the ACLI **select** command) the configuration that you want to edit.

4. **options**—Set the options parameter by typing options, a Space, and then the option name. Then type the equal sign (=), open quotation mark (“), the list of methods you want supported separated by commas, and closed quotation mark (”). Then press Enter. Valid values are INVITE, ACK, BYE, CANCEL, UPDATE, INFO, PRACK, OPTIONS. Default is blank.

```
ACMEPACKET (surrogate-agent) # options +auth-methods=' INVITE ,CANCEL ,ACK ,BYE'
ACMEPACKET (surrogate-agent) # options +auth-info=refresh
```

If you type the option without the plus sign, you will overwrite any previously configured options. In order to append the new options to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.

## IMS Implicit Service Route

---

The Oracle Communications Session Border Controller provides implicit service route support in situations where it is deployed between user equipment (UE) and the P-CSCF, and where the IMS core network does not support the Service-Route header.

When this feature is enabled, the Oracle Communications Session Border Controller sends requests to the P-CSCF and does not include the Service-Route header received in the 200 OK response (to a REGISTER message) as Route headers in subsequent requests. The Oracle Communications Session Border Controller also includes a Route header of the P-CSCF address in subsequent requests, and includes the loose-route parameter in the Route header. Because inclusion of the loose-route parameter is not needed in all cases, you can set this feature to strict in the SIP interface configuration.

Recent enhancements address the following issues:

- Even when IMS is disabled for a SIP interface, the Net-Net 4000 caches and uses the Service-Route headers from SIP REGISTER responses received from the REGISTER. Therefore, you must use SIP HMR to remove the Service-Route headers from the response, while having no mechanism to replace the Service-Routes from the REGISTER response with an implicit Service Route.
- In the Net-Net 4000 global SIP configuration, the presence of the option route-registrar-no-service-route sets the behavior for using the Service-Route header in the REGISTER request. The new enhancements greatly simplify the process of determining proper use of the header in both IMS and non-IMS environments.
- You can configure the Net-Net 4000 with an option to keep it from using the Service-Route header for REGISTER requests when sent to an out-of-service session agent. The enhancements make this behavior the default—because otherwise these REGISTER requests fail.

When implicit service route support is enabled, the Oracle Communications Session Border Controller stores the Service Route URIs from the Service-Route headers that are included in 200 OK responses to REGISTER messages. The Service Route URIs are included in the Route headers in subsequent Request messages, except for REGISTER messages.

The Net-Net SBC also supports the ability to keep the loose-route parameter from being included in the implicit Route URI that the Oracle Communications Session Border Controller generates and includes as a Route header in the Request messages.

Once an endpoint registers successfully, the Oracle Communications Session Border Controller caches the Service-Route header (if any) to use for routing all subsequent requests from the endpoint—with the exception of any subsequent REGISTER requests.

You can set whether or not you want the Oracle Communications Session Border Controller to route subsequent REGISTER requests using the cached Service Route, and whether the endpoint is engaged in an active session through the Oracle Communications Session Border Controller. If you decide not to use the Service Route for endpoints engaged in active sessions, then the Oracle Communications Session Border Controller uses the local policy to make routing decisions.

For endpoints not in found in the Oracle Communications Session Border Controller's registration cache, the Oracle Communications Session Border Controller again uses the local policy to make routing decisions.

## IMS Implicit Service Route Configuration

To configure implicit service route support:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router) # sip-interface
```

4. **implicit-service-route**—To enabled implicit service route support, change this parameter from **disabled** to **enabled**. The default value is **disabled**. Supported values are:

- absent | disabled (the default) | enabled | replace | strict
- absent—An implicit service route to the session agent to which the REGISTER request was sent is constructed when the successful REGISTER response contains no Service-Route headers.
- disabled (default)—Turns off the implicit service route feature; Oracle Communications Session Border Controller constructs service route the Service-Route headers in a successful REGSITER response.
- enabled—Turns on this feature, meaning that an implicit service route to the session agent to which the REGISTER request was sent is inserted in front of Service-Route header in a successful REGISTER response; the inserted URI includes the ;lr parameter if the session agent has loose routing enabled.
- replace—An implicit service route to the session agent to which the REGISTER request was sent is used to construct the service route. The Oracle Communications Session Border Controller ignores Service-Route headers in successful REGISTER responses.
- strict—An implicit service route to the session agent to which the REGISTER request was sent is inserted in front of Service-Route header in a successful REGISTER response; the inserted URI does not the ;lr parameter if the session agent has loose routing enabled, overriding the loose routing behavior configured for the session agent.

Save and activate your configuration.

## IMS Service Route Configuration

To configure how you want the Service Route used:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) #
```

4. **register-service-route**—Enter the way you want the Oracle Communications Session Border Controller to use the service route:

- **always**—The Oracle Communications Session Border Controller always uses the cached service route for an endpoint for routing REGISTER requests.
- **never**—The Oracle Communications Session Border Controller never uses the service route, and makes routing decisions based on local policies instead.
- **removal**—The Oracle Communications Session Border Controller uses the cached service route for an endpoint when routing REGISTER requests that remove the endpoint's contact. It uses the local policy for refresh and query REGISTER requests.
- **session**—The Oracle Communications Session Border Controller uses the cached service route when routing REGISTER requests that appear while an endpoint has an active session traversing it. When an endpoint does not have an active session, the Oracle Communications Session Border Controller uses the local policy to make routing decisions.
- **session+removal**—Combining the session and removal values, the Oracle Communications Session Border Controller uses the cached service route: when routing REGISTER requests that remove the endpoint's contact and when REGISTER requests appear while while an endpoint has an active session traversing the system. Otherwise, the Oracle Communications Session Border Controller uses the local policy to make routing decisions.

5. Save and activate your configuration.

### Notes About Upgrading

There are Oracle Communications Session Border Controllers currently deployed that use the `route-registrar-no-service-route` option, and these enhancements provide for backward compatibility.

When you upgrade to a release that has the new `register-service-route` parameter in the SIP configuration, the system checks for the presence of the `route-registrar-no-service-route` option. If the system finds the option, then it translates the value configured for the option like this:

Old <code>route-registrar-no-service-route</code> value	New <code>register-service-route</code> value
<empty> or all	never
refresh	removal
all, idle	session
refresh; idle	session+removal

You must save your configuration for these changes to take place, allowing you to fall back to the previous software image.

### IMS Charging Vector Mode Adaptation

---

This adaptation to the Oracle Communications Session Border Controller's IMS functionality provides the ability to remove the P-Charging-Vector from incoming requests for a session and store it. Then the Oracle Communications Session Border Controller inserts it into outbound responses related to that session in a P-Charging-Vector header.

### IMS Charging Vector Mode Configuration

Typically, the ACLI `charging-vector-mode` parameter is set to `delete-and-respond` (which supports removing and storing the P-Charging-Vector for later insertion in outbound response) on the core, trusted interface. On the access, untrusted side, this same parameter is set to `insert`.

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type `session-router` and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter. If you are adding this feature to a pre-existing SIP interface, you need to select and edit that configuration.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. **charging-vector-mode**—Change this parameter to **delete-and-respond** to remove the P-Charging-Vector from incoming requests for a session and store it. Then the Oracle Communications Session Border Controller inserts it into outbound responses related to that session in a P-Charging-Vector header.
5. Save and activate your configuration.

## IMS P-CSCF Endpoint Identification Using Address and Port

You can configure the Oracle Communications Session Border Controller, acting as a P-CSCF, to match a Request it receives to a registration cache entry based only on the IP address and port from which the Request came. When you enable this behavior, the Oracle Communications Session Border Controller will perform this kind of endpoint identification even when there nothing in the message matches the cache entry.

### IMS P-CSCF Endpoint Identification Configuration

For this behavior to work as designed, you must also have the **reg-via-key** option enabled for the SIP interface to which you are adding the **reg-via-match** option.

To configure P-CSCF endpoint identification using address and port:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter. If you are editing an existing configuration, select the one on which you want to enable this feature.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. **options**—Set the options parameter by typing **options**, a Space, the option name **reg-via-match** with a plus sign in front of it, and then press Enter.

```
ACMEPACKET(sip-interface)# options +reg-via-match
```

**If you type options and then the option value for either of these entries without the plus sign, you will overwrite any previously configured options. In order to append the new option to this configuration's options list, you must prepend the new option with a plus sign as shown in the previous example.**

5. Save and activate your configuration.

## IMS-AKA

The Oracle Communications Session Border Controller supports IP Media Subsystem-Authentication and Key Agreement (IMS-AKA).

Defined in 3GPP7 (specifications in TS 33.203 and call flows in TS 24.228), IMS-AKA can be used as a framework for authentication and for securing the signaling path between a UE and the Oracle Communications Session Border

Controller (when the Oracle Communications Session Border Controller is acting as a P-CSCF or as a B2BUA) across the Gm interface.

In addition, the Oracle Communications Session Border Controller's serving as an IMS-AKA termination point is valuable because it allows IMS-AKA use behind by multiple endpoints sitting behind a NAT device. IMS-AKA support also works when there are no NAT devices between endpoints and the Oracle Communications Session Border Controller acting as a P-CSCF, and when the Oracle Communications Session Border Controller sits behind a third-party P-CSCF. In addition, you can use IMS-AKA when the endpoint uses SIP UDP.

## Requirements

IMS-AKA use assumes that you have installed the appropriate IPsec module on your Oracle Communications Session Border Controller, or that it has come from Oracle with those modules pre-installed. IMS-AKA will not work without this hardware.

In addition, your configuration must have SIP registration caching enabled.

## The refreshRegForward Option

The Oracle Communications Session Border Controller provides a the user with a means of ignoring its registration refresh half-life timer, and send all applicable registration refreshes received via IMS-AKA to the core for authentication.

By default, the Oracle Communications Session Border Controller uses its half-life function and attempts to manage registration refreshes prior to half-life expiry without forwarding the refresh to the core. The Oracle Communications Session Border Controller sends registration refreshes that arrive after the half-life expiry to the core.

The user changes this behavior by setting the **refreshRegForward** in the applicable IMS-AKA profile to as follows.

```
ACMEPACKET (ims-aka-profile) # options +refreshRegForward
```

When this option is set, the system forwards every refresh registration to the IMS core regardless of the half-life timer's status.

## Monitoring

The ACLI **show sipd endpoint-ip** command is updated to show the IMS-AKA parameters corresponding to each endpoint. The display shows the algorithms used, the ports used, and the security parameter indexes (SPIs) used.

In addition, the **show sa stats** command now shows the security associations information for IMS-AKA.

## ACLI Instructions and Examples

You enable IMS-AKA by configuring the following:

- An IMS-AKA profile
- Certain parameters in the global IPsec configuration
- Certain parameters in the SIP interface, and in the SIP interface's SIP port

### Setting Up an IMS-AKA Profile

An IMS-AKA profile establishes the client and server ports to be protected, and it defines lists of encryption and authentication algorithms the profile supports. You can configure multiple IMS-AKA profiles, which are uniquely identified by their names.

You apply an IMS-AKA profile to a SIP port configuration using the name.

To configure an IMS-AKA profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```



2. Type **security** and press Enter.

```
ACMEPACKET (configure) # security
ACMEPACKET (security) #
```

3. Type **ims-aka-profile** and press Enter.

```
ACMEPACKET (system) # ims-aka-profile
ACMEPACKET (ims-aka-profile) #
```

4. **name**—Enter the name you want to give this IMS-AKA profile. This is the value you will use to apply the profile to a SIP port configuration. This parameter is required, and it has no default value.
5. **protected-server-port**—Enter the port number of the protected server port, which is the port on which the Oracle Communications Session Border Controller receives protected messages. The protected server port should not overlap with the port range defined in the steering ports configuration using the same IP address and the SIP interface. If there is overlap, the NAT table entry for the steering port used in a call will prevent SIP messages from reaching the system's host processor.

This parameter defaults to 0, which disables the function associated with the parameter. The valid range for values is 1025 to 65535.

6. **protected-client-port**—Enter the port number of the protected client port, which is the port on which the Oracle Communications Session Border Controller sends out protected messages. Like the protected server port, the protected client port should not overlap with the port range defined in the steering ports configuration using the same IP address and the SIP interface. If there is overlap, the NAT table entry for the steering port used in a call will prevent SIP messages from reaching the system's host processor.

This parameter defaults to 0, which disables the function associated with the parameter. The valid range for values is 1025 to 65535.

7. **encr-alg-list**—Enter the list of encryption algorithms. You enter more than one value by separating the algorithms by <Spaces> and enclosing all values in quotations marks:

```
ACMEPACKET (ims-aka-profile) # encr-alg-list "aes-cbc null"
```

This parameter defaults to the following three values: **aes-cbc**, **des-ede3-cbc**, and **null**.

8. **auth-alg-list**—Enter the list of authentication algorithms. You enter more than one value by separating the algorithms by <Spaces> and enclosing all values in quotations marks:

```
ACMEPACKET (ims-aka-profile) # auth-alg-list "hmac-sha-1-96 hmac-md5-96"
```

This parameter defaults to **hmac-sha-1-96**.

## Setting Up an IPSec Profile for IMS-AKA Use

Using the global IPSec configuration, you establish the parameters governing system-wide IPSec functions and behavior. This configuration also contains parameters required for IMS-AKA support. The IPSec global configuration is a single instance element, meaning there is one for the whole system.

To configure the global IPSec parameters that apply to IMS-AKA:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET # configure terminal
ACMEPACKET (configure) #
```

2. Type **security** and press Enter.

```
ACMEPACKET (configure) # security
ACMEPACKET (security) #
```

3. Type **ipsec** and press Enter.

```
ACMEPACKET (system) # ipsec
ACMEPACKET (ipsec) #
```

4. Type **ipsec-global-config** and press Enter. If you are editing a pre-existing IPSec global configuration, then you need to select the configuration before attempting to edit it.

```
ACMEPACKET(system)# ipsec-global-config
ACMEPACKET(ipsec-global-config)#
```

5. **red-ipsec-port**—Specify the port on which the Oracle Communications Session Border Controller should listen for redundancy IPSec synchronization messages. The default is 1994, and valid values are in the range from 1025 to 65535.
6. **red-max-trans**—Enter the maximum number of redundancy transactions to retain on the active. The default is 10000, and valid values range up to a 999999999 maximum.
7. **red-sync-start-time**—Enter the time in milliseconds before the system starts to send redundancy synchronization requests. The default is 5000, and valid values range up to a 999999999 maximum.
8. **red-sync-comp-time**—Enter the time in milliseconds to define the timeout for subsequent synchronization requests once redundancy synchronization has completed. The default is 1000, and valid values range up to a 999999999 maximum.

### Enabling IMS-AKA Support for a SIP Interface

To enable IMS-AKA for a SIP interface, you must set the **ims-aka-feature** parameter to enabled.

To enable IMS-AKA for a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter. If you are adding this feature to a pre-existing SIP interface, you need to select and edit that configuration.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. **ims-aka-feature**—Change this parameter to **enabled** if you want to use IMS-AKA on this SIP interface. By default, this parameter is **disabled**.

### Applying an IMS-AKA Profile to a SIP Port

The final step in setting up IMS-AKA support is to apply an IMS-AKA profile to a SIP port. Enter the **name** value from the IMS-AKA profile you want to apply in the SIP port's **ims-aka-profile** parameter.

To apply an IMS-AKA profile to a SIP port:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. Type **sip-interface** and press Enter. If you are adding this feature to a pre-existing SIP port, you need to select and edit that configuration.

```
ACMEPACKET(session-interface)# sip-ports
ACMEPACKET(sip-port)#
```

5. **ims-aka-profile**—Enter the **name** value for the IMS-AKA profile configuration you want applied to this SIP port. This parameter has no default.

6. Save and activate your configuration.

## IPSec IMS-AKA

Compliance with the VoLTE specification (GSMA PRD IR.92) requires cluster member support for IPsec IMS-AKA (IP Multimedia Services Authentication and Key Agreement) as defined in 3GPP TS 24.299, *IP Multimedia Call Control Protocol Based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP)*: Stage 3, and TS 33.203, *3G Security: Access Security for IP-based Services*.

Support for IMS-AKA requires no new additional configuration elements.

## Sample IMS-AKA Configuration

The following formatted extract from **show running-config** CLI output shows a sample IMS-AKA profile configuration.

```
ims-aka-profile
name                dut2.test
protected-client-port 4060
protected-server-port 4060
encr-alg-list        aes-cbc des-ede3-cbc null
auth-alg-list        hmac-sha-1-96 hmac-md5-96
last-modified-by     admin@172.30.11.18
last-modified-date   2012-01-10 17:31:59
```

## Sample Security Policy Configuration

The following formatted extracts from **show running-config** CLI output shows three associated security policies.

The first policy, and the one with the highest priority, opens Port 5060 for SIP traffic.

```
security-policy
name                pol1
network-interface   M10:0.6
priority            0
local-ip-addr-match 3fff:c0ac::c0ac:ce12
remote-ip-addr-match ::
local-port-match    5060
remote-port-match   0
trans-protocol-match ALL
direction           both
local-ip-mask        ::
remote-ip-mask       ::
action              allow
ike-sainfo-name
outbound-sa-fine-grained-mask
local-ip-mask        ffff:ffff:ffff:ffff:ffff:ffff:ffff:ffff
remote-ip-mask       ffff:ffff:ffff:ffff:ffff:ffff:ffff:ffff
local-port-mask      65535
remote-port-mask     65535
trans-protocol-mask 0
valid               enabled
vlan-mask            0xFFF
last-modified-by     admin@console
last-modified-date   2012-01-10 17:48:59
```

The second policy opens Port 4444 for CCP traffic.

```
security-policy
name                pol2
network-interface   M10:0.6
priority            2
```

## IMS Support

---

```
local-ip-addr-match      3fff:b623::b623:ce02
remote-ip-addr-match    3fff:b623::b623:ce01
local-port-match        4444
remote-port-match       4444
trans-protocol-match    ALL
direction               both
local-ip-mask           ffff:ffff:ffff:ffff:ffff:ffff:ffff:ffff
remote-ip-mask          ffff:ffff:ffff:ffff:ffff:ffff:ffff:ffff
action                  allow
ike-sainfo-name
outbound-sa-fine-grained-mask
local-ip-mask           ffff:ffff:ffff:ffff:ffff:ffff:ffff:ffff
remote-ip-mask          ffff:ffff:ffff:ffff:ffff:ffff:ffff:ffff
local-port-mask         65535
remote-port-mask        65535
trans-protocol-mask     0
valid                   enabled
vlan-mask               0xFFF
last-modified-by       admin@console
last-modified-date     2012-01-10 17:49:15
```

The third policy, the policy with the least priority, and, consequently, the last policy applied, requires IPsec on all ports.

```
security-policy
name                    pol3
network-interface       M10:0.6
priority                10
local-ip-addr-match    3fff:c0ac::c0ac:ce12
remote-ip-addr-match   ::
local-port-match        0
remote-port-match       0
trans-protocol-match    ALL
direction               both
local-ip-mask           ffff:ffff:ffff:ffff:ffff:ffff:ffff:ffff
remote-ip-mask          ::
action                  ipsec
ike-sainfo-name
outbound-sa-fine-grained-mask
local-ip-mask           ffff:ffff:ffff:ffff:ffff:ffff:ffff:ffff
remote-ip-mask          ffff:ffff:ffff:ffff:ffff:ffff:ffff:ffff
local-port-mask         65535
remote-port-mask        65535
trans-protocol-mask     0
valid                   enabled
vlan-mask               0xFFF
last-modified-by       admin@console
last-modified-date     2012-01-10 17:50:42
```

## Sec-Agree

---

Version S-CZ7.2.0 introduces support for RFC 3329, Security Mechanism Agreement for the Session Initiation Protocol, commonly referred to as Sec-Agree. The RFC defines three SIP headers, Security-Client, Security-Server, and Security-Verify that provide the ability for SIP UAs and other SIP entities (servers, proxies, and registrars) to negotiate next-hop security mechanisms. Note that this initial implementation does not provide support for server-initiated security negotiation, nor does it support media-plane security. That is, support is limited to client-initiated negotiation during initial registration, and to signalling security.

Currently the P-CSCF functionality includes support for IMS-AKA feature for VoLTE deployments. In order to support RCS clients along with VoLTE P-CSCF functionality needs to be enhanced to support RFC 3329, Security Mechanism Agreement for the Session Initiation Protocol (commonly referred to as Sec-Agree), which includes support for TLS as security mechanism.

Sec-Agree defines three SIP headers, Security-Client, Security-Server and Security-Verify, to negotiate security agreements during initial REGISTER transactions. Header definitions are as follows:

```

security-client = "Security-Client" HCOLON
                 sec-mechanism *(COMMA sec-mechanism)
security-server = "Security-Server" HCOLON
                 sec-mechanism *(COMMA sec-mechanism)
security-verify = "Security-Verify" HCOLON
                 sec-mechanism *(COMMA sec-mechanism)
sec-mechanism  = mechanism-name *(SEMI mech-parameters)
mechanism-name = ( "digest" / "tls" / "ipsec-ike" /
                  "ipsec-man" / token )
mech-parameters = ( preference / digest-algorithm /
                   digest-qop / digest-verify / extension )
preference      = "q" EQUAL qvalue
qvalue         = ( "0" [ "." 0*3DIGIT ] ) / ( "1" [ "." 0*3("0") ] )
digest-algorithm = "d-alg" EQUAL token
digest-qop      = "d-qop" EQUAL token
digest-verify   = "d-ver" EQUAL LDQUOTE 32LHEX RDQUOTE
extension       = generic-param

```

The Security-Client header contains one or more security mechanisms and associated parameters proposed by the initiating client. This initial implementation supports two security mechanisms: TLS and ipsec-3gpp.

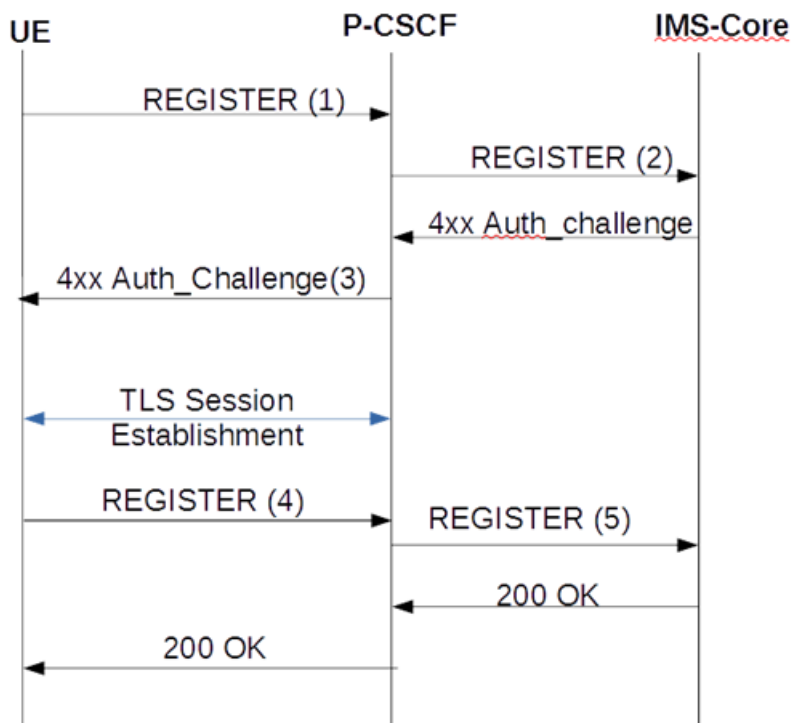
The Security-Server header contains the security mechanism chosen by the server from those mechanisms proposed by the client.

The Security-Verify header contains the contents of the Security-Server header.

Two additional header fields, Require and Proxy-Require, are also used in support of Sec-Agree negotiations. Both headers are required in client transmissions.

## TLS Session Setup During Registration

This call flow depicts a TLS session setup during the registration procedure. Only relevant header fields are noted.



(1) REGISTER

```
Proxy-Require:sec-agree
Security-Client: ipsec-3gpp;alg=hmac-md5-96;ealg=aes-
cbc;prot=esp;mod=trans;spi-c=8765423;port-c=7524;spi-s=1234563;port-s=1358,
ipsec-3gpp;alg= hmac-sha-1-96;ealg=aes-cbc;prot=esp;mod=trans;spi-
c=8765423;port-c=7524;spi-s=1234563;port-s=1358, tls
```

### (2) REGISTER

```
Authorization: Digest
uri="sip:ims.mnc007.mcc262.3gppnetwork.org",username="262073900320132@ims.mnc0
07.mcc262.3gppnetwork.org",response="",realm="ims.mnc007.mcc262.3gppnetwork.or
g",nonce=""
(No integrity-protected field will be present if TLS is selected as the
security mechanism)
```

### (3) 401

```
Security-Server: tls
```

### (4) REGISTER

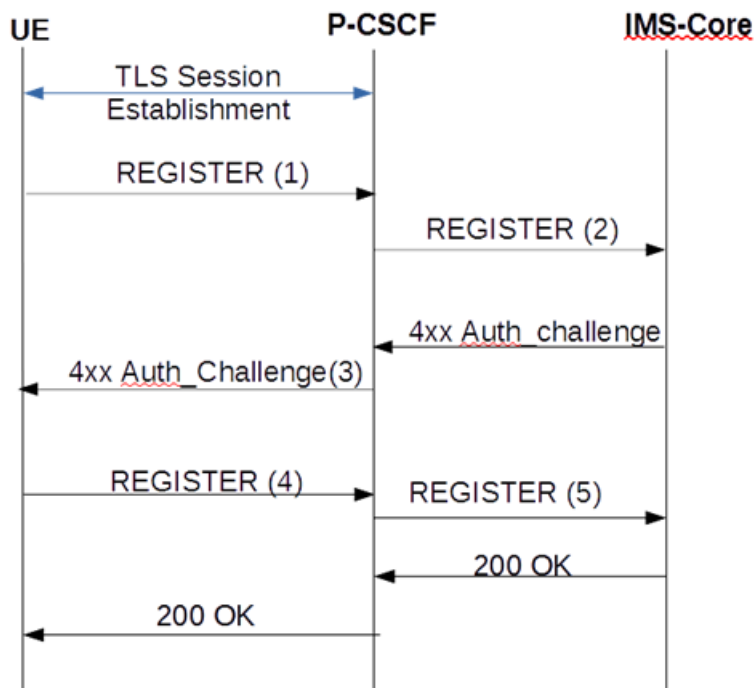
```
Proxy-Require: sec-agree
Security-Verify: tls
```

### (5) REGISTER

```
Authorization: Digest
username="262073900320132@ims.mnc007.mcc262.3gppnetwork.org",realm="imstest1.t
elefonica.de",uri="sip:ims.mnc007.mcc262.3gppnetwork.org",qop=auth,nonce="Ms81
2xeF3141b0VO8fK3KFMSKlv1sQAATdN2NpFUCgU=",nc=00000001,cnonce="3063397945",algo
rithm=AKAv1-MD5,response="3779ff40a057f999a2f8288bbfafc10d", integrity-
protected=tls-pending
```

## TLS Session Setup Prior to Registration

This call flow depicts a TLS session setup prior to the registration procedure.



### (1) REGISTER

```
(No Security-Client or Proxy-Require header present)
```

**(2) REGISTER**

```
Authorization: Digest
uri="sip:ims.mnc007.mcc262.3gppnetwork.org",username="262073900320132@ims.mnc0
07.mcc262.3gppnetwork.org",response="",realm="ims.mnc007.mcc262.3gppnetwork.or
g",nonce=""
(No integrity-protected field will be present)
```

**(3) 401**

```
(No Security-Server header present)
```

**(4) REGISTER**

```
(No Security-Client, or Security-Verify or Proxy-Require header present)
```

**(5) REGISTER**

```
Authorization: Digest
username="262073900320132@ims.mnc007.mcc262.3gppnetwork.org",realm="imstest1.t
elefonica.de",uri="sip:ims.mnc007.mcc262.3gppnetwork.org",qop=auth,nonce="Ms8l
2xeF3l4lb0VO8fK3KFMSKlv1sQAATdN2NpFUCgU=",nc=00000001,cnonce="3063397945",algo
rithm=AKAv1-MD5,response="3779ff40a057f999a2f8288bbfafc10d",integrity-
protected=tls-pending
```

Regardless of TLS Session setup procedure, if the newly added configurable item `sec-agree` feature is enabled, any messages on unprotected port will be rejected except REGISTER messages or messages related to emergency services.

For refresh registration, if the `Sec_Agree` occurred during Registration, it verifies for the presence or change of Security-Client & Security-Verify headers, if they differ it will be rejected with 4xx response and also Authorization header fields are verified irrespective of the above methods and if they differ with previous association it will be rejected with 403 (Forbidden) response. Also when the refresh REGISTER is being forward to the core, it will set the integrity-protected field to "tls-yes".

## SEC-agree Configuration

1. Access the **sip-interface** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

2. Select the **sip-interface** object to edit.

```
ACMEPACKET(sip-interface)# select
<RealmID>:
1: realm01 172.172.30.31:5060

selection: 1
ACMEPACKET(sip-interface)#
```

3. **sec-agree-feature**—Set this parameter to enable or disable for Sec-Agree support. By default, support is disabled.

4. **sec-agree-pref**—Configure this parameter to specify security protocol preferences.

- **ipsec3gpp** — support only IMS-AKA protocol
- **tls** — support only TLS protocol
- **ipsec3gpp-tls** — support both IMS-AKA and TLS, preferred protocol is IMS-AKA
- **tls-ipsec3gpp** — support both TLS and IMS-AKA, preferred protocol is TLS

5. Type **done** to save your configuration.

### IMS AKA over TCP

---

IMS-AKA registration is conducted over UDP or TCP protocol only. The Oracle Communications Session Border Controller supports both transport protocols.

Within mobile IMS VoLTE/RCS deployments, IP packets carrying SIP messages can be large due to IPv6 headers, IMS-AKA specific headers, extensive codec policies, and other 3GPP related headers. Because of this, IPv6 VoLTE signaling messages using IMS-AKA frequently exceed 1300 bytes and require TCP according to RFC3261 section 18.1.1.

### IMS-AKA Secure Call Registration over TCP

To register and place a call into the network, a UE creates 3 TCP connections. The first insecure connection is established to the port (usually 5060) specified in the sip-port for the first registration request. You should create a sip-ports configuration element with port 5060 and an ims-aka-profile parameter that references an ims-aka-profile configuration element. The ims-aka-profile configuration element initiates the process that creates secure connections. For example:

#### sip-ports

```
address          sd-ip-address
port             5060
transport-protocol TCP
tls-profile
multi-home-addr
allow-anonymous  registered
ims-aka-profile  profile
```

The ims-aka-profile configuration element defines the protected-server-port (PORT-SDS) and protected-client-port (PORT-SDC). The protected-server-port is opened for both inbound TCP and UDP traffic. For example:

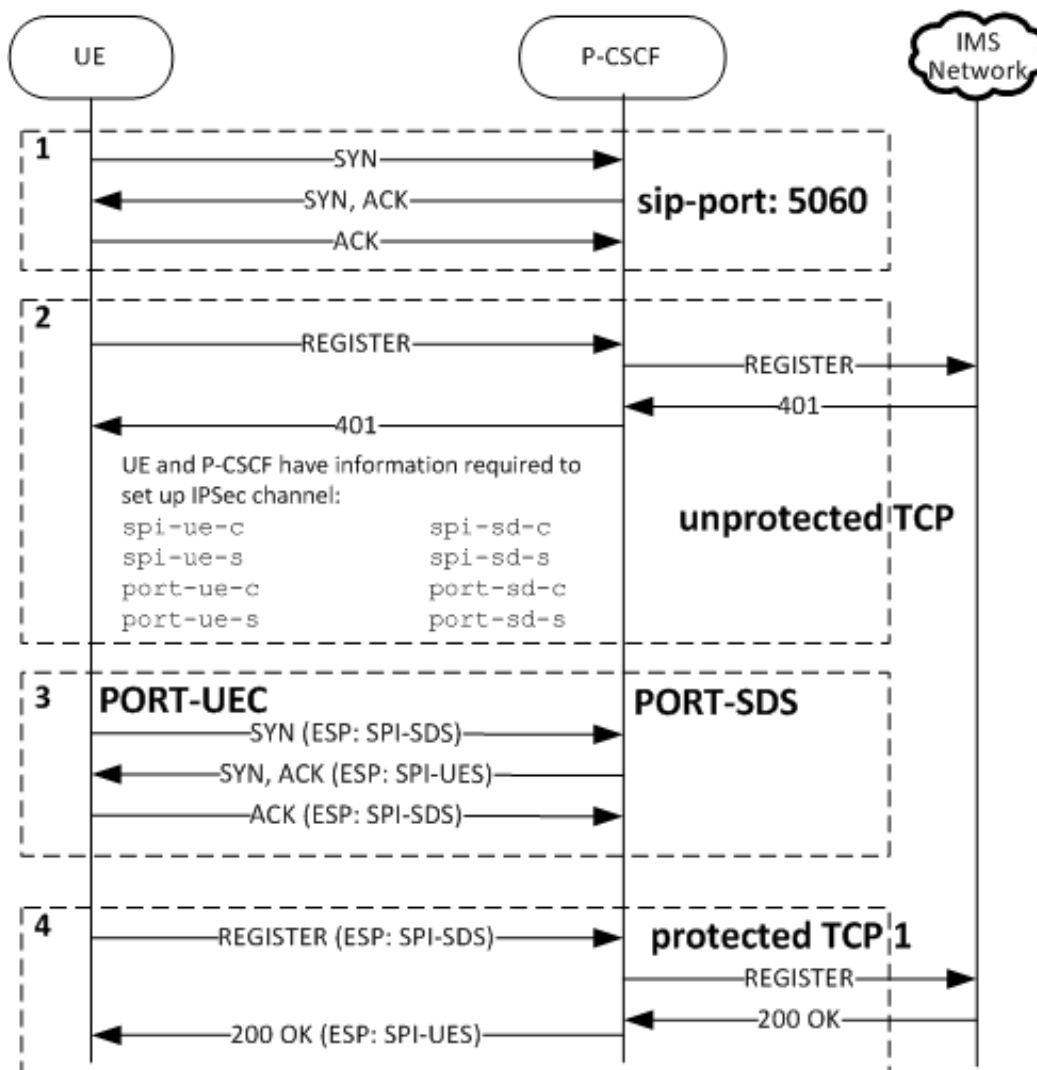
#### ims-aka-profile

```
name             profile
protected-client-port PORT-SDC
protected-server-port PORT-SDS
encr-alg-list    aes-cbc des-ede3-cbc null
auth-alg-list    hmac-sha-1-96
```

When the UE receives the 401Unauthorized challenge from the Oracle Communications Session Border Controller acting as P-CSCF, both devices have the information to set up security association for two IPSec channels. The UE establishes the second TCP connection via IPSec channel from the UE's PORT-UEC to the P-CSCF's PORT-SDS, and the registration process continues.

Hereafter, the UE uses the IPSec channels from communication.





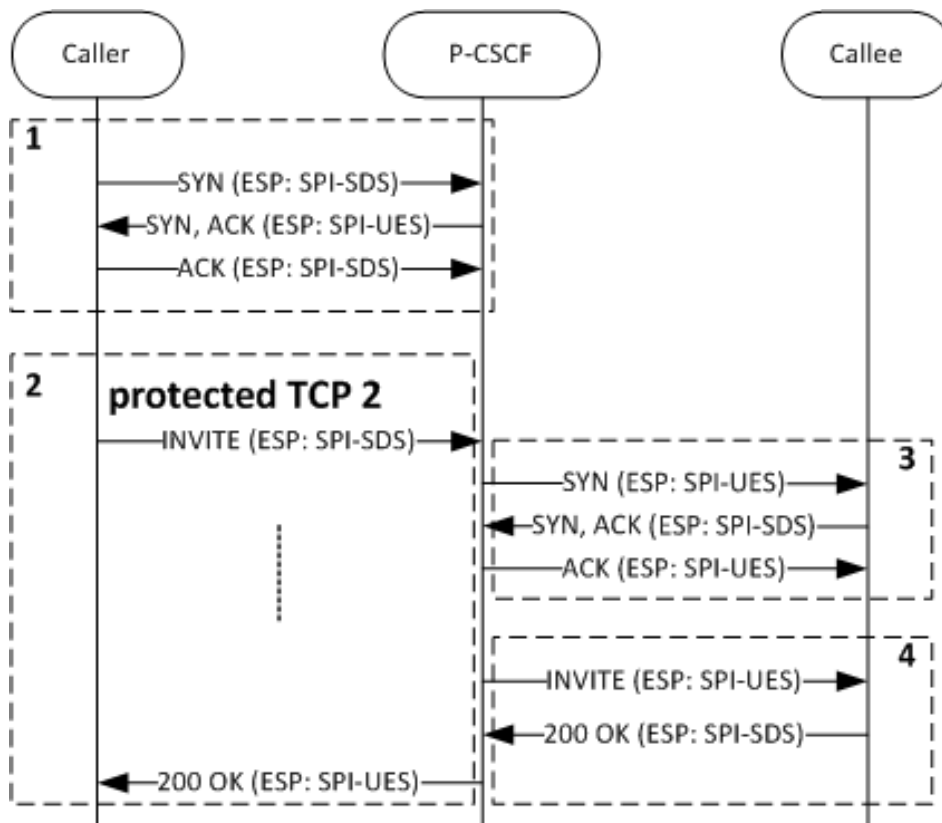
1. The UE and P-CSCF set up the TCP connection.
2. The UE sends an unauthenticated SIP Register message to the P-CSCF's unprotected server port (usually 5060). The Register message is forwarded to the UE's Home S-CSCF. The S-CSCF then replies with a the SIP 401 Authentication Required response back to P-CSCF. This message contains encryption keys and authentication information.

The P-CSCF modifies the 401 message back to UE. At this point, both UE and P-CSCF should have all the information need to establish secure IPsec channels.

3. The UE and P-CSCF create a TCP connection over a secure channel from port-ue-c to port-sd-s.
4. The UE sends an authenticated REGISTER over the secure channel via the P-CSCF to the S-CSCF. If the authentication is valid, the P-CSCF will forward the 200 OK response from the S-CSCF to the UE. The 200 OK response will be sent in the same secure TCP connection.

## IMS-AKA Call Establishment over TCP

For the UE to send a new request into the network and establish a call, a third TCP connection is created using the information configured/generated prior to creating the first secure connection.



1. If the TCP connection over a secure channel does not exist, the Caller will establish it.
2. The Caller initiates the call by sending SIP INVITE from its PORT-UEC to the P-CSCF's PORT-SDS.
3. The P-CSCF forwards the INVITE to the Callee. If not present, it will create a secure TCP connection from its PORT-SDC to the callee's PORT-UES.
4. The INVITE is then forwarded to the Callee securely.

### SIP SUBSCRIBE and NOTIFY over TCP IMS-AKA

SUBSCRIBE and NOTIFY messages are exchanged between a UE and the P-CSCF in a manner similar to the previous INVITE example whereby the secure channel is first created and then the SIP messages are exchanged securely.

### IMS-AKA Change Client Port

The Oracle Communications Session Border Controller is now in compliance with 3GPP TS 33.203, *Access Security for IP-Based Services*. Previous releases did not comply with requirements specified in Section 7.4, Authenticated re-registration, which reads in part:

Every registration that includes a user authentication attempt produces new security associations. If the authentication is successful, then these new security associations shall replace the previous ones. This clause describes how the UE and P-CSCF handle this replacement and which SAs to apply to which message.

When security associations are changed in an authenticated re-registration then the protected server ports at the UE (port\_us) and the P-CSCF (port\_ps) shall remain unchanged, while the protected client ports at the UE (port\_uc) and the P-CSCF (port\_pc) shall change.

If the UE has an already active pair of security associations, then it shall use this to protect the REGISTER message. If the S-CSCF is notified by the P-CSCF that the REGISTER message from the UE was integrity-protected it may decide not to authenticate the user by means of the AKA protocol. However, the UE may send unprotected REGISTER messages at any time. In this case, the S-CSCF shall authenticate the user by means of the AKA protocol.

In particular, if the UE considers the SAs no longer active at the P-CSCF, e.g., after receiving no response to several protected messages, then the UE should send an unprotected REGISTER message.”

Prior releases failed to change the protected client ports after a successful re-registration.

## Protected Ports

Within IMS networks, the P-CSCF provides the network access point and serves as the outbound proxy server for user equipment -- smart phones, tablets, and similar devices. The UE must connect to the P-CSCF prior to registration and initiation of SIP sessions. Connection to the P-CSCF, which can be in the user's home network, or in a visited network if the UE is roaming, is accomplished using Dynamic Host Control Protocol (DHCP) P-CSCF discovery procedures.

After successful discovery, the P-CSCF and UE negotiate IPsec security associations (SAs) which are used to establish four protected (authenticated and encrypted using Encapsulating Security Payload protocol) ports between the UE and the P-CSCF.

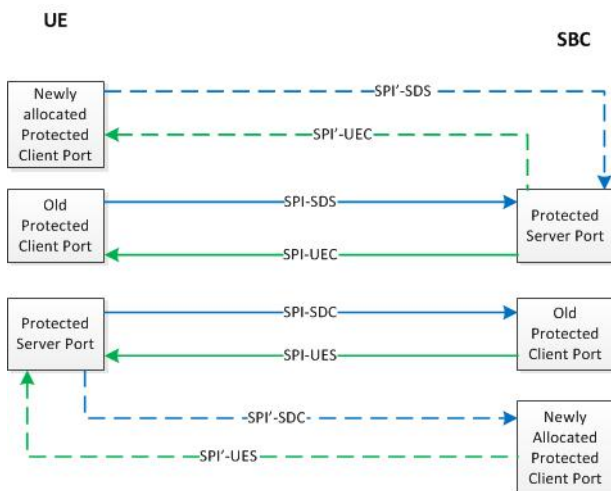
The four protected ports are shown in the following illustration:



As required by Section 7.4 of 3GPP TS 33.203, the protected client ports, one on the UE and the other on the SBC, must be changed after each successful re-registration.

To fulfill this requirement, this release adds a new attribute to the existing `ims-aka-profile` configuration object. This attribute (**`end-protected-client-port`**) works in conjunction with **`start-protected-client-port`** (**`protected-client-port`** in previous releases) to enable the identification of a pool of protected client ports, which will be used for re-registration scenarios where the SBC/P-CSCF is required to change the client port.

The SBC creates new protected client ports, one on the UE and the other on the SBC, after every re-registration. Old protected client ports, along with their associated SAs, are maintained for 30 seconds after re-registration to ensure correct handling or any pending responses to previously transmitted messages.



After successful re-registration, the SBC updates the registration cache with updated port information and checkpoint with the HA peer, if present.

---

## IMS-AKA Change Client Port Configuration

An IMS-AKA profile establishes the client and server ports to be protected, and it defines lists of encryption and authentication algorithms the profile supports. You can configure multiple IMS-AKA profiles, which are uniquely identified by their names.

You apply an IMS-AKA profile to a SIP port configuration using the name.

To configure an IMS-AKA profile:

1. From Superuser mode, use the following command sequence to navigate to `ims-aka-profile` configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# ims-aka-profile
ACMEPACKET(ims-aka-profile)#
```

2. **name**—Enter the name you want to give this IMS-AKA profile. This is the value you will use to apply the profile to a SIP port configuration. This parameter is required, and it has no default value.
3. **protected-server-port**—Enter the port number of the protected server port, which is the port on which the Oracle Communications Session Border Controller receives protected messages. The protected server port should not overlap with the port range defined in the steering ports configuration using the same IP address and the SIP interface. If there is overlap, the NAT table entry for the steering port used in a call will prevent SIP messages from reaching the system's host processor.

This parameter defaults to 0, which disables the function associated with the parameter. The valid range for values is 1025 to 65535.

4. **start-protected-client-port (protected-client-port in Release S-CX6.3.3M2 and earlier releases)**—Enter the start value for the pool of port numbers available following a successful re-authentication. Like the protected server port, the protected client port pool should not overlap with the port range defined in the steering ports configuration using the same IP address and the SIP interface. If there is overlap, the NAT table entry for the steering port used in a call will prevent SIP messages from reaching the system's host processor.

Any existing configuration for **protected-client-port** will be mapped to both **start-protected-client-port** and **end-protected-client-port** parameter values.

This parameter defaults to 0, which disables the function associated with the parameter. The valid range for values is 1025 to 65535.

5. **end-protected-client-port**—Enter the end value for the pool of port numbers available following a successful re-authentication. Ensure that this value is greater than the value assigned to **start-protected-client-port**. Note that the maximum supported pool contains 5 entries. Like the protected server port, the protected client port pool should not overlap with the port range defined in the steering ports configuration using the same IP address and the SIP interface. If there is overlap, the NAT table entry for the steering port used in a call will prevent SIP messages from reaching the system's host processor.

This parameter defaults to 0, which disables the function associated with the parameter. The valid range for values is 1025 to 65535.

6. **encr-alg-list**—Enter the list of encryption algorithms. You enter more than one value by separating the algorithms by <Spaces> and enclosing all values in quotations marks:

This parameter defaults to the following three values: **aes-cbc**, **des-ede3-cbc**, and **null**.

7. **auth-alg-list**—Enter the list of authentication algorithms. You enter more than one value by separating the algorithms by <Spaces> and enclosing all values in quotations marks:

This parameter defaults to **hmac-sha-1-96**.

### Sample IMS-AKA Configuration

The following formatted extract from `show running-config` CLI output shows a sample IMS-AKA profile configuration.

```
ims-aka-profile
name TS33.203
start-protected-client-port 4060
end-protected-client-port 4064
protected-server-port 4070
encr-alg-list aes-cbc des-ede3-cbc null
auth-alg-list hmac-sha-1-96 hmac-md5-96
last-modified-by admin@172.30.11.18
last-modified-date 2013-06-15 14:58:08
```

## SIP IMS P-CSCF P-Asserted Identity in Responses

In releases earlier than Release S-C6.1.0, the Oracle Communications Session Border Controller—operating as a P-CSCF—removes the P-Preferred-Identity header (if present) on receipt of a 1xx or 2xx response. It also inserts a P-Asserted-Identity header with the value received in the P-Preferred-Identity header.

Release S-C6.1.0 changes this behavior. Now the Oracle Communications Session Border Controller:

- Caches a copy of the P-Called-Party-ID header when it receives one of the following destined for a UE prior to forwarding the request:
  - An initial request for dialog
  - A request for a standalone transaction
  - A request for an unknown method that does not related to an existing dialog

The SIP interface receiving the request should have the SIP IMS feature enabled.

- Removes the P-Preferred-Identity header (if present) and inserts a P-Asserted-Identity header with the value saved from the P-Called-Party-ID header on receipt of a 1xx or 2xx response.

### Important Notes

Note the following:

- The endpoint to which the response is being sent must be a trusted endpoint. The option **disable-ppi-to-pai** should not be configured in the global SIP configuration's **options** list.
- If the P-Preferred-Identity header is present in the response, the Oracle Communications Session Border Controller will delete the header.
- If the P-Asserted-Identity header is present in the response, the Oracle Communications Session Border Controller will overwrite that -Asserted-Identity.

## SIP IMS P-CSCF P-Asserted Identity in Responses Configuration

This behavior is enabled automatically. You do not need to perform any configuration steps.

## SIP IMS P-CSCF S-CSCF Target Caching and Invalidation

In IMS architectures, the Oracle Communications Session Border Controller can form these roles:

- An access session border controller, acting as the media front end to a third-party proxy CSCF (P-CSCF)
- Combined access session border controller and P-CSCF

In both, the Oracle Communications Session Border Controller needs to resolve the next-hop signaling element with DNS using these methods: NAPTR resource record, DNS SRV, and DNS address query (A-query).

In addition to this use of DNS, the Oracle Communications Session Border Controller can also now be configured to use DNS for the purposes of load balancing toward the core network elements and resiliently tracking failures. Using DNS for core load balancing simplifies initial provisioning and support more graceful failover when upstream elements fall out of service. Although the DNS-based service route can appear active despite an IP address having

failed, the Oracle Communications Session Border Controller provides mechanisms to learn targets and their subsequent invalidation via DNS.

You use session agents to set much of the Oracle Communications Session Border Controller's behavior for S-CSCF target caching and invalidation. The following table describes special ways or change to session agent configuration you must make in order to achieve the desired Oracle Communications Session Border Controller behavior.

Behavior	Configuration
You can configure the Oracle Communications Session Border Controller to resolve the P-CSCF for a registering UE using a DNS NAPTR query.	Set the applicable session agent's port to 0 and its transport-method to *.
You can configure the system to resolve the P-CSCF for a registering UE using a DNS SRV query.	Set the applicable session agent's port to 0 and its transport-method to a value other than *.
You can configure the system to resolve the P-CSCF for a registering UE using a DNS A-query.	Set the applicable session agent's port to value greater than 0.
When you have configured the Oracle Communications Session Border Controller to resolve the P-CSCF for a registering UE using a DNS NAPTR query, you can establish the way it learns the next hop that will handle the UE's signaling. If multiple NAPTR RRs result or if the order values is the same, the one with the lowest order value becomes the target for registration. If there is no in-service destination available, the NAPTR RR with lowest preference of the remaining options becomes the target.	No special configuration is required.
When you can configure the Oracle Communications Session Border Controller to resolve the P-CSCF for a registering UE using a DNS SRV query, you can establish the way it learns the next hop that will handle the UE's signaling. If the SRV target is an FQDN host that resolves to multiple A RRs, the Oracle Communications Session Border Controller either selects a single A RR by hunting or round robins across multiple A RRs.	<p>To support the round-robin method, you can set the applicable session agent's load-balance-dns-query to round robin. Or, you can leave that parameter set to hunt (its default).</p> <p>Note that the round robin method does not work then the session agent has the dns-load-balance option configured. That option distributed requests to IP address resolved from SRV responses with the same weight and priority.</p> <p>Note that if the ping is enable for the session agent and load-balance-dns-query is set to round robin, the Oracle Communications Session Border Controller pings all the IP addresses resolved from the DNS query.</p>

### Acting as a B2BUA Front End to Third-Party P-CSCF

This section describes the support DNS load balancing provides for an Oracle Communications Session Border Controller acting as a B2BUA front end to a third-party P-CSCF.

#### NAPTR

The Oracle Communications Session Border Controller performs resolution of the P-CSCF for a registering UE using a DNS NAPTR query. This happens when a session agent is defined with its port set to 0 and its transport method set either to any or \*, or when there is no session agent defined. When no session agent is defined, the system performs the NAPTR query according to the registering UE's URI domain. And if no transport method is defined for the session agent, the default query type is DNS NAPTR. Note that the configuration of how to resolve the P-CSCF is defined for a specific session agent.

When it receives a UE'S initial registration message, the system performs an NAPTR DNS query for the next hop that will handle the UE's signalling. The Oracle Communications Session Border Controller selects the NAPTR resource record (RR) with the lowest order value as the target for the registration when multiple RRs are returned. If the order value is the same, the system uses the lowest preference RR AS THE target. And if no in-service destination is available, the system selects the RR with the lowest preference from the remaining RRS as the target registration.

## DNS SRV

The Oracle Communications Session Border Controller performs resolution of the P-CSCF for a registering UE, starting with a DNS SRV query for a session agent with its port set to 0 and transport method given a setting other than any or \*. Note the system provides this support with DNS SRV-based session agents.

When it receives a UE'S initial registration message, the Oracle Communications Session Border Controller performs a DNS SRV query for the next hop target that will handle the UE's signalling. If the SRV target is an FQDN host that then resolves to multiple A RRs, the system then:

- Selects a single A RR through hunting if the **load-balance-dns-query** parameter is set to **hunt** (i.e., the first of several A RRs will always be selected unless it goes out of service)
- Round-robins across multiple A RRs for a given SRV target (if the **load-balance-dns-query** parameter is set to **round-robin**)

By default, the system hunts for and selects a single A RR. If the **dns-load-balance** option is configured, then round-robin will not work. This option distributes the requests to IP addresses that are resolved from SRV responses with the same weight and priority. Also, if you enable the session agent ping and set the **load-balance-dns-query** to **round-robin**, then all IP addresses resolved through the DNS query will be pinged.

Note that when no session agent exists, standard SRV ordering and A record hunting are used.

## DNS

The Oracle Communications Session Border Controller generates a DNS query to resolve the P-CSCF for a registering UE if the session is with a port greater than zero (0).

## Acting as a P-CSCF

This section describes the support DNS load balancing provides for an Oracle Communications Session Border Controller acting as a P-CSCF.

For each UE registering through the system as a P-CSCF and using any of the three defined methods, the top service route resolved for the UE using DNS is used for subsequent refreshed and INVITE messages without querying DNS or using any selection method. DNS is used in these instances:

- The DNS result TTL expires and the new results are different or changed.
- The cached target associated with the UE is out of service.

For SIP messages the UE initiates other than a REGISTER, the system routes the messages according to the top service route. The DNS-resolved target is used for such messaging.

## NAPTR

The Oracle Communications Session Border Controller performs resolution of the I-CSCF for a registering UE with an NAPTR query for a session agent with its port set to 0 and transport method given a setting other than any or \*. When no session agent is defined, the system performs the NAPTR query according to the registering UE's URI domain. And if no transport method is defined for the session agent, the default query type is DNS NAPTR. Note that the configuration of how to resolve the P-CSCF is defined for a specific session agent.

When it receives a UE'S initial registration message, the Oracle Communications Session Border Controller performs an NAPTR DNS query for the next hop that will handle the UE's signalling. The system selects the NAPTR resource record (RR) with the lowest order value as the target for the registration when multiple RRs are returned. If the order value is the same, the system uses the lowest preference RR AS THE target. And if no in-service destination is available, the system selects the RR with the lowest preference from the remaining RRS as the target registration.

### DNS SRV

The Oracle Communications Session Border Controller performs resolution of the I-CSCF for a registering UE, starting with a DNS SRV query for a session agent with its port set to 0 and transport method given a setting other than any or \* or \*. Note that the system provides this support with DNS SRV-based session agents.

When it receives a UE'S initial registration message, the system performs a DNS SRV query for the next hop target that will handle the UE's signaling. If the SRV target is an FQDN host that then resolves to multiple A RRs, the Oracle Communications Session Border Controller then:

- Selects a single A RR through hunting if the **load-balance-dns-query** parameter is set to **hunt** (i.e., the first of several A RRs will always be selected unless it goes out of service)
- Round-robins across multiple A RRs for a given SRV target (if the **load-balance-dns-query** parameter is set to **round-robin**)

By default, the system hunts for and selects a single A RR. If the **dns-load-balance** option is configured, then round-robin will not work. This option distributes the requests to IP addresses that are resolved from SRV responses with the same weight and priority. Also, if you enable the session agent ping and set the **load-balance-dns-query** to **round-robin**, then all IP addresses resolved through the DNS query will be pinged.

Note that when no session agent exists, standard SRV ordering and A record hunting are used.

### DNS

The system generates a DNS query to resolve the I-CSCF for a registering UE if the session is with a port greater than zero (0).

## S-CSCF Configuration

To set the port, transport-method, and load-balance-query parameters for a session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type and **session-router** press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter. If you are editing a pre-existing configuration, you need to select the configuration before making changes.

```
ACMEPACKET ((session-router) # session-agent  
ACMEPACKET ((session-agent) #
```

4. **port**—Change this parameter to the port value you want, or leave it set to 5060 (default).
5. **transport-method**—Leave this parameter set to UDP (default), or enter an asterisk (\*) to support all transport methods.
6. **load-balance-dns-query**—To perform load balancing across multiple A RRs, change this value from **hunt** (default) to **round-robin**.
7. Save and activate your configuration.

## E-CSCF Support

---

An Emergency Call Session Control Function (E-CSCF) is an IMS core element that aids in routing emergency calls to an appropriate destination, such as a PSAP. E-CSCF functionality can be performed by the Oracle Communications Session Border Controller with appropriate local policy and network management control configuration.

The E-CSCF feature let the Oracle Communications Session Border Controller internally prioritize and route emergency calls to the corresponding Emergency Service Center, based either on the calling party's request URI, or based on location information retrieved from a CLF (Connectivity Location Function) for wireline/TISpan networks.



By integrating E-CSCF functionality into the P-CSCF (Oracle Communications Session Border Controller), networks can satisfy the common local requirement that certain telephony elements be deployed locally, rather than use single, centralized elements. Functions like the E-CSCF likely fall into this category.

## Service URN Support

To enable E-CSCF functionality, the Oracle Communications Session Border Controller can parse service URNs for local policy lookup keys, and as desination identifiers in network management controls (NMC). Ensure that the match-URN is entered correctly as: urn:service:sos” or "urn:service:sos.type or the Oracle Communications Session Border Controller will interpret the URN as a hostname. Please see RFC 5031 for more information on compliant URN construction.

## E-CSCF Configuration Architecture

There are four elements which comprise and enable E-CSCF support on the Oracle Communications Session Border Controller :

- CLF Connectivity
- NMC Emergency Call Control
- Local Policy
- Emergency Local Route Table

### CLF Connectivity

The Oracle Communications Session Border Controller must be configured with Diameter-based CLF support. This is accomplished by creating an appropriate external policy server configuration.

When the Oracle Communications Session Border Controller requests authorization from the CLF server, a Line-Identifier AVP which includes a location string is expected to be returned for the call. The returned location string will be used later for an LRT query.

### NMC Emergency Call Control

By configuring a Network Management Control (NMC), the Oracle Communications Session Border Controller can flag a call for special priority early after it is received and validated by the system. The **destination identifier** must be configured in the NMC with the service URN of an incoming emergency call. Also, the NMC configuration must have its **next hop** parameter left blank. This lets the Oracle Communications Session Border Controller route the emergency call with local policies.

For example, if **urn:service:sos** is the configured value in the NMC’s **destination identifier**, and an INVITE arrives on the Oracle Communications Session Border Controller with urn:service:sos in the request URI, the call will be flagged for emergency handling. The next step in call processing is for the INVITE to be evaluated by local policy.

### Local Policy

Local policies must be configured to match and then route an incoming emergency call. Once a local policy match is made, the Oracle Communications Session Border Controller looks to the configured policy attributes for where to forward the INVITE. A matching policy attribute’s next hop should be configured to point to an emergency LRT that contains specific destinations for emergency calls. In addition, the **elec str lkup** parameter must be set to enabled so the Oracle Communications Session Border Controller will perform an LRT lookup based on the location string returned in the CLF response.

The **eloc str match** parameter identifies the attribute, whose value in the location string will be used as the lookup key in the emergency LRT. For example, if the returned location string is:

```
loc=xxx;noc=yyyy;line-code=zzzz
```

and the **eloc str match** parameter is set to **noc**, then when the Oracle Communications Session Border Controller performs a local policy route search, it will search the LRT for yyyy. If the **eloc str match** parameter

left empty or if there is no match when `eloc str lkup` is enabled, the entire location string is used as the lookup key.


### Emergency LRT

The Oracle Communications Session Border Controller needs to be configured with an emergency LRT to route emergency calls to their destination.

As stated in the previous section, when searching an emergency LRT, any user defined parameter within a Location String may be used as the key to look up next-hop routing information.

LRT files support `<user type = string>` which enables the Oracle Communications Session Border Controller to perform searches on free form attributes that may appear in the returned location-string. The `<user type = string>` value for an entry in the emergency LRT should be set to a part or whole value returned in the CLF's location string. For example:

```
<?xml version="1.0" encoding="UTF-8" ?>
<localRoutes>
  <route>
    <user type="string">1234</user>
    <next type="regex">!^.*$!sip:911@192.168.200.140:5060!</next>
  </route>
  <route>
    <user type="string">loc=xxx;noc=yyyy;line-code=zzzz</user>
    <next type="regex">!^.*$!sip:911@192.168.1.139:5060!</next>
  </route>
</localRoutes>
```

 **Note:** Given that the Location String is not a well-defined string, care should be taken when defining and configuring the LRT tables.

LRTs must be individually uploaded to both the active and standby systems in an HA node; LRTs are not automatically replicated across nodes.

### CLF Response Failure

If there is no location string in a CLF's response or the CLF rejects the call, the Oracle Communications Session Border Controller uses the `default location string` parameter from the ingress SIP interface to populate the PANI header. The emergency call proceeds normally using this location string's information for emergency LRT lookups.

## E-CSCF Configuration

This procedure assumes that the Oracle Communications Session Border Controller is configured to communicate with a CLF. In addition, this procedure assumes the Oracle Communications Session Border Controller is configured and loaded with an appropriate LRT for E-CSCF Use.

To configure an NMC for E-CSCF use (baseline parameters are not mentioned):

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type `session-router` and press Enter to access the signaling-related configurations.

```
ACMEPACKET(configure)# session-router
```

3. Type `net-management-control` and press Enter.

```
ACMEPACKET(session-router)# net-management-control
```

4. **name**—Enter the name of this network management control rule; this value uniquely identifies the control rule. There is no default for this parameter.
5. **state**—Enable or disable this network management control rule. The default value is `enabled`. The valid values are:

- enabled | disabled
6. **type**—Set this parameter to **priority** so that the Oracle Communications Session Border Controller will flag incoming calls with a matching destination identifier as a priority calls.
  7. **treatment**—Set this parameter to **divert**.
  8. **next-hop**—Leave this parameter blank so that the call's processing will go directly to local policy.
  9. **destination-identifier**—Enter the service URN that endpoints in your network include in their request URIs to identify themselves as emergency calls.
  10. Save your configuration.

To configure local policy for E-CSCF use (baseline parameters are not mentioned):

11. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

12. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

13. Type **local-policy** and press Enter. If you are adding this feature to a pre-existing local policy configuration, you will need to select and edit your local policy.

```
ACMEPACKET (session-router) # local-policy
ACMEPACKET (local-policy) #
```

14. **to-address**—Set this parameter to the lookup key for matching emergency calls. You can now use a service URN as lookup key criteria.

15. Save your configuration.

To configure policy attributes for E-CSCF use (baseline parameters are not mentioned):

16. Type **policy-attributes** and press Enter. If you are adding this feature to a pre-existing local policy configuration, you will need to select and edit your local policy.

```
ACMEPACKET (local-policy) # policy-attributes
ACMEPACKET (policy-attributes) #
```

17. **next-hop**—Set this parameter to **lrt: name-of-elrt-file.gz** for this policy attribute to lookup routes in the named lrt file.
18. **eloc-str-lookup**—Set this parameter to **enabled** for the Oracle Communications Session Border Controller to parse the emergency location string, as received in a CLF Line Identifier AVP, for emergency LRT lookup.
19. **eloc-str-match**—Set this parameter to the attribute name found in the location string whose value will be used as a lookup key in the LRT named in the next-hop parameter. Common values include "loc" or noc.
20. Save and activate your configuration.

## Maintenance and Troubleshooting

The **show lrt route-entry** command displays two entries, if the username 1234 has a "string" type and "E164" type entries.

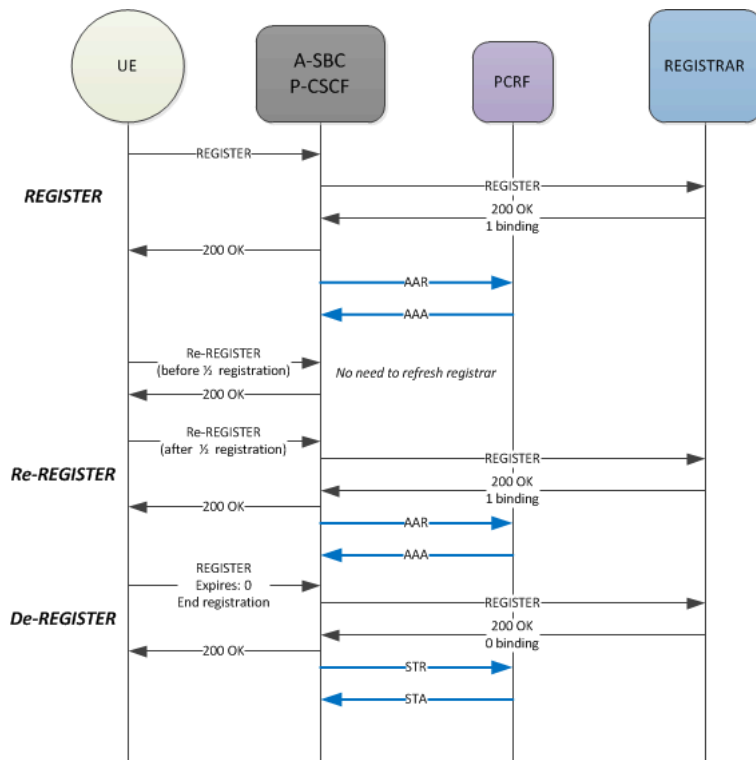
```
ACMEPACKET# show lrt route-entry emergency_lrt 1234
UserName <1234>
  User Type= E164
  NextHop= !^.*$!sip:911@192.168.200.139:5060!
  NextHop Type= regexp
UserName <1234>
  User Type= string
  NextHop= !^.*$!sip:911@192.168.200.140:5060!
  NextHop Type= regexp
```

## 2774 - Provisioning of SIP Signaling Flow Information

This feature supports of Provisioning of SIP Signalling Flow Information as described in 3GPP TS 29.213 section B1b [1], and the procedures specified in TS 29.214 section 4.4.5a.

The feature applies to the scenario when the Oracle Communications Session Border Controller (A-SBC / P-CSCF) uses its external policy server functionality and connects to a PCRF via Rx interface in bandwidth-management mode.

This feature deals with the information that the Oracle Communications Session Border Controller includes in an AAR message it sends to the PCRF when an endpoint registers, re-registers, and de-registers. The following diagram shows the typical scenario.



### Initial Registration

When an endpoint registers, the Oracle Communications Session Border Controller creates and sends an AAR message to a PCRF which includes the following information:

<AA-Request> ::= < Diameter Header: 265, REQ, PXY >

AVP	AVP Contents
Session-ID	OriginHost;0;0;<Key to Registration Cache> Key is usually the endpoint's AoR.
Auth-Application-ID	Application-ID of Rx (16777236)
Origin-Host	<ext-policy-server_name>.<ext-policy-realm>.<domain-name-suffix>
Origin-Realm	<ext-policy-realm>.<domain-name-suffix>
Destination-Realm	<IP Address of endpoint>@<destination realm of this AAR>
Framed-IP-Address AVP - v4 address	Layer 3 Endpoint-IP-address v4 - Framed-IP-Addressor

AVP	AVP Contents
Framed-IPv6-Prefix AVP - v6 address	v6 - Framed-IPv6-Prefix
Media-Component	Grouped AVP description follows

Media-Component-Description AVP ::= < AVP Header: 517 >

AVP	AVP Contents
Media-Component-Number	0
Media-Sub-Component	Grouped AVP description follows

Media-Sub-Component ::= < AVP Header: 519 >

AVP	AVP Contents
Flow-Number	1
Flow-Description	Permit in <ip> from <Endpoint IP:Port> to <SBC Sip Interface IP:Port> Permit out <ip> from <SBC SIP Interface IP:Port> to <Endpoint IP:Port> Where <ip> is (UDP: 17, TCP: 6) if wildcard-trans-protocol = disabled.
Flow-Status	Set to: ENABLED (2)
Flow-Usage	Set to: AF_SIGNALLING (2)
AF-Signalling-Protocol	Set to: SIP (1)

## Register Refresh

When a registration Refresh is received before the half time of the registration expiry, the registration cache is not updated and the Oracle Communications Session Border Controller responds with 200OK to the UE. This is standard registration cache functionality. No additional AAR is sent to the PCRF.

If a registration Refresh is received after the half time of the registration expiry, or if any registration information changes, the Oracle Communications Session Border Controller sends an AAR to the PCRF after it receives and forwards a 200 OK response from registrar. The AAR includes the same session-id as the initial AAR that was to PCRF. This lets PCRF correlate the AAR with the earlier AAR that it received.

## De-Registration

When an contact de-registers with expires=0, it means that the endpoint is removing all of its contacts from registration. When this occurs, the Oracle Communications Session Border Controller sends an STR message to the PCRF to terminate the session. If the de-registration message does not reflect a complete de-registration, the Oracle Communications Session Border Controller does not send the STR message to the PCRF. The STR message includes the following information:

<ST-Request> ::= < Diameter Header: 275, REQ, PXY >

AVP	AVP Contents
Session-ID	OriginHost;0;0;<Key to Registration Cache> Key is usually the endpoint's AoR.
Auth-Application-ID	Application-ID of Rx (16777236)

AVP	AVP Contents
Origin-Host	<ext-policy-server_name>.<ext-policy-realm>.<domain-name-suffix>
Origin-Realm	<ext-policy-realm>.<domain-name-suffix>
Destination-Realm	<IP Address of endpoint>@<destination realm of this AAR>
Destination-Host	
Termination-Cause	DIAMETER LOGOUT (1) - User initiated a disconnect

### Failure Response to Re-Register

Upon reception of a failure response from the REGISTRAR for a subsequent Registration refresh from endpoint, the Oracle Communications Session Border Controller performs de-registration actions, i.e. an STR message to the PCRF.

### Provisioning SIP Signaling Flows Configuration

To enable correct provisioning of signaling flows upon registration:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)#
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)#
```

3. Type **ext-policy-server** and press Enter.

```
ACMEPACKET(media-manager)# ext-policy-server
ACMEPACKET(ext-policy-server)#
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI select command) the external policy server that you want to edit.

4. **provision-signaling-flow**—Set this to enabled for the Oracle Communications Session Border Controller to send AAR messages to a PCRF on endpoint registrations.
5. Save and activate your work.

### Troubleshooting

Upon receipt of AAA response from PCRF for the AAR sent, the Oracle Communications Session Border Controller logs the status of the AAA received. Successful AAAs are noted by session-ID in EBMD logs. The log will look like:

```
Provision of SIP Signaling Flow (<session-ID>) via Diameter Rx on realm
<realm-id> successful.
```

Failed AAA response are noted at NOTICE level EBMD logs citing the failure response to AAR request with the session-ID sent. An a MINOR non-health-affecting alarm is also raised with the following text:

```
Provision of SIP Signaling Flow (<session-ID>) via Diameter Rx on realm
<realm-id> failed.
```

### show ext-band-mgr

The show ext-band-mgr command has been augmented to include the Register and DeRegister counts. For example:

```
ACMEPACKET# show ext-band-mgr
13:55:49-166
EBM Status          -- Period -- ----- Lifetime -----
                   Active   High   Total   Total  PerMax   High
Client Trans       0      0      0      7      4      1
```

Server Trans	0	0	0	0	0	0
Sockets	1	1	0	1	1	1
Connections	0	0	0	1	1	1
		---- Lifetime ----				
	Recent		Total	PerMax		
Reserve	0		0	0		
Modify	0		0	0		
Commit	0		4	2		
Remove	0		2	1		
Register		1		1	1	
DeRegister	1		1	1		
EBM Requests	0		6	3		
EBM Installs	0		6	3		
EBM Req. Errors	0		0	0		
EBM Rejects	0		0	0		
EBM Expires	0		0	0		
EBMD Errors	0		0	0		

## RTP and RTCP Bandwidth Calculation and Reporting

The Oracle Communications Session Border Controller supports changing bandwidth requirements in an ad-hoc multi-party conference by tracking reduced bandwidth needs as parties are placed on hold during the initiation of a multi-party call. The combination of the 5 AVPs are considered by network elements for this functionality.

This section is applicable to the Oracle Communications Session Border Controller's Rx implementation when acting as a P-CSCF and connecting with a PCRF. The 5 AVPs considered in this section are created and sent in AAR messages.

### Max-Requested-Bandwidth-UL & Max-Requested-Bandwidth-DL AVPs

These AVPs reflect RTP bandwidth requirements for a call. The default behavior (i.e., no options are configured) that dictates how these AVPs are created follows.

The average rate limit scenario is when no `b=AS:` parameter in the SDP and the media-profile > **average-rate-limit** parameter is configured to a value greater than 0. The Oracle Communications Session Border Controller inserts the media-profile > **average-rate-limit** parameter  $\times 8$  into both Max-Requested-Bandwidth-UL & Max-Requested-Bandwidth-DL AVPs.

The SDP bandwidth scenario is when there is a `b=AS:` parameter in the SDP and the media-profile configuration element has the following configurations:

- `average-rate-limit = 0`
- `sdp-rate-limit-headroom > 0`
- `sdp-bandwidth = ENABLED`

When these conditions are met, the Max-Requested-Bandwidth-UL & Max-Requested-Bandwidth-DL AVPs are populated with the value in the `b=AS:` parameter + **sdp-rate-limit-headroom** CLI parameter.

### Optional AVP Creation

When the `get-bw-from-sdp` option is configured in the sip-config, the following occurs:

When SDP contains the

```
b=AS :
```

parameter with valid value, the Oracle Communications Session Border Controller multiplies that value  $\times 1000$  and inserts the result in the Max-Requested-Bandwidth-UL and Max-Requested-Bandwidth-DL AVPs in an AAR message.

If there is no b=AS: line in the SDP, the value is taken from the media-profile > **average-rate-limit** parameter multiplied ×8 to get a bps value then inserted into the Max-Requested-Bandwidth-UL and Max-Requested-Bandwidth-DL AVPs in an AAR message.

The Oracle Communications Session Border Controller uses the b=AS: line (or chosen codec via the media-profile) from the final answer SDP for the session as the value for creation of these two AVPs.

### RR-Bandwidth & RS-Bandwidth AVPs

These AVPs reflect RTCP bandwidth requirements for a call. When SDP contains:

```
b=RR :  
b=RS :
```

parameters and values, the Oracle Communications Session Border Controller inserts the values (after the :) into the respective RR-Bandwidth (521) and RS-Bandwidth (522) AVPs. When these parameters are not present in the SDP, the RR-Bandwidth and RS-Bandwidth AVPs are not created in AAR messages sent to the PCRF. The sip-config > get-bw-from-sdp option must be configured to enable creation of these AVPs.

### Flow-status AVP

The Flow-status AVP (511) is based on SDP direction. Tests for SDP are performed in the following order:

SDP State	Flow-Status AVP (511) Set to
port in the SDP m-line is 0	REMOVED (4)
Transport in m-line is "TCP" or "TCP/MSRP"	ENABLED (2)
a=recvonly and <SDP direction> = UE originated	ENABLED-DOWNLINK (1)
a=recvonly and <SDP direction> = UE terminated	ENABLED-UPLINK (0)
a=sendonly and <SDP direction> = UE originated	ENABLED-UPLINK (0)
a=sendonly and <SDP direction> = UE terminated	ENABLED-DOWNLINK (1)
a=inactive	DISABLED (3)
a=sendrecv or no direction attribute	ENABLED (2)

UE originated - Call originator creates the SDP being considered.

UE terminated - Call terminator creates the SDP being considered.

Flow-status AVP is always created according and requires no configuration.

## 2629 - IR.92 Compliance via SIP 380 Response

---

This feature furthers the Oracle Communications Session Border Controller's compliance with GSMA's Voice over LTE specification (IR.92) to redirect VoLTE originated emergency calls to a circuit switched network.

When the Oracle Communications Session Border Controller receives an emergency call, which it can not complete, it returns a 380 (Alternative Service) response to the sender. Some examples of when the Oracle Communications Session Border Controller can not forward such an emergency call are:

- The Next-hop does not exist to route the call via NMC means
- The NSEP calls are rejected due to the Oracle Communications Session Border Controller hitting a load limit
- The NSEP calls are rejected due to the target session agent exceeding constraints
- The NMC treatment for the call is set to Reject



## 380 Response Format

The Oracle Communications Session Border Controller's 380 SIP response to the sender includes:

- Content-Type header field set to application/3gpp-ims+xml
- P-Asserted-Identity header field set to the value of the SIP URI of the last entry on the Path header field value received during registration; It is the value of the SIP URI of the P-CSCF.
- 3GPP IM CN subsystem XML body containing an <ims-3gpp> element with the "version" attribute will be set to "1" and with an <alternative-service> child element set to "alternative service"
  - a <type> child element set to emergency
  - a <reason> child element set to the value of configuration element send-380-response
  - an <action> child element set to emergency-registration

### 380 Response Example

```
SIP/2.0 380 Alternative Service
Via: SIP/2.0/UDP 192.168.15.2:5060;branch=z9hG4bK-23615-1-0
From: sipp <sip:911@192.168.15.2:5060>;tag=1
To: sut <sip:911@192.168.101.11:5060>
Call-ID: 1-23615@192.168.15.2
CSeq: 1 INVITE
Content-Type: application/3gpp-ims+xml
Content-Length: 209
P-Asserted-Identity: sip:911-44e2etbufgibf@172.16.101.11:5060
Reason: Q.850; cause=63
<?xml version='1.0' encoding="UTF-8"?>
<ims-3gpp version="1.0">
<alternative-service>
<type>emergency</type>
<action>emergency-registration</action>
<reason>sample reason</reason>
</alternative-service>
</ims-3gpp>
```

## IR.92 Compliance Configuration

To configure the 380 SIP response for IR.92 compliance:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET(configure)# session-router
```

3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

From this point, you can configure SIP interface parameters. To view all sip-interface parameters, enter a ? at the system prompt.

4. If configuring an existing interface, enter the select command to select the interface.
5. **send-380-response**—Enter a reason phrase enclosed in quotes to place in the reason tag of the <ims-3gpp> element in a 380 response for a failed-to-route emergency call.
6. Type **done** and continue.

## Emergency Calls from Unregistered Users

Version S-CZ7.2.0 provides compliance with Section 5.1.6.2 of 3GPP TS24.229. This section mandates that unregistered, or roaming User Endpoints (UE) must perform an emergency registration before sending any SIP request (for example, an INVITE) related to the emergency situation.

Version S-CZ7.2.0 adds a new parameter (options reject-unreg-priority-calls) that enforces compliance with Section 5.1.6.2, Initial Emergency Registration, of 3GPP TS24.229, IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3. Section 5.1.6.2 mandates that unregistered, or roaming User Endpoints (UE) must perform an emergency registration before sending any SIP request (for example, an INVITE) related to the emergency situation.

Prior to Version S-CZ7.2.0, the default SBC behavior (acting as a P-CSCF) was not compliant with Section 5.1.6.2 of 3GPP 24.229. Rather, an emergency call from an unregistered or roaming UE was processed and routed in exactly the same way as a similar call from a currently registered UE.

Processing of emergency calls received from an unregistered or roaming UE as specified by 3GPP TS24.229 can be summarized as follows.

First, the emergency call (for example an INVITE) is rejected by the SBC with a 380, Alternative Service response message, which includes

1. a Content-Type header field set to

```
application/3gpp-ims+xml
```

2. a P-Asserted-Identity header set to the value of the SIP URI of the last entry on the path header field value received during registration -- the SIP URI of the P-CSCF. This is the UE-side Service Route that the SBC sends out when sip-ims-feature is enabled.

3. an Alternative Service child element containing a Type element set to "emergency" ;

```
emergency-value = %x65.6D.65.72.67.65.6E.63.79 ; "emergency"
```

an Action element set to "emergency-registration"

```
%x65.6D.65.72.67.65.6E.63.79.2D.72.65.67.69.73.74.72.61.74.69.6F.6E ;  
"emergency-registration"
```

a Reason element set to

```
"priority calls not allowed from unregistered UEs"
```

Second, upon receipt of the 380, Alternative Response, the UE responds with an emergency registration.

Finally, upon receipt of the emergency registration, the SBC processes that registration as it would any other registration request.

## Emergency Calls from Unregistered Users Configuration

1. Access the **sip-interface** configuration element.

```
ACMEPACKET# configure terminal  
ACMEPACKET(configure)# session-router  
ACMEPACKET(session-router)# sip-interface  
ACMEPACKET(sip-interface)#
```

2. Select the **sip-interface** object to edit.

```
ACMEPACKET(sip-interface)# select  
<RealmID>:  
1: realm01 172.172.30.31:5060  
  
selection: 1  
ACMEPACKET(sip-interface)#
```

3. **options +reject-unreg-priority-calls**—Add this option to require compliance with Section 5.1.6.2 of 3GPP TS24.229 . If this option is enabled, emergency calls from unregistered or roaming UEs are rejected with a 380,

Alternative Service response, and the UE is required to perform an emergency registration prior to sending an emergency message. If this option is not present, emergency calls, by default, are processed and routed regardless of registration status.

```
ACMEPACKET(sip-interface)# options +reject-unreg-priority-calls
```

4. **disallow-priority-calls**—ensure that this option has not been enabled on the current SIP interface because it summarily rejects all emergency calls on the current SIP interface. Ensure that this option is NOT present to ensure compliance with 3GPP TS24.229.

```
ACMEPACKET(sip-interface)# options -disallow-priority-calls
```

5. **options send-380-response**—ensure that this option has not been enabled on the current SIP interface. The **send-380-response** option provides the text for the Reason header. The correct text is supplied by the 380 Alternative Service message. Ensure that this option is NOT present to ensure compliance with 3GPP TS24.229.

```
ACMEPACKET(sip-interface)# options -send-380-response
```

6. Type **done** to save your configuration.

## IR.94 Support

The Oracle Communications Session Border Controller supports IR.94 (IMS Profile for Conversational Video Service) for use in an IMS and LTE environment.

The Oracle Communications Session Border Controller supports both video and audio and can keep a call up by switching to audio mode when the video session is dropped. The Oracle Communications Session Border Controller receives a Specific-Action AVP in the RAR from the PCRF indicating the loss of video bearer. The Oracle Communications Session Border Controller then replies with a RAA, and finally sends a STR failing the call.

This behavior contradicts TS 29.214 that indicates the P-CSCF should wait for an ASR from the PCRF (indicating the bearer has been removed), and should not fail the call until all bearers have been removed.

The following behaviors enable IR.94 Support:

- The Oracle Communications Session Border Controller supports the IR.94 re-INVITE procedures for adding or removing video mid-call.
- The Oracle Communications Session Border Controller recognizes that a UE is capable of handling video calls when in the REGISTER message, the sip.video media feature is present in the Contact: header.
- The Oracle Communications Session Border Controller supports Audio-Video Profile with Feedback (AVPF) and SDP Capability Negotiation (RFC 5939) as documented in TS 26.114 section 6.2.1a.
- When the Oracle Communications Session Border Controller supports sessions (200 OK received after an INVITE) with a single media stream throughout the session duration, it may receive any of the following notifications in the Specific-Action AVP of the RAR from the PCRF:
  - INDICATION\_OF\_LOSS\_OF\_BEARER (2)
  - INDICATION\_OF\_RELEASE\_OF\_BEARER (4)
  - INDICATION\_OF\_OUT\_OF\_CREDIT (7)
  - INDICATION\_OF\_FAILED\_RESOURCES\_ALLOCATION (9)

The Oracle Communications Session Border Controller then:

1. Replies with an RAA acknowledging the RAR
  2. Sends a BYE to both the UE and the Core, then
  3. Sends an STR to the PCRF indicating the session has been terminated.
- When the Oracle Communications Session Border Controller supports sessions (200 OK received after an INVITE) with multiple simultaneous media streams sometime during the session, it may receive any of the following notifications in the Specific-Action AVP of the RAR from the PCRF:
    - INDICATION\_OF\_LOSS\_OF\_BEARER (2)
    - INDICATION\_OF\_RELEASE\_OF\_BEARER (4)

- INDICATION\_OF\_OUT\_OF\_CREDIT (7)
- INDICATION\_OF\_FAILED\_RESOURCES\_ALLOCATION (9)

The Oracle Communications Session Border Controller then:

1. Sends an RAA acknowledging the RAR, and no further action is required. When an eSR-VCC handover occurs, the ATCF (Oracle Communications Session Border Controller) sends a INVITE/UPDATE without video (voice-only SDP) toward the target UE.

### IR.94 Loss Of Voice Bearer

The Oracle Communications Session Border Controller provides compliance with an IMS Profile for Conversational Video Service (IR.94) requirement that specifies the termination of a multi-media session (voice and video) if the voice bearer is lost.

Version S-CZ7.2.0 adds a new parameter (options **terminate-on-voice-bearer-release**) that addresses an unlikely, but possible, corner case in which the voice bearer is lost but the video bearer remains in service. Prior to Version S-CZ7.2.0, the Oracle Communications Session Border Controller would retain the call as a video-only session. This behavior is not compliant with IR.94, which specifies that the video-only session should not be allowed to continue.

The option **terminate-on-voice-bearer-release** enables compliance with the IR.94 standard. Compliance is ensured by basing the decision to terminate or continue a multi-media session on the state of the voice bearer exclusively; the state of any other media bearer (video, for example) plays no role in the decision process. Consequently, if the voice bearer fails, the call terminates; failure of the video bearer, or other media stream, is not pertinent to call termination or continuance.

### IR.94 Loss Of Voice Bearer Configuration

Use the following procedure to enable IR.94 compliance.

1. Access the **ext-policy-server** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# ext-policy-server
ACMEPACKET(ext-policy-server)#
```

2. Select the **ext-policy-server** object to edit.

```
ACMEPACKET(ext-policy-server)# select
<name>:1: name=extpoll

selection: 1
ACMEPACKET(ext-policy-server)#
```

3. Use options **terminate-on-voice-bearer-release** to enable I.94 compliance.

```
ACMEPACKET(ext-policy-server)# options +terminate-on-voice-bearer-release+
ACMEPACKET(ext-policy-server)#
```

4. Type **done** to save your configuration.

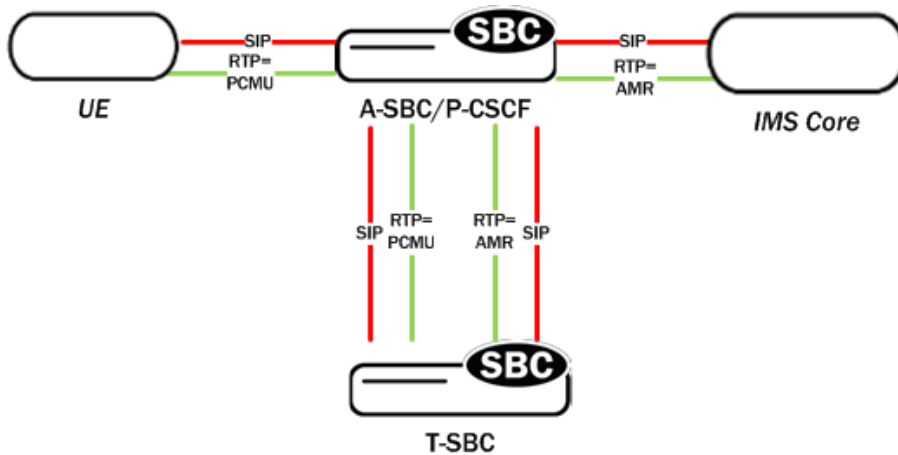
## Pooled Transcoding

---

The term pooled transcoding refers to a deployment model for IMS environments involving two or more Oracle Communications Session Border Controllers. The first is an A-SBC acting as the P-CSCF, and the others are one or more SBCs equipped with transcoding hardware (referred to as T-SBCs). The T-SBC provides transcoding resources—a pool—that the A-SBC can invoke on-demand.

In the pooled transcoding model, the A-SBC sits between realms or between user endpoints that require transcoding between their preferred codecs to communicate. This deployment model conserves resources on both the A-SBC and the T-SBC. While the A-SBC serves as the access function with encryption support, the T-SBC supports transcoding in a tunneling gateway (TG) configuration to meet high-density transcoding requirements.

The following diagram shows an A-SBC positioned between an access UE and the IMS core. The A-SBC compares SDP offers and answers from the elements it sits between, and uses the results to determine whether or not a given session requires transcoding. If transcoding is required, the A-SBC invokes T-SBC's services. Acting as a B2BUA, T-SBC uses information from the P-CSCF's SIP messaging to transcode between the applicable codecs and then to route SIP signaling back to the P-CSCF on the egress.



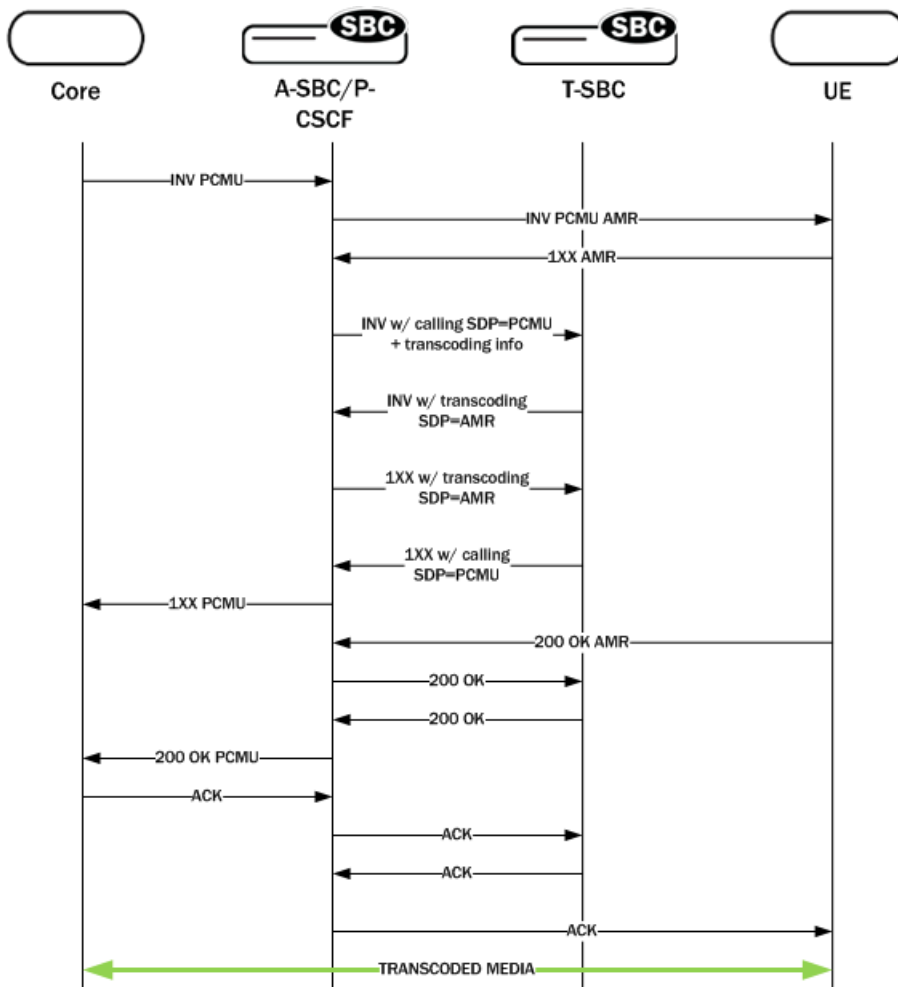
When a call comes in, the system creates two ports for the caller and callee realms respectively. When the call needs pooled transcoding, Oracle Communications Session Border Controller creates four ports on the pooled transcoded realm and two additional ports on the caller realm. All the ports are removed at the end of the call.

For example, for a call between the realms net172 (Offer) and net145 (Answer) :

Before pooled transcoding	During pooled transcoding	After pooled transcoding
Net172 -> 2 Ports	Net172 -> 4 Ports	Net172 -> 2 Ports
Net145 -> 2 Ports	Net145 -> 2 Ports	Net145 -> 2 Ports
Net192 (Transcoding Realm) -> 0 Ports	Net192 (Transcoding Realm) -> 4 Ports	Net192 (Transcoding Realm) -> 0 Ports

In the previous example, the SBC uses four ports on Net192 are for communicating with the transcoding SBC. The SBC uses the two newly created ports on Net172 for RTP communication, while the original two ports are not used.

The following diagram shows what a call flow between entities in such a deployment model looks like:



### Supported Codecs

This is a list of supported transcodable codecs Oracle Communications Session Border Controller that may be used within the pooled transcoding deployment model.

- PCMU
- CMA
- G729
- G729A
- iLBC
- telephone-event
- G726
- G726-16
- G726-24
- G726-32
- G726-40
- G722
- G723
- GSM
- AMR
- AMR-WB

## Hardware and Software Requirements

Pooled transcoding deployments have specific hardware and software requirements. See the Coproduct Support section in the Release Notes for more information.

## Implementation Details

From the A-SBC/P-CSCF perspective, the T-SBC is represented as a transcoding agent whose services the P-CSCF can invoke when offer-answer exchanges reveal transcoding is required. Once that determination is made, the P-CSCF initiates communication with the T-SBC. The P-CSCF uses its public SIP interface and a corresponding realm reserved for communication with a T-SBC, which is configured as a transcoding agent on the P-CSCF. Multiple transcoding agents can be configured, and can be IP addresses, session agents, or session agent groups (SAGs).

If there is more than one transcoding agent for the P-CSCF to choose from, the P-CSCF bases its selection on the order in which the transcoding agents were entered in the list. When conditions call for it, the P-CSCF will try each transcoding agent listed one by one until it:

- Receives a 2xx response from a transcoding agent,
- The list is exhausted, or
- The original transaction times out.

If the transcoding agents are session agents with hostnames, the P-CSCF uses DNS to resolve the hostnames and then tries the hosts in order. In the case when transcoding agents are SAGs, the P-CSCF selects the session agent according to the selection strategy configured for the SAG. The P-CSCF will recurse through all members of the SAG when SAG recursing is enabled. Whenever session agents and SAGs do not have ports or transport protocols specified, the defaults are 5060 and UDP respectively.

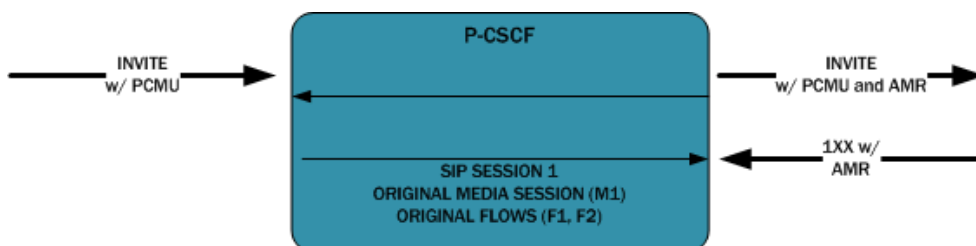
Once it identifies a transcoding agent, the P-CSCF sends an INVITE to which the transcoding agent—the T-SBC—responds with a 2xx message. Configuring the P-CSCFs as a session agent and disabling dialog transparency (in global SIP configuration) on the T-SBC allows it to accept SIP messages from the P-CSCF. Then the T-SBC acts as a B2BUA, using the information received in the P-CSCF's INVITES to invoke transcoding and to route SIP signaling back to the P-CSCF on the egress.

## Application Scenarios

This section discusses two application scenarios for pooled transcoding, one for how the P-CSCF handles an INVITE with SDP and one for how it handles an INVITE without SDP.

### Scenario 1 INVITE with SDP

When the P-CSCF receives an INVITE with SDP, it creates a SIP session and an associated media session that has two flows for audio. The P-CSCF applies the appropriate codec policy that has the **add-on-egress** configured, so the egress INVITE has this codec present.



The P-CSCF receives an answer to this INVITE, and if it contains the added codec, the P-CSCF invokes the T-SBC's transcoding services using an INVITE with the same SDP as the INVITE received on ingress. The communication between the P-CSCF and the T-SBC is a stand-alone dialog, and a new media session is associated with it.

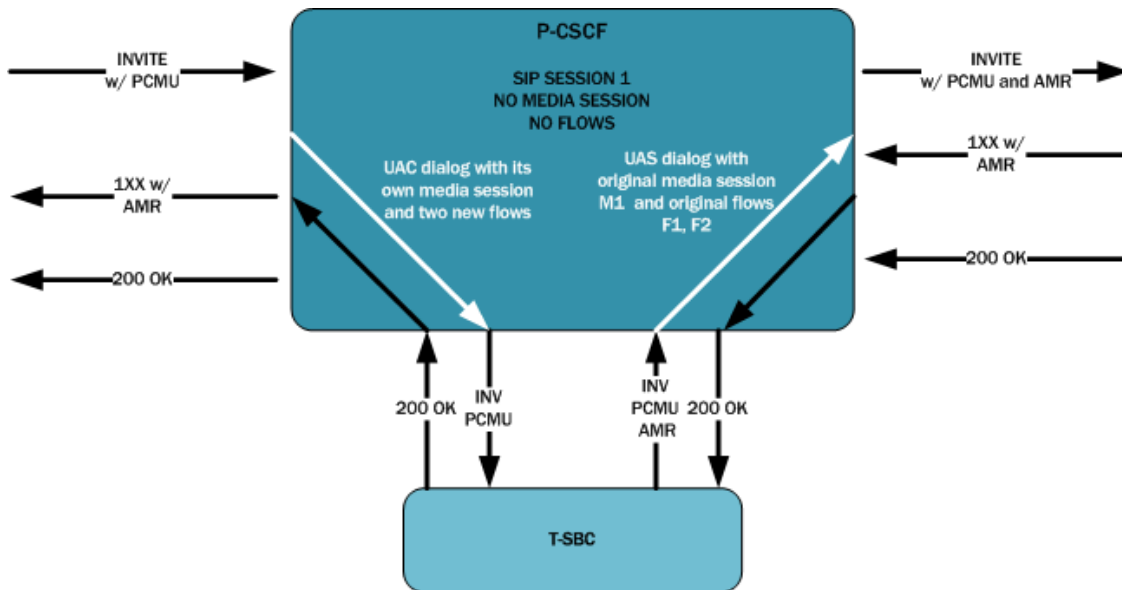
This is an example of what the INVITE the P-CSCF sends to the T-SBC looks like:

```
INVITE sip:192.168.101.78:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.101.18:5060;branch=z9hG4bK10dspa3058b1io4rk721
Max-Forwards: 70
```

```

Call-ID: 9fc71d79b43b4b95de41acefd71a8bc4@192.168.101.78
To: sip:192.168.101.78:5060
Contact: sip:172.16.101.18
From: sip:172.16.101.18;tag=bcf80756721880b7996ccb488b6a1b2f
CSeq: 1 INVITE
Target-Dialog: 1-16881@172.16.18.5;local-tag=1;remote-tag=16880SIPpTag011;realm=net192
Acme-Codec-Policy: ;ingress;name="in-2833";allow-codecs="* ";add-codecs-on-egress="";force-ptime="disabled";packetization-time="20";dtmf-in-audio="disabled";order-codecs=""
Acme-Codec-Policy: ;egress;name="out-2833";allow-codecs="* ";add-codecs-on-egress="PCMA ";force-ptime="disabled";packetization-time="20";dtmf-in-audio="disabled";order-codecs=""
Content-Type: application/sdp
Content-Length: 196^M
P-Visited-Network-ID: open-ims.test
Route: <sip:192.168.101.18:5060;lr;transport=UDP>
v=0
o=user1 53655765 2353687637 IN IP4 192.168.101.18
s=-
c=IN IP4 192.168.101.18
t=0 0
m=audio 20002 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
    
```

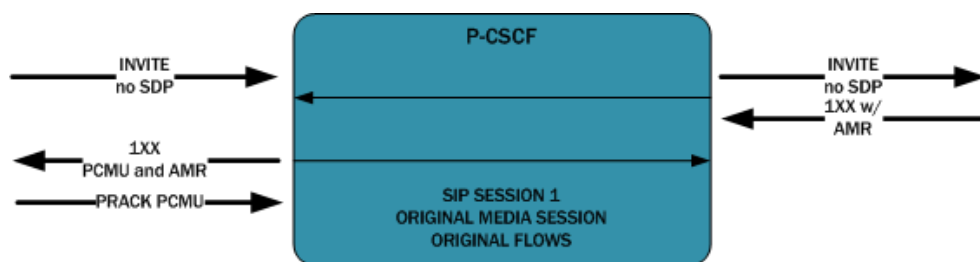
Note that the Target Dialog and Acme-Codec-Policy headers communicate operational parameters for pooled transcoding. Using this information, the T-SBC applies two codec policies and returns the INVITE to the P-CSCF. The P-CSCF moves the media session to itself, but the SIP session does not have a media session at this point.



**Scenario 2 INVITE without SDP**

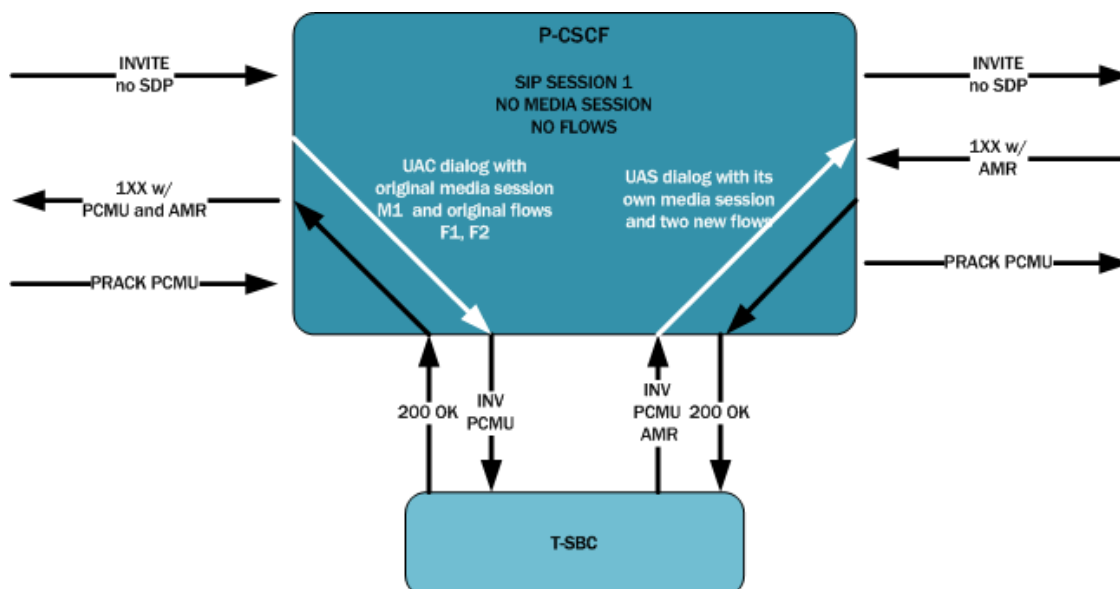
For an offerless call flow, the media session is created when the offer comes in a reliable provisional or final response. The P-CSCF applies the codec policy and sends the egress offer to the calling UE.





When it receives an answer in either a PRACK or ACK message, the P-CSCF compares the offer and answer. An answer containing the codec added in the egress offer causes the P-CSCF to invoke the T-SBC and create a standalone UAC dialog, just as in Scenario 1.

Since the P-CSCF advertised its media address in the egress offer to the calling UE, the original media session is used by the UAC dialog.



### Re-INVITES and Updates with SDP

Once a specific session on the P-CSCF invokes a T-SBC resources, it continues to use the same T-SBC for the session's entire duration. Anytime it receives a SIP message containing modified SDP, the P-CSCF communicates the modification to the T-SBC.

### RFC 2833 Considerations

For legacy RFC 2833 interworking to function properly, the P-CSCF communicates the RFC 2833 configuration to the T-SBC in the UAC dialog.

The following is a sample of what a P-CSCF's INVITE with RFC 2833 information sent to the T-SBC looks like:

```
INVITE sip:192.168.101.78:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.101.18:5060;branch=z9hG4bK10dspa3058blio4rk721
Max-Forwards: 70
Call-ID: 9fc71d79b43b4b95de41acefd71a8bc4@192.168.101.78
To: sip:192.168.101.78:5060
Contact: sip:172.16.101.18
From: sip:172.16.101.18;tag=bcf80756721880b7996ccb488b6a1b2f
CSeq: 1 INVITE
Target-Dialog: 1-16881@172.16.18.5;local-tag=1;remote-tag=16880SIPpTag011;realm=net192
Acme-Codec-Policy: ;ingress;name="in-2833";allow-codecs="* ";add-codecs-on-egress="";force-ptime="disabled";packetization-time="20";dtmf-in-audio="disabled";order-codecs=""
```

## IMS Support

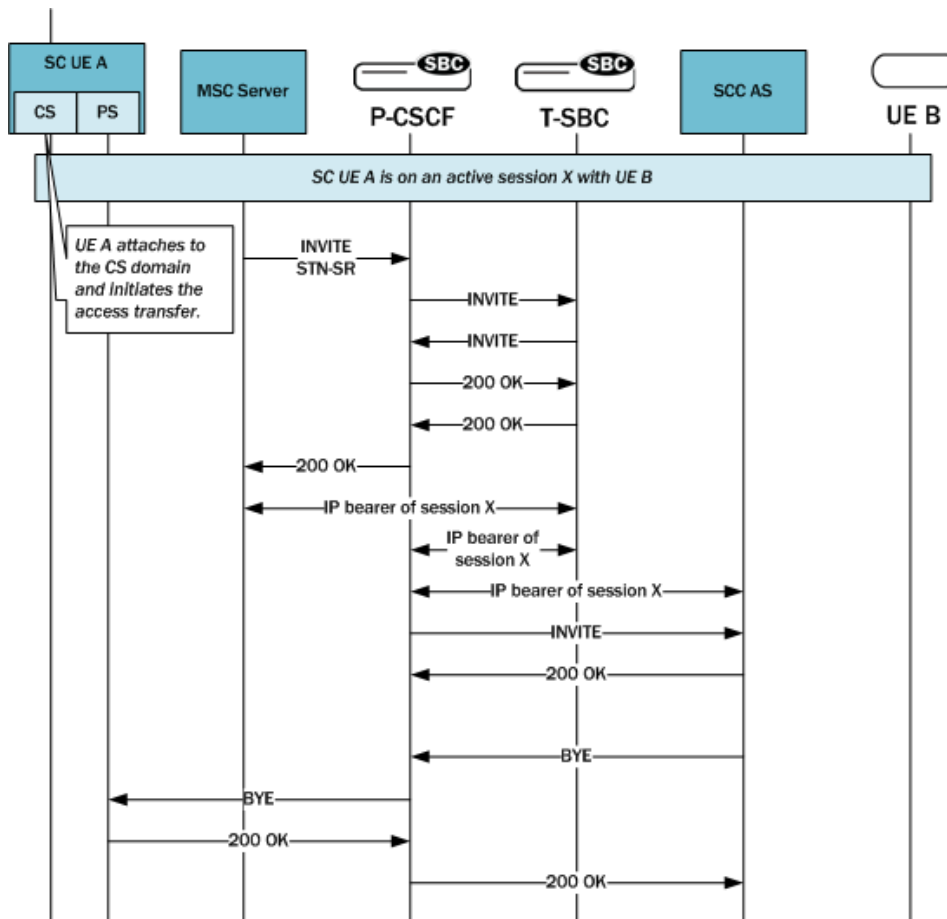
```

Acme-Codec-Policy: ;egress;name="out-2833";allow-codecs="* ";add-codecs-on-
egress="PCMA ";force-ptime="disabled";packetization-time="20";dtmf-in-
audio="disabled";order-codecs=""
Acme-Rfc2833: ;ingress;Digit-
Mode=0;PtTo=101;PtFrom=0;TransNon2833=disabled;TransNonInband=disabled
Acme-Rfc2833: ;egress;Digit-Mode=1;PtTo=101;PtFrom=0
Content-Type: application/sdp
Content-Length: 196
P-Visited-Network-ID: open-ims.test
Route: <sip:192.168.101.18:5060;lr;transport=UDP>
v=0
o=user1 53655765 2353687637 IN IP4 192.168.101.18
s=-
c=IN IP4 192.168.101.18
t=0 0
m=audio 20002 RTP/AVP 0 101
a=rtptime:0 PCMU/8000
  
```

## eSRVCC Support

For enhanced Signal Radio Voice Call Continuity (eSRVCC), the pooled transcoding deployment model supports instances when:

- Original call is not transcoded, but the handover call is
- Original call is transcoded, but the handover call is not
- Original call is transcoded, and so is handover call
- 



## Configuration Requirements and Verification

Pooled transcoding requires specific configuration of the Oracle Communications Session Border Controller and the T-SBC. Note that the Oracle Communications Session Border Controller does not enforce these requirements, so set up your pooled transcoding configuration with care. However, verifying your configuration can help to identify issues.

### A-SBC P-CSCF Requirements

Your A-SBC/P-CSCF configuration must include the following:

- A public SIP interface the A-SBC/P-CSCF can use for communication with the T-SBC.
- A transcoding realm, which is a separate realm for the public SIP interface and is only used for communication with the T-SBC.
- Appropriate codec policies, including ones set up to add SDP on the egress leg of the session.
- A global SIP configuration, with **transcoding-realm** set to the name where valid transcoding agents reside, and a **transcoding-agents** list configured with one or any combination of: a DNS hostname for a single session agent that can resolve to one or more IP addresses, an IP v4 or v6 address with or without the port specified (if not specified, the default port used is 5060); the name of a session agent group.

### T-SBC Requirements

A valid T-SBC configuration must include the following:

- A global SIP configuration, with **dialog-transparency** set to **disabled**—Dialog transparency must be disabled on the T-SBC so the T-SBC's INVITE will have a different call ID and From tag.
- A public realm with **codec-manip-in-realm** set to **enabled**, because the same realm is used for transcoding. In addition, the **mm-in-realm** parameter must be enabled.

### Configuration Verification

You can verify your configuration using the ACLI **verify-config** command. When verifying your configuration on the A-SBC, errors messages will appear if:

- The **transcoding-realm** value configured is not a valid realm.
- Either one of the **transcoding-realm** or **transcoding-agents** parameters is not configured.
- One or more session agent names defined in the **transcoding-agents** list is not a valid session agent.
- If the IP address version for an agent in the **transcoding-agents** list is not supported in the transcoding realm you identify; e.g., if you list an IPv6 agent in the list and the transcoding realm does not support IPv6, the system will return an error when you verify your configuration.
- If the transcoding agent is a hostname and not a valid session agent.

## ACLI Configuration

This section shows you how to configure a transcoding realm and transcoding agents on your A-SBC, acting as the P-CSCF in a pooled transcoding deployment model. You set these as part of the global SIP configuration.

To configure a transcoding realm and transcoding agents for the P-CSCF:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET(session-router) # sip-config
ACMEPACKET(sip-config) #
```

If you are adding this support to a pre-existing SIP configuration, you have to select the configuration before you make changes.

4. **transcoding-realm**—Enter the name of a configured realm designated as the separate realm for the public SIP interface, to be used only for communication with the T-SBC in pooled transcoding deployments.

```
ACMEPACKET(sip-config) # transcoding-realm net157
ACMEPACKET(sip-config) #
```

5. **transcoding-agents**—Enter the IP address, IP address and port combination, session agent hostname, or SAG name in this list if you want them to be used as transcoding agents. You can make multiple entries in any combination of these values. For example, you might list an IPv6 address and port, a session agent, and a SAG.
  - To make multiple entries in the list using in one command line, enclose the entire list of value in parentheses (()), separating each with a Space as in the example below.
  - To add a transcoding agent to an existing list, put a plus sign before the value you want to add, e.g. +154.124.2.8.
  - To remove a transcoding agent from an existing list, put a minus sign before the value you want to remove, e.g. -154.124.2.8.

```
ACMEPACKET(sip-config) # transcoding-agent (sag:sag1 192.168.4.7.3
192.168.2.6:5061)
ACMEPACKET(sip-config) #
```

6. Save and activate your configuration.

## Monitoring P-CSCF—T-SBC Dialogs

You can monitor pooled transcoding communications between the P-CSCF and the T-SBC using the **show sipd pooled-transcoding** CLI command. The display shows you information for the client and server User Agents on the P-CSCF.

- The UAC Dialogs section shows the number of UAC dialogs the P-CSCF initiated with the T-SBC; the count is cumulative and not for any specific session.
- The UAS Dialogs section shows the number of UAS dialogs the P-CSCF created upon receipt of an INVITE from the T-SBC; the count is cumulative and not for any specific session.
- The XCodeSessions counts the number of transcoded sessions on the P-CSCF.

```
ACMEPACKET# show sipd pooled-transcoding
18:11:28-125
SIP Pooled Transcoding Status
-----
-- Period --
Active High Total Total PerMax Lifetime -----
High
UAC Dialogs      1      1      1      1      1      1
  Early          0      0      0      0      0      0
  Confirmed      1      1      1      1      1      1
  Terminated    0      0      0      0      0      0
UAS Dialogs      1      1      1      1      1      1
  Early          0      0      0      0      0      0
  Confirmed      1      1      1      1      1      1
  Terminated    0      0      0      0      0      0
XCodeSessions    1      1      1      1      1      1
```

## Per-Method Statistics

If you set **extra-method-stats** to **disabled** in your global SIP configuration, you can display the P-CSCF's per-method statistics for the public transcoding-realm.

```

ACMEPACKET# show sipd realms net157 invite
INVITE (18:14:59-36)
----- Server -----
Message/Event   Recent   Total   PerMax   Client
----- Client -----
INVITE Requests      0         1         1         0         1         1
Retransmissions     0         0         0         0         0         0
100 Trying           0         1         1         0         1         1
200 OK               0         1         1         0         1         1
Response Retrans     0         0         0         0         0         0
Transaction Timeouts -         -         -         0         0         0
Locally Throttled   -         -         -         0         0         0

Avg Latency=0.000 for 0
Max Latency=0.000
BurstRate Incoming=1 Outgoing=0

ACMEPACKET# show sipd realms net157 bye
BYE (18:15:13-50)
----- Server -----
Message/Event   Recent   Total   PerMax   Client
----- Client -----
BYE Requests      0         1         1         0         1         1
Retransmissions     0         0         0         0         0         0
200 OK            0         1         1         0         1         1
Transaction Timeouts -         -         -         0         0         0
Locally Throttled -         -         -         0         0         0

Avg Latency=0.000 for 0
Max Latency=0.000
BurstRate Incoming=1 Outgoing=0

```

## Notes on the DIAMETER Rx Interface

DIAMETER Rx support for external bandwidth management is specialized for pooled transcoding.

For pooled transcoding, the identifier appears in this altered format: <policy server name>;<policy server realm>;<dns suffix>. Because there are two dialogs (one for the server UA and one for the client UA), there are two separate media sessions. This situation requires the Oracle Communications Session Border Controller to generate two different DIAMETER session identifiers. At the same time, the Oracle Communications Session Border Controller needs to maintain the AAR and STR associations with the original SIP session. To keep this relationship clear, the Oracle Communications Session Border Controller keeps the DIAMETER session identifier between the two media sessions the same for the two UAs.

The MBCD session identifier associated with the media session for the original SIP session is retained for the DIAMETER session identifier for the duration of the SIP session.

## Accounting

There is no difference between an ACR record for a typical SIP session and one for a transcoded section, except that the direction of the forward codec (Acme-FlowType\_FS1\_\_F) and the reverse codec (Acme-FlowType\_FS1\_\_R). For both RADIUS and DIAMETER records, the IP address and port corresponding to the original two endpoints will be reflected; the T-SBC's details will not appear in the records.

## Dynamic Sessions Agents for Home-Remote S-CSCF Liveliness

For IMS applications, you can configure your Oracle Communications Session Border Controller to create session agents dynamically for remote S-CSCFs on in-coming service routes. Dynamic session agents inherit properties of the static session agents with which they are associated, and the Oracle Communications Session Border Controller takes them out of service when they are deemed no longer responsive according to the liveliness mechanism you set.

### Discovery

The Oracle Communications Session Border Controller, acting as a P-CSCF, can discover remote S-CSCFs using Service-Route header that returns with a 200 OK response from the registrar for a REGISTER request from the endpoint. The system takes the top Service Route from the response and uses it as the first hop in the egress route for the endpoint. Because the creation of dynamic session agents is based on the Service Route returned in the 200 OK, there is no impact to handling AoRs with multiple contacts.

In addition, the system stores the Service Route header data with the endpoint's registration cache. If the Service Route is an FQDN, a DNS look-up is used to provide the route to the S-CSCF.

### Creation

For your Oracle Communications Session Border Controller to create session agents dynamically, you must enable the **create-dynamic-sa** in the global SIP configuration. This parameter defaults to **disabled**, meaning that no dynamic session agents will be created.

When you set the parameter to **enabled**, the Oracle Communications Session Border Controller decides whether or not to create dynamic session agents in the following ways. The system will create up to five (5) dynamic session agents.

- If the Service Route is an IP address, the Oracle Communications Session Border Controller attempts to find an exact match for that IP address against pre-existing session agents. If no exact match appears, the system will not create dynamic session agents.
- If the Service Route is an FQDN and matches an existing session agent, the Oracle Communications Session Border Controller assigns the remote S-CSCF to that session agent.

Service Routes with FQDNs might not match any session agents. However, the Service Route's FQDN can match the DNS suffix of a wildcard session agent. When this wildcard matching occurs, the Oracle Communications Session Border Controller creates a new dynamic session agent with the original wildcard session agent as its parent. The new dynamic session agent inherits all configuration properties of the parent session agent.

- If the Service Route is neither an IP address nor an FQDN and the system is unable to match any statically defined or wildcard session agents, then the Service Route is not associated with any session agents.

### Property Inheritance

Because dynamic session agents are created on the basis of static or wildcarded session agents, you can think of them as children of the original--or parent--session agent. This relationship means that the child, dynamic session agent inherits the configuration properties of its parent, static or wildcard session agent.

These inherited configuration properties and their effects are:

- The FQDN's DNS resolution of the new dynamic session agent to IP addresses.
- Monitoring for liveness via your configuration settings for the ping method and interval, transaction timeout, or OOS response. The dynamic session agent also inherits its parent's criteria for being taken out of service, with the exception that a dynamic session agent will not be taken out of service until it expires on its own. This gives the dynamic session agent time to return to service.
- The setting for **ping-all-address**, which when enabled causes new routes (internal session agents) fork and makes the dynamic session agent the parent. The system caps the limit of five routes (or internal session agents) per dynamic session agent. If the FQDN resolves to more than five IP addresses, the system only uses the first five to create routes (internal session agents), and then pings the internal session agents.
- Suppression of the heartbeat (or ping) of the IP address in the presence of traffic. If traffic to a specific IP address stops, then the Oracle Communications Session Border Controller resumes pinging within the time you set for the **ping-interval**.
- The setting for the **invalidate-registration** option. If the dynamic child session agent goes out of service, the corresponding registration cache entries of users that have the Service Route pointing to this session agent will be invalidated.

This invalidation means that the next REGISTER request from that user will not receive a local response. Any other services this user requires will not use the Service Route information stored in its registration cache. Instead,

the system will route it to the next hop as determined by other means, such as the local policy. At this time, the user would must be re-registered by the registrar, a process that might return a new Service Route to be updated in the registration cache.

## Deletion

The Oracle Communications Session Border Controller needs to delete dynamic session agents no longer in use. But dynamic session agents should not be deleted too soon, in case they return to service. So, deletion occurs according to the process this section describes.

For each dynamically created session agent, the Oracle Communications Session Border Controller assigns and tracks the last registration expiry time. It determines this time by doubling the registered endpoint's core side expiry. Where X is the last registration expiry value and Y is the registered endpoint's core side expiry, the Oracle Communications Session Border Controller performs checks according to this criteria:

$$X > (2 * Y)$$

The the system updates the expiry time with the greater value. This way, whenever the dynamic session agent re-registers, it will have an updated expiry timer value.

Timeouts can also cause dynamic session agents to be deleted. Pings, status changes, transaction timeouts, DNS expiries and other system occurrences can trigger timeouts. When the Oracle Communications Session Border Controller detects that a timeout has occurred, the dynamic session agent is deleted.

## How to Wildcard a Session Agent

You can create a wildcard session agent using the session agent's hostname parameter when configured with an FQDN.

To configure a wildcard session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
ACMEPACKET (session-router) #
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET (session-router) # session-agent
ACMEPACKET (session-agent) #
```

If you are wildcarding and existing session agent, you have to select the configuration before you make changes.

4. **hostname**—To wildcard a session agent, you simply replace the value you want to wildcard with an asterisk (\*). Also note that your value must lead with the asterisk (\*), as in the following example.

```
ACMEPACKET (session-router) # hostname *xyz.com
ACMEPACKET (session-agent) #
```

5. Save and activate your configuration.

## Enabling the Global SIP Configuration for Dynamic Session Agents

To use dynamic session agents for remote S-CSCFs on in-coming service routes, you need to set the create-dynamic-sa parameter in the global SIP configuration to enabled.

To configure the ping mode for a session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
ACMEPACKET (configure) #
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-config** and press Enter.

```
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

If you are adding this support to a pre-existing SIP configuration, you have to select the configuration before you make changes.

4. **create-dynamic-sa**—To support the creation of dynamic session agents for remote S-CSCFs on in-coming service routes, change this parameter from **disabled** (default) to **enabled**.
5. Save and activate your configuration.

## Enhanced eSRVCC Call Continuity

---

As the LTE Evolved Packet Core (EPC) continues to expand, voice deployments (VoLTE) appear more and more, with the Oracle SBC and P-CSCF playing key roles. 3GPP standards that define how LTE communications take place also continue to evolve, identifying and solving critical issues to increase effectiveness and efficiency. One such issue is session continuity, keeping session transfers between LTE and existing 2G and 3G networks as seamless as possible. Single Radio Voice Call Continuity (eSRVCC) offers one solution for the session continuity issue.

In its role as the P-CSCF and IMS-GW, the Oracle Communications Session Border Controller can provide eSRVCC by acting as signaling and media anchor points to handover calls to 3G networks smoothly. These anchoring points are called Access Transfer Control Function (ATCF) and the Access Transfer Gateway (ATGW), both of which are logical additions to the Oracle Communications Session Border Controller's IMS support.

## Handsets and Session Continuity

Session continuity can be effected by handsets, which can be dual-mode (3G+WiFi and 3G+WiMAX, for example). Such a handset has two receivers or radios to initiate calls simultaneously. An LTE handset has only one receiver and is able to attach to a single LTE or 3G network at a given time.

The question of session continuity appears in 3GPP standards Release 8, 9, and 10—each building on the previous. Release 8 defines Single Radio Voice Call Continuity (SRVCC) as the mechanism for moving active voice sessions between LTE and existing 2G or 3G circuit networks. In moving over sessions such as these, it is key to keep latency as low as possible to increase the possibility of successful handovers.

Release 9, because of variable signaling latencies within the core network, could not guarantee smooth handovers to circuit networks. And though IMS provides great flexibility in locating application servers remotely from the UE, that flexibility actually increases the latency in signaling media changes to the access network. Due to the high total signaling latency and the difficulty of successfully coordinating handover timing between 3G and 4G call legs, the possibility of call drops increased.

## Anchors for Signaling and Media

Release 10 addresses the latency concern by proposing these two logical entities, the anchoring points called the ATCF and ATGW. The Oracle Communications Session Border Controller can fulfill the tasks set to both entries:

- ATCF—A signaling anchor point co-located with the UE access network. The ATCF is responsible for:
  - Allocating the Session Transfer Number for Single Radio (STN-SR).
  - Instructing the ATGW to anchor the media path for originating and terminating sessions.
  - Tracking sessions (in alerting, active, or held states) so it can perform the access transfer of the selected session. Tracking this information allows the ATCF to support transferring the first session.
  - Performing Access Transfer and updating the ATGW with the new media path for the access call leg, without requiring updating from the remote leg.

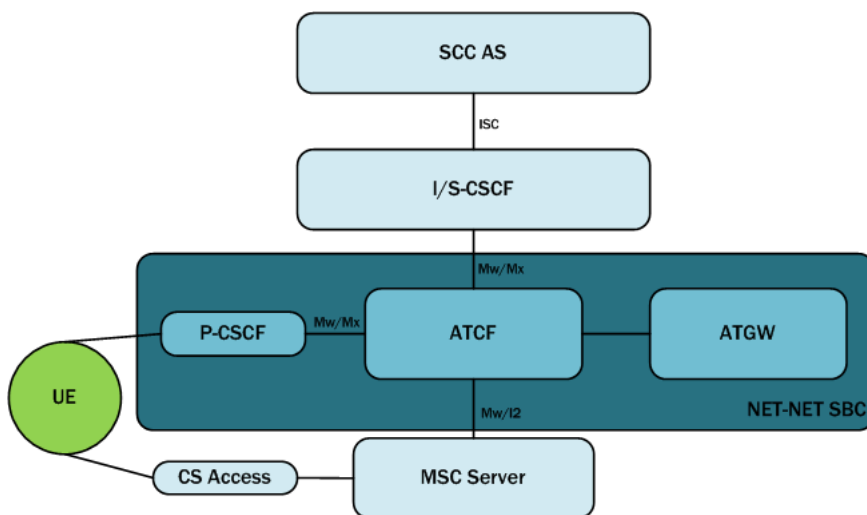


- After the access transfer, updating the Service Centralization and Continuity Application Server (SCC AS) that the transfer has taken place, ensuring that the Terminating Access Domain Selection (T-ADS) has updated information about the access currently used.
- Handling failures during the access transfer.
- Handling mid-call support for the access transfer using MSC server-assisted mid-call support.
- Access Transfer Gateway (ATGW)—A media anchor point co-located with the UE access network. Controlled by the ATCF, the ATGW anchors media both for the duration of a call and after the access transfer, based on the local configuration in the serving network.

Originating and terminating sessions are anchored in the ATCF and ATGW already during session set-up. For the first transferred session and the second established session, the SCC-AS provides session state information for the alerting, held, and conference states.

When a UE makes or receives a call, signaling and media are anchored at the ATCF and ATGW. At the point call handover point, the Visited-Mobile Switching Center (V-MSC) receives the handover message from Mobility Management Entity (MME, an EPC network element). The V-MSC then sends a call request to the local ATCF rather than sending the call request home or to the SCC-AS, as defined in 3GPP Release 8. Sending the call request to the ATCF reduces the number of hops required to initiate a media stream change to a new access network.

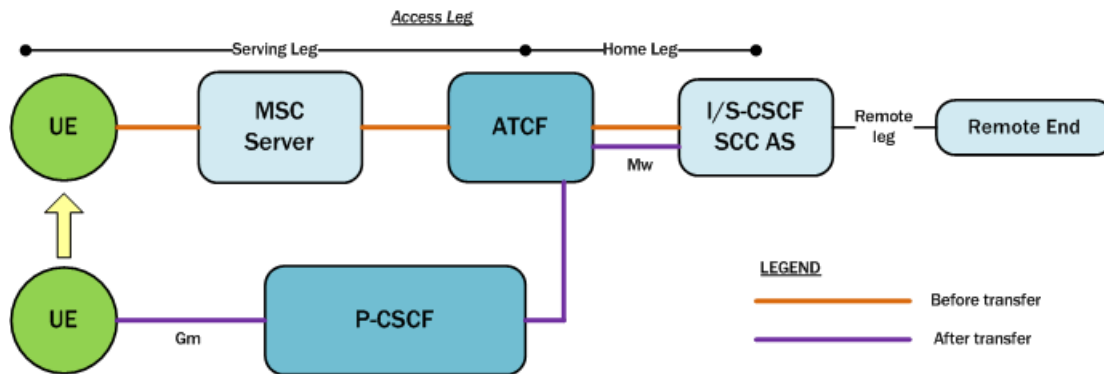
Acting as the ATGW, the Oracle Communications Session Border Controller can immediately affect media switchovers without further core signaling. In this role, the Oracle Communications Session Border Controller keeps RTP media continuity between endpoints using its Hide Media Update (HMU) functionality: The elements on the core side of the eSRVCC update will not see changes to SDP sources and destinations because the Synchronization Source identifiers (SSRCs) are masked. Media can thus flow directly to and from the 3G-attached MSC and onward to the UE with minimal interruption.



Note that if the MSC server-assisted mid-call feature is not supported or MSC server is not enhanced for ICS support, the interface between the MSC server and the ATCF will be Mw. If the MSC server-assisted mid-call feature is supported or the MSC server is enhanced for ICS, the interface between the MSC server and the ATCF will be I2.

## Architectural View

This diagram is an architectural view of the control and user planes, depicting communication both before and after transfer. This depiction assumes the PGW and the P-CSCF are in the serving network and support IMS voice roaming (if not the home network). So, the ATCF resides in the serving networking (home network if not roaming). The access leg of the session is divided into the serving leg and home leg. In an actual network, there might be other server IMS nodes.



## IMS Registration Details

This section discussed the IMS registration process when a UE attempts to register with the home network, but must do so using an ATCF/P-CSCF. Based on the operator policy and the home network's support for eSRVCC, the ATCF allocates an STN-SR to the session and includes itself in the signaling path for subsequent registration-related messaging. To understand whether or not the home network supports eSRVCC, the ATCF uses service-level agreements and can also determine if eSRVCC is activated in the SCC AS by the reception of a C-MSISDN/ATU-SI during session start-up.

If the ATCF generates an STN-SR, it includes the STN-SR in requests it forwards to the I/S-CSCF. The path and route information for the SIP REGISTER request sent to the S-CSCF is:

- Path—ATCF URI for terminating requests (uniquely identifies registration or registration flow), followed by P-CSCF URI for terminating requests.

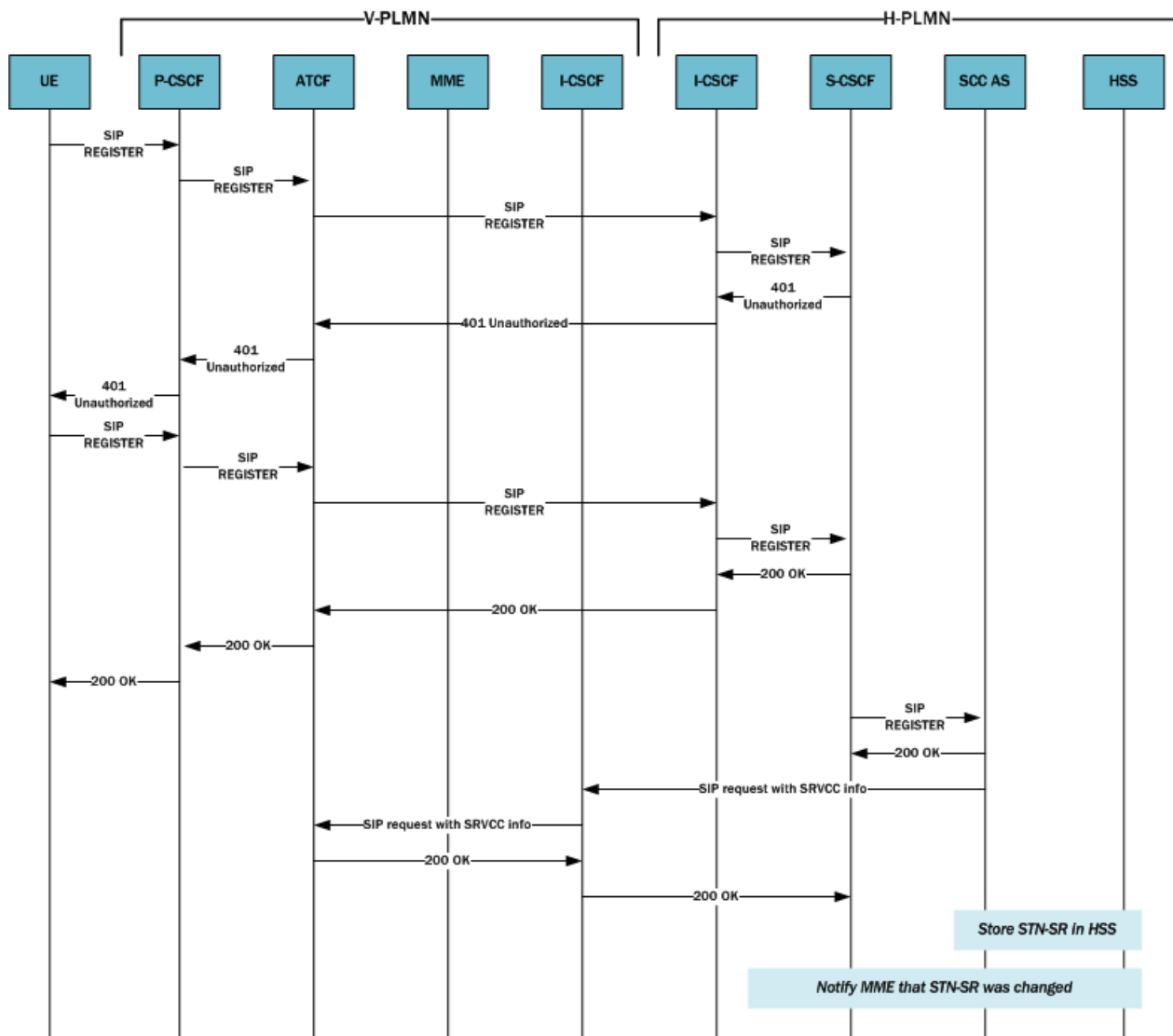
The header field containing the ATCF URI for terminating requests contains the `g.3gpp.atcf` media feature tag (indicating this URI supports the ATCF role). This tag contains the STN-SR and the ATCF PSI, showing this URI can receive SIP message requests with SRVCC-related information.

- Route—URI of the entry point of the UE's home network.

The ATCF tracks existing registrations for the UEs it has served. Each registration is identified by the P-CSCF path URI. The following information remains in the ATCF's registration cache: the S-CSCF service route URI, the ATU-STI, and the C-MSISDN. When a UE's registration expires or it is de-registered, the ATCF can remove any SRVCC information bound to the registration.

This ladder diagram shows the IMS registration process between these entities:

- The Visited Public Land Mobile Network (V-PLMN)—The network a mobile subscriber uses when that subscriber is roaming.
- The Home Public Land Mobile Network (H-PLMN)—The mobile subscriber's home network.



### SIP Register Request UE to ATCF

The following is an example of the SIP REGISTER request the UE sends to the ATCF/ P-CSCF.

```
REGISTER sip:home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8
Max-Forwards: 70
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
From: <sip:user1_public1@home1.net>;tag=2hiue
To: <sip:user1_public1@home1.net>
Contact: <sip:[5555::aaa:bbb:ccc:eee];comp=sigcomp>;+sip.instance="urn:gsma:imei:90420156-025763-0";+g.3gpp.icsi-ref="urn:3Aurn-7:3gpp-service.ims.icsi.mmtel"
Call-ID: E05133BD26DD
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net", nonce="", uri="sip:home1.net", response=""
Security-Client: ipsec-3gpp; alg=hmac-sha-1-96; spi-c=23456789; spi-s=12345678; port-c=1234; port-s=5678
Require: sec-agree
Proxy-Require: sec-agree
CSeq: 1 REGISTER
Supported: path, gruu
Content-Length: 0
```

### SIP Register Request ATCF to S-CSCF

The following is an example of the SIP REGISTER request the ATCF sends to the S-CSCF.

```
REGISTER sip:home1.net SIP/2.0
Path: <sip:termsdgdgdfwe@actf.visited2.net>;+g.3gpp.atcf="<tel:+1-237-888-9999>";+g.3gpp.atcf-
psi="<sip:actf.visited2.net>",<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Route: <sip:icscf.home1.net;lr>
P-Visited-Network-ID:
P-Charging-Vector:
Via: SIP/2.0/UDP actf.visited2.net:5060;branch=z9hG4bKnas5889; SIP/2.0/UDP
pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
[5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:bbb:
ccc:eee
Max-Forwards: 68
P-Access-Network-Info:
From:
To:
Contact:
Call-ID:
Authorization:
Require:
Proxy-Require:
CSeq:
Supported:
Content-Length:
```

### SIP 200 OK from S-CSCF

The following is an example of the SIP 200 OK from the S-CSCF.

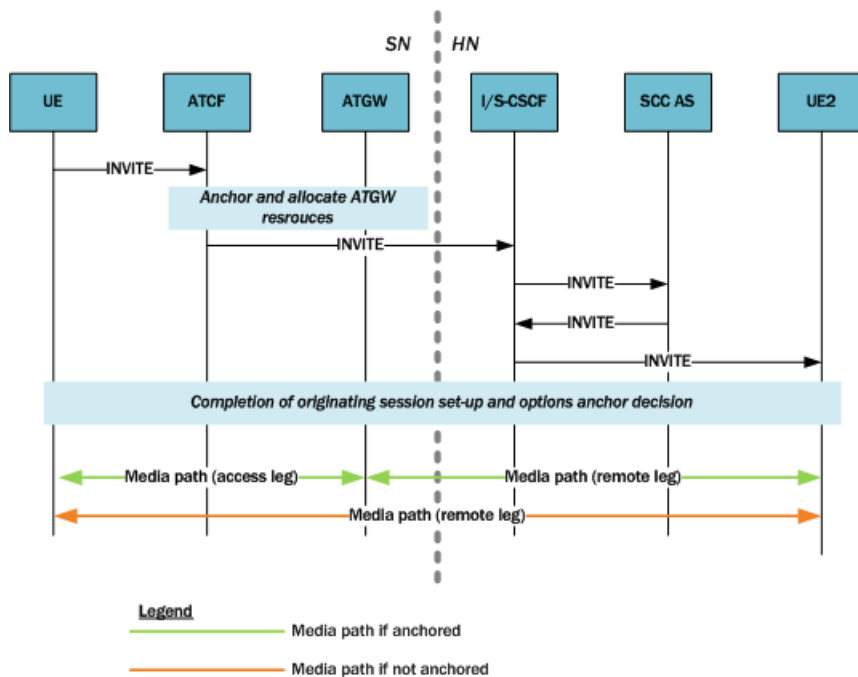
```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1_p.home1.net;branch=z9hG4bK351g45.1, SIP/2.0/UDP
pcscf1.visited1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Path: <sip:term@pcscf1.visited1.net;lr;ob>
Service-Route: <sip:orig@scscf1.home1.net;lr>
From:
To:
Call-ID:
Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>;
pub-gruu="sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6"
;temp-gruu="sip:tgruu.7hs==jd7vnzga5w7fajsc7-ajd6fabz0f8g5@example.com;gr"
;+sip.instance="< urn:gsma:imei:90420156-025763-0 >"+g.3gpp.icsi-ref="urn:3Aurn-7*3gpp-
service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype="cellular1"
;expires=600000;+g.3gpp.iut-controller
CSeq:
Supported: path, outbound
Require: outbound
Date: Wed, 11 July 2001 08:49:37 GMT
P-Associated-URI: <sip:user1_public2@home1.net>, <sip:user1_public3@home1.net>, <sip:+1-212-555-
1111@home1.net;user=phone>
Content-Length:
```

### Originating Sessions for SRVCC with ATCF

For initial SIP requests, the ATCF distinguishes SIP INVITE requests with the ATCF URI for originating requests in the topmost Route header field. And when receiving such an originating SIP INVITE request, the ATCF will do the following prior to forwarding it:

- Insert the Record-Route header field with its own SIP URI.
- If the latest SRVCC information received for a session contains a C-MSISDN and ATU-STI:

- The ATCF will associate the session being established with the C-MSISDN and ATU-STI bound to the registration.
- The ATCF will replace the SDP offer in the originating SIP INVITE with updated SDP the ATGW provides. Replacement occurs if the originating SIP INVITE contains SDP and a determination to anchor media has been made (according to the operator policy as specified in the 3GPP standard).



### SIP INVITE for SRVCC Using the ATCF

The following is an example of the SIP INVITE the UE sends to the ATCF/P-CSCF.

```
INVITE tel:+1-212-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited2.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfgkj490333
Cseq: 127 INVITE
Require: sec-agree
Supported: precondition, 100rel, gruu
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-
c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
00a0c91e6bf6;comp=sigcomp>;+g.3gpp.icsi-ref="urn:urn-7:3gpp-service.ims.icsi.mmtel"
Accept-Contact: *;+g.3gpp.icsi-ref="urn:urn-7:3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (...)

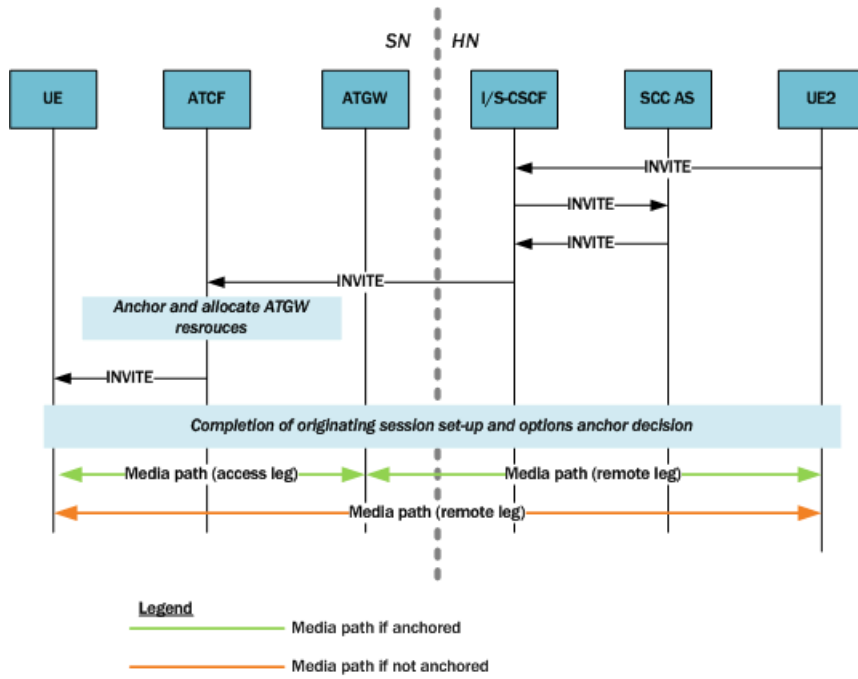
v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

## Terminating Sessions for SRVCC with ATCF

For initial SIP requests, the ATCF identifies SIP INVITE requests with the ATCF URI for terminating requests in the topmost Route header field. These are called terminating SIP INVITE requests.

When it receives a terminating SIP INVITE, the ATCF does the following if the INVITE has a Record-Route header field with the g.3gpp.srvcc media feature tag:

- The ATCF inserts a Record-Route header field with its own SIP URI
- If the latest SRVCC information received for a session contains a C-MSISDN and ATU-STI,
  - The ATCF associates the session being established with the C-MSISDN and ATU-STI bound to the registration, and
  - The ATCF replaces the SDP offer in the originating SIP INVITE with updated SDP provided by the ATGW. Replacement occurs if the terminating SIP INVITE contains an SDP offer and a determination to anchor media has been made (according to the operator policy as specified in the 3GPP standard).



### SIP INVITE from UE2 ATCF

The following is an example of the SIP INVITE UE2 sends to the ATCF/P-CSCF.

```
INVITE < sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6> SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bKnas34r5
Max-Forwards: 67
Route: <sip:scscf1.home1.net:lr>
P-Asserted-Identity: <tel: +1-237-555-2222>
P-Charging-Function-Addresses: ccf=[5555::b99:c88:d77:e66]; ccf=[5555::a55:b44:c33:d22];
    ecf=[5555::1ff:2ee:3dd:4ee]; ecf=[5555::6aa:7bb:8cc:9dd]
P-Charging-Vector: icid-value="AyretyU0dm+602IrT5tAFrbHLso=023551024"; orig-ioi="type3home1.net"
P-Access-Network-Info:
Privacy: none
From: <tel: +1-237-555-2222; gr=hdg7777ad7aflzig8sf7>;tag=171828
To: <tel:+1-237-555-1111>
Call-ID: cb03a0s09a2sdfgkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg= hmac-sha-1-96; spi=87654321; port=7531
Contact: < sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
    ad76cc7fc74>;+g.3gpp.icsi-ref="urn:3Aurn-7*3gpp-service.ims.icsi.mm1tel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=message 0 TCP/MSRP 98
a=accept-types:text/plain
```

### TS 24.237 Proposed Changes

The Oracle Communications Session Border Controller implements a proposed change in the processing of failed or cancelled SRVCC sessions. The new processing model has been presented to the 3GPP for inclusion in TS 24.237, IP Multimedia (IM) Core Network (CN) subsystem IP Multimedia Subsystem (IMS) service continuity; Stage 3.

Sections 12.2.4.13, 12.3.3.1, and 12.3.3.1A of 3GPP TS 24.237, IP Multimedia (IM) Core Network (CN) subsystem IP Multimedia Subsystem (IMS) service continuity; Stage 3, describe procedures in response to SRVCC handover cancellation. These procedures allow the UE to generate a e-INVITE request with the Reason header field containing a value of "SIP", a "cause" parameter set to a value of "487" (Request Terminated), and with reason-text parameter set to a value of either "handover cancelled" or "failure to transition to CS domain".

SCC AS processing, which does not release the original access leg, is defined in Section 12.3.3.1 and subclause 12.3.3.1A. Section 13.3.1 describes the handling of subsequent UPDATE requests. There are, however, no defined procedures at the ATCF response to this scenario with the result that the ATCF fails to reconfigure the ATGW to reconnect the bearer in LTE.

Oracle Corporation, in conjunction with other interested parties has proposed the addition of a new Section 12.7.2.3.3 to TS 24.237. This section requires the following ACTF behavior.

Requirement 1: When the ATCF receives either a



- SIP BYE request on the Source Access Leg containing a Reason header field containing a SIP 503 (Service Unavailable) response code, that is terminating an established dialog or an early dialog on the Source Access Leg
- SIP CANCEL request on the Source Access Leg with the Reason header field containing a SIP 503 (Service Unavailable) response code then, that is terminating an early dialog on the Source Access Leg originated by the SC UE, or
- SIP 503 (Service Unavailable) response on the Source Access Leg, that is terminating an early dialog on the Source Access Leg terminating at the SC UE

Then -- The ATCF shall retain session state information and ATGW resources associated with the session until either it receives a SIP INVITE request due to STN-SR, or a specified time period elapses (default value is 8 seconds).

The session remains recognizable for SRVCC access transfer as described in Section 12.7.2.1.

The SIP BYE request is forwarded to the SCC AS, which also delays release of the session, as described in Section 12.3.3.2.

Requirement 2: If the transferable session set determined in Section 12.7.2.1 does not contain any sessions and the identity in the P-Asserted-Identity header field is a C-MSISDN that is not bound to a registration path in the ATCF, the ATCF shall respond with a SIP 404 (Not Found) response.

Requirement 3: When the ATCF receives a SIP re-INVITE request containing Reason header field containing protocol "SIP" and reason parameter "cause" with value "487" on the original source access leg,

- after having initiated an access transfer that was triggered by a SIP INVITE request due to STN-SR, and
- the SIP INVITE request due to ATU-STI transaction is not yet completed

Then -- The ATCF shall wait until this transaction has completed and then continue with the steps described in Requirement 4.

Requirement 4: When the ATCF receives a SIP re-INVITE request(s) containing protocol "SIP" and reason parameter "cause" with value "487" after having performed an access transfer that was triggered by a SIP INVITE request due to STN-SR,

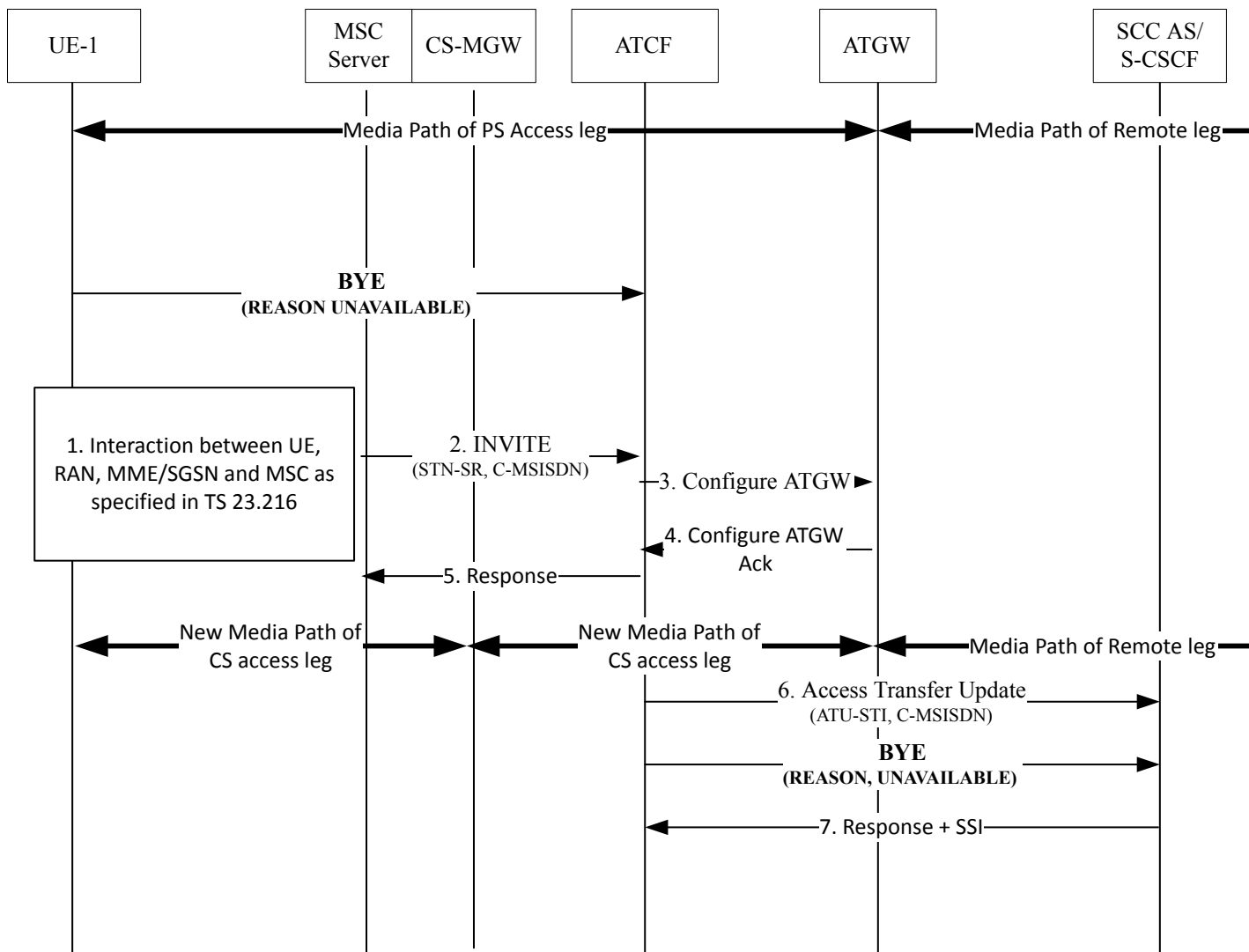
Then -- The ATCF shall act as B2BUA as described in Section 5.6 and shall.

1. Interact with ATGW to provide information needed in the procedures below and to request ATGW to start forwarding the media from the remote UE to the local UE. The details of interaction between ATCF and ATGW are out of scope of this document.
2. Send a SIP 200 (OK) response to the received SIP re-INVITE request. The SIP 200 (OK) response contains the SDP answer that includes the ATGW ports and the IP addresses as provided by the ATGW and the media used on the original source access leg as before the access transfer; and
3. Forward the received reINVITE with the Reason header intact to the SCC AS on the existing source dialog with the SDP offer containing the ATGW IP addresses and ports towards the remote UE as provided by the ATGW.

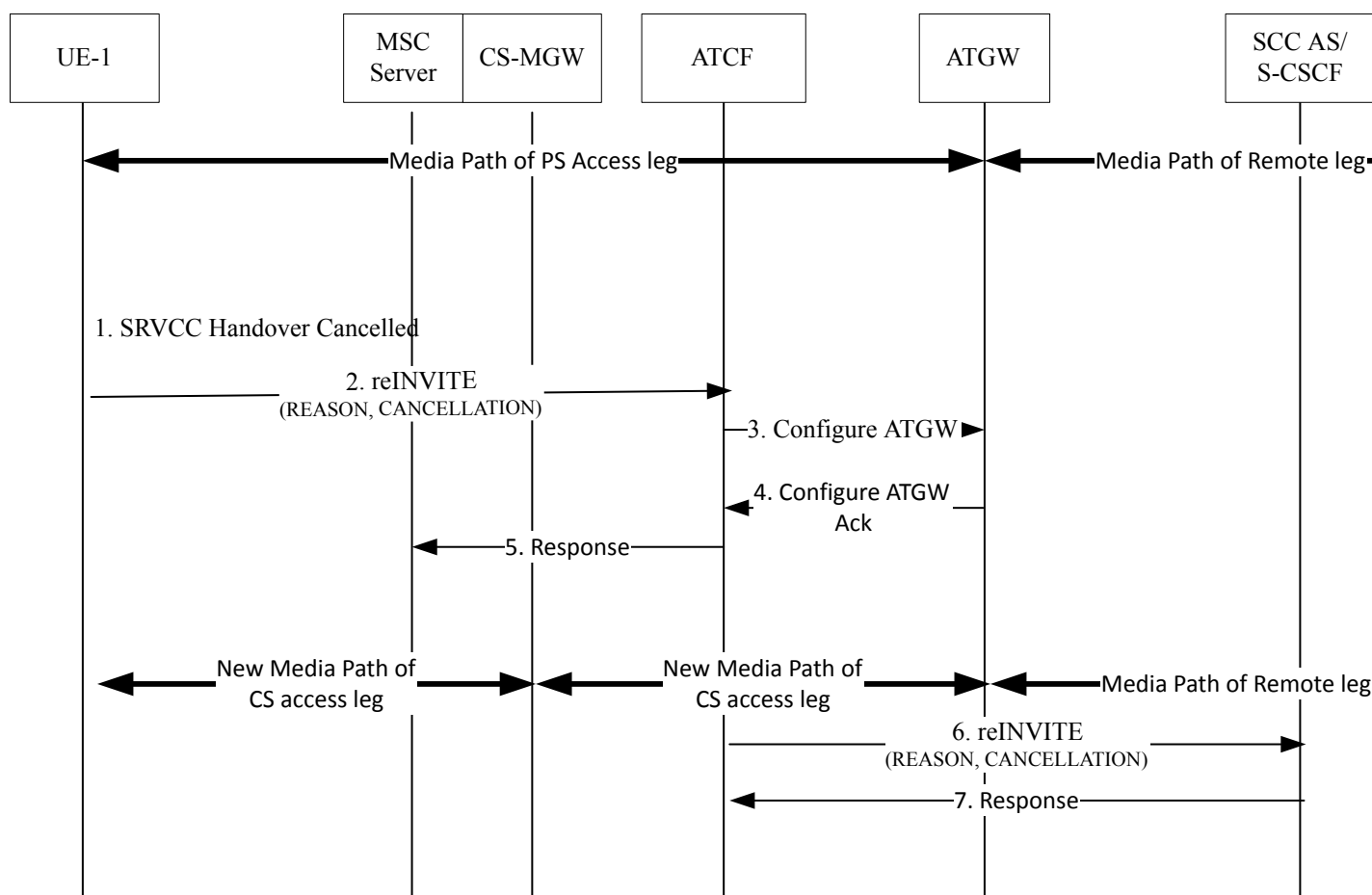
This proposed behavior, which requires no user configuration) is implemented in Version S-CX7.2.0 and later releases.

Related call flows are shown below.

BYE before handover INVITE



Cancellation after handover complete



## SRVCC PS-CS Access Transfer

This section describes various scenarios for transferring sessions between Packet-Switched (PS) and Circuit-Switched (CS) networks, providing details about the following cases:

- Mobile Switching Center (MSC) server-assisted mid-call feature supported by the SCC AS
- Call flows for the MSC server-assisted mid-call feature supported by the SCC AS
- Failure cases
- Confirmed dialog
- Early dialog

For PS-CS transfers in SRVCC, the ATCF performs several actions based on the STN-SR when it first receives the SIP INVITE. First, the ATCF determines the set of transferrable sessions. It defines this set as the SRVCC-transferrable sessions that are associated with a C-MSISDN matching the URI in the P-Asserted-Identity header field in the INVITE. The ATCF compares the INVITE to its set of transferrable sessions to see if the INVITE is part of the set; if so, it responds with 2xx response for the initial INVITE.

Active session transfers require the ATCF to act as a back-to-back user agent (B2BUA) if the session undergoing transfer has media anchored at the ATGW. In this case, the ATCF responds with a 200 OK to the INVITE; the OK carries an SDP answer with ATGW IP address and port information. Then the ATCF updates the following information and sends the INVITE to the remote UE:

- Original SDP offer is replaced with an SDP offer containing media information currently in use with the ATGW IP address and port information.
- The Request-URI with the ATU-SI associated with the session being transferred is inserted.
- The Target-Dialog header field with the dialog identifier of the session being transferred is inserted.

When the ATCF receives an SDP answer and if media is anchored at the ATGW for the session being transferred, the ATCF directs the ATGW to start sending and receiving media to/from the remote UE according to the newly received answer.

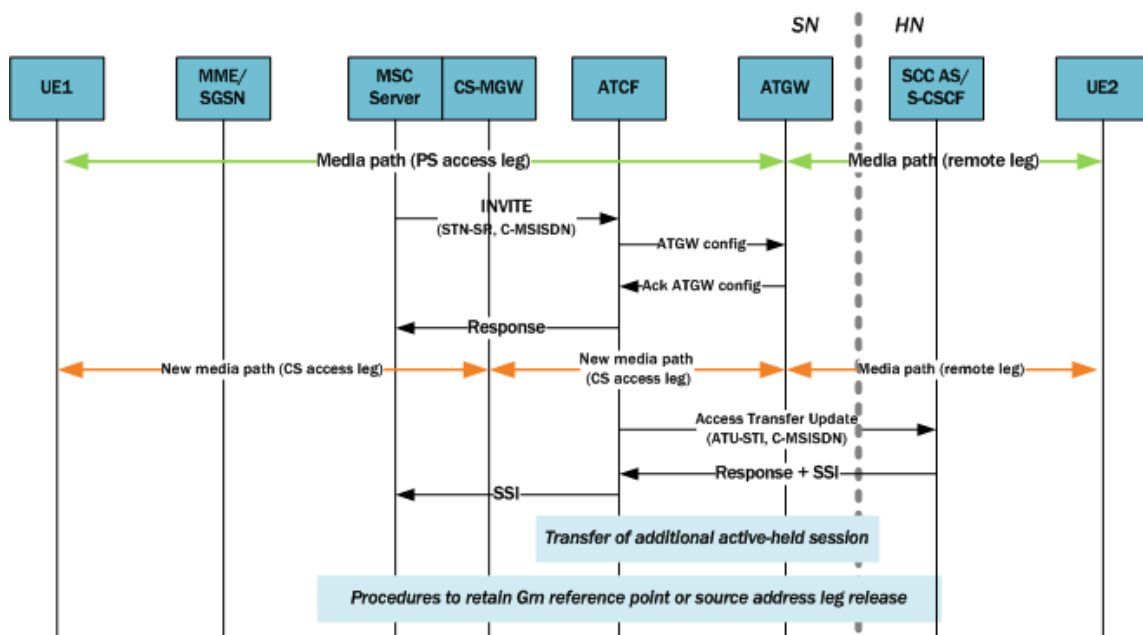
If it receives a BYE with a 503 Service Unavailable header on the source access leg, the ATCF retains session state information and ATGW resources associated with the session until either:

- The ATCF receives an INVITE request because of STN-SR, or
- An operator-defined time period elapses. The default value for the operator-defined time period is eight (8) seconds.

Thus, the session remains recognizable for SRVCC transfer for a time. Release of the session is also delayed by the ATCF's forwarding the BYE to the SCC AS. If the transferrable session cannot progress, the ATCF responds with a 404 Not Found. For transfer cases when only hold or alerting sessions exist and the session in question is an early dialog or a confirmed dialog with inactive media, the ATCF replaces the Request-URI received in the original INVITE with the ATU-STI associated with a session in the transferrable set before forwarding the request; this proxy-role behavior complies with the 3GPP standard.

### MSC Server-Assisted Mid-Call Feature Supported by CSS AS

The following scenario assumes an active session between UA1 and UA2, and that UA1 attaches to the CS domain. It is also assumed that the CS-MGW supports the codecs used for LTE voice calls, minimizing the likelihood that the ATCF will need to instruct the ATGW to insert codecs.



The MSC server initiates a transfer and (if supported) indicates its support for the MSC server-assisted mid-call feature. The MSC server also provides all voice codecs it supports in the Access Transfer message.

The following shows a sample INVITE from the MSC server to the ATCF. Note the following:

- The Request-URI header contains the STN-SR, as routed to the ATCF.
- The P-Asserted-Identity header reflects the C-MSISDN of the UE being served.
- The From header reflects the C-MSISDN of the serving UE.
- The SDP contains a pre-configured set of codecs the MGW supports.

```

INVITE tel: +1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-icid=visit1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel: +1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Accept-Contact: *;+g.3gpp.icsi-ref="urn:3Aurn-7%3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip:msc1.home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6>;+g.3gpp.icsi-ref="urn:3Aurn-7%3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20

```

Upon receiving the Access Transfer message, the ATCF identifies the correct anchored session and proceeds with the transfer of the most recently active session. Using a Configure ATGW message, the ATCF replaces the existing PS access leg media path information with new CS access leg media path information for the ATGW. However, the ATCF might instruct the ATGW to continue using the local port of the PS access leg media path. Once the ATGW acknowledges the ATCF's Configure message, the ATCF sends the MSC server an Access Transfer response and the media path is moved to the CS when receiving SDP information.

Voice break interruption starts either when media moves to the CS MGW (controlled by the MSC server enhanced for SRVCC) or when the UE relocates to the target—whichever comes first. When the UE tunes to the target or media switches to the CS MGW (whichever is last), the voice break interruption ends. The assumption is that media is switched to the CS MGW during the time to UE tunes to the target.

After receiving the Access Transfer message, the ATCF re-establishes communication with the SCC AS and updates the SCC AS. When the MSC server supports the mid-call feature, the ATCF also communicates this fact to the SCC AS. The ATU message initiates a new dialog between the ATCF and SCC AS, a dialog the CSS AS associates with the old dialog using the C-MSISDN. This new dialog is necessary because it replaces the old dialog set up over PS access, thereby assuring that if the PS user registration expires the new home leg will not be released or even affected.

The following shows a sample INVITE request the ATCF sends to the S-CSCF. Note the following:

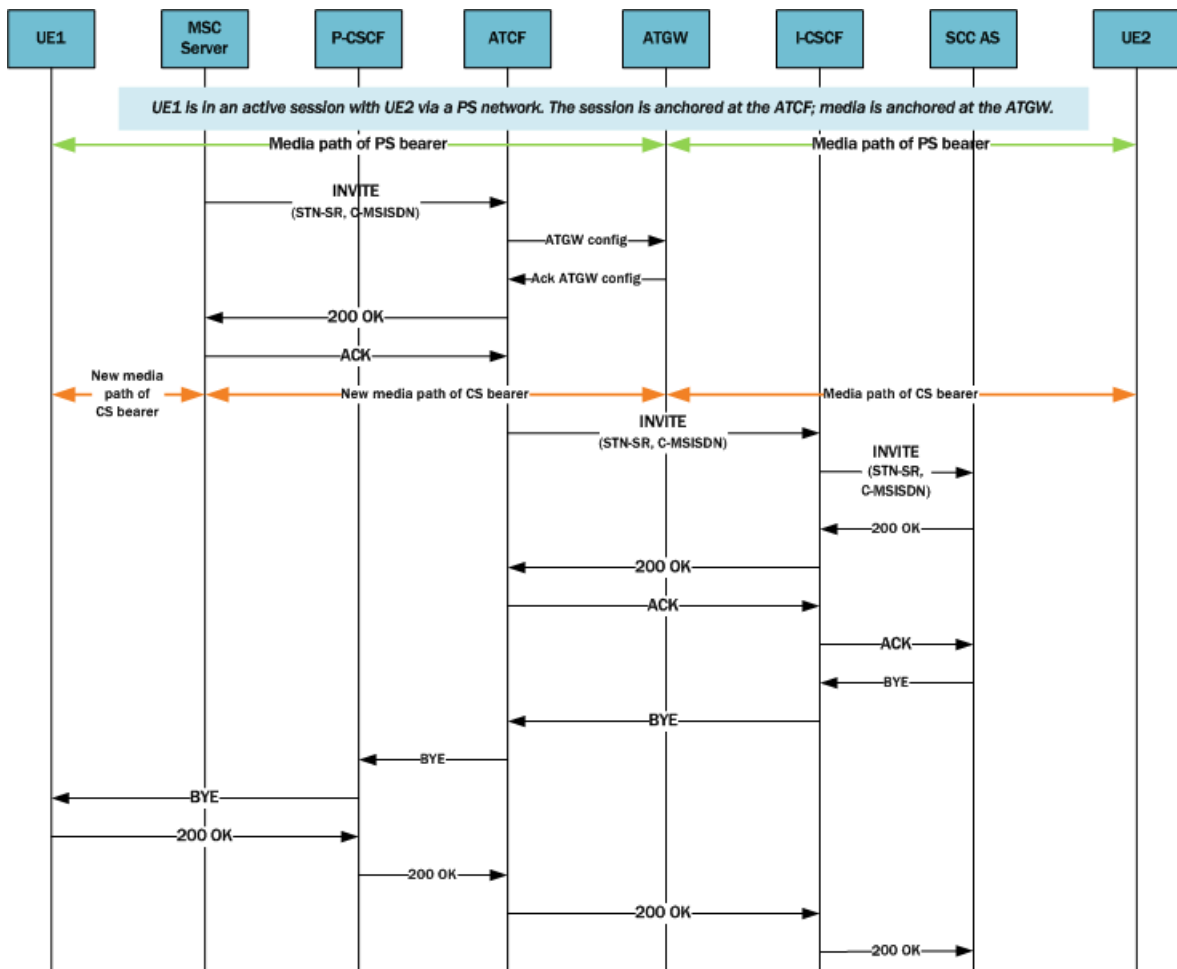
- The Request-URI header contains the ATU-STI, as routed to the SCC AS.
- The Target dialog header specifies the existing dialog is associated with this request.
- The P-Asserted-Identity header reflects the C-MSISDN of the UE being served.
- The From header reflects the C-MSISDN of the serving UE.
- The SDP reflects the media information at the ATGW.

```
INVITE sip:AUT-STI1@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-voi=visit1.net
Privacy: none
From: <tel:+1-237-555-9999>;tag=171828
To: <tel: +1-237-555-4444>
Call-ID: cb03a0s09a2sdfgklj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Target-Dialog: me03a0s09a2sdfgklj491777; to-tag=774321; from-tag=64727891
Accept-Contact: *;+g.3gpp.icsi-ref="urn:urn-7:3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip:msc1.home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6>;+g.3gpp.icsi-
ref="urn:urn-7:3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (...)

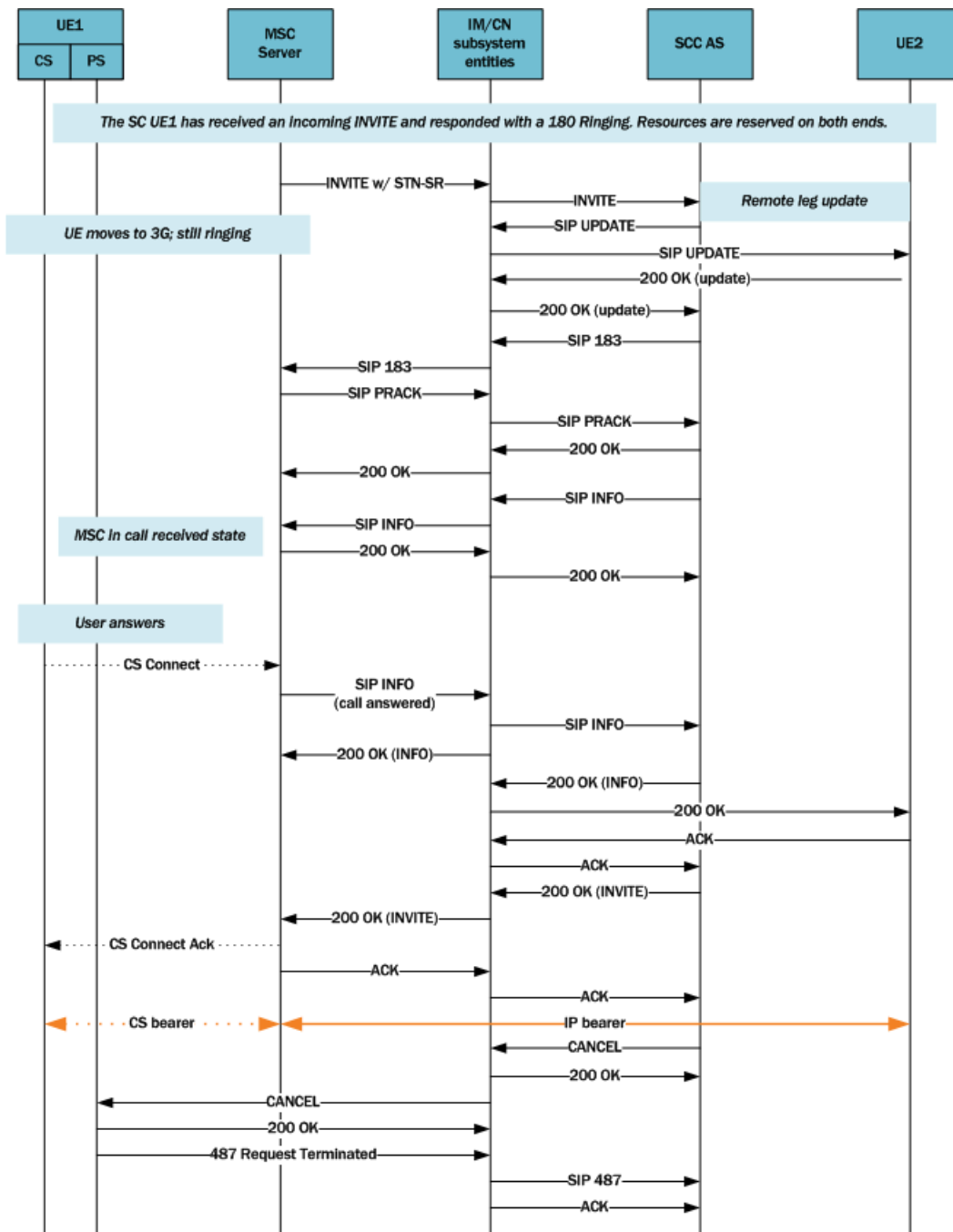
v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ggg
s=
c=IN IP6 5555::aaa:bbb:ccc:ggg
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

The SCC AS sends the ATCF a confirmation, including the session state information (SSI) if the SCC AS and the MSC server support the mid-call feature. The MSC server receives the SSI from the ATCF; the access leg for the control has moved to the CS access. When MSC server receives SSI for more multiple active or inactive speech sessions, it initiates transfer directed at the SSC AS for the additional sessions.

The following diagram shows a call session in the active phase for MSC server assisted mid-call feature when supported by the CSS AS.

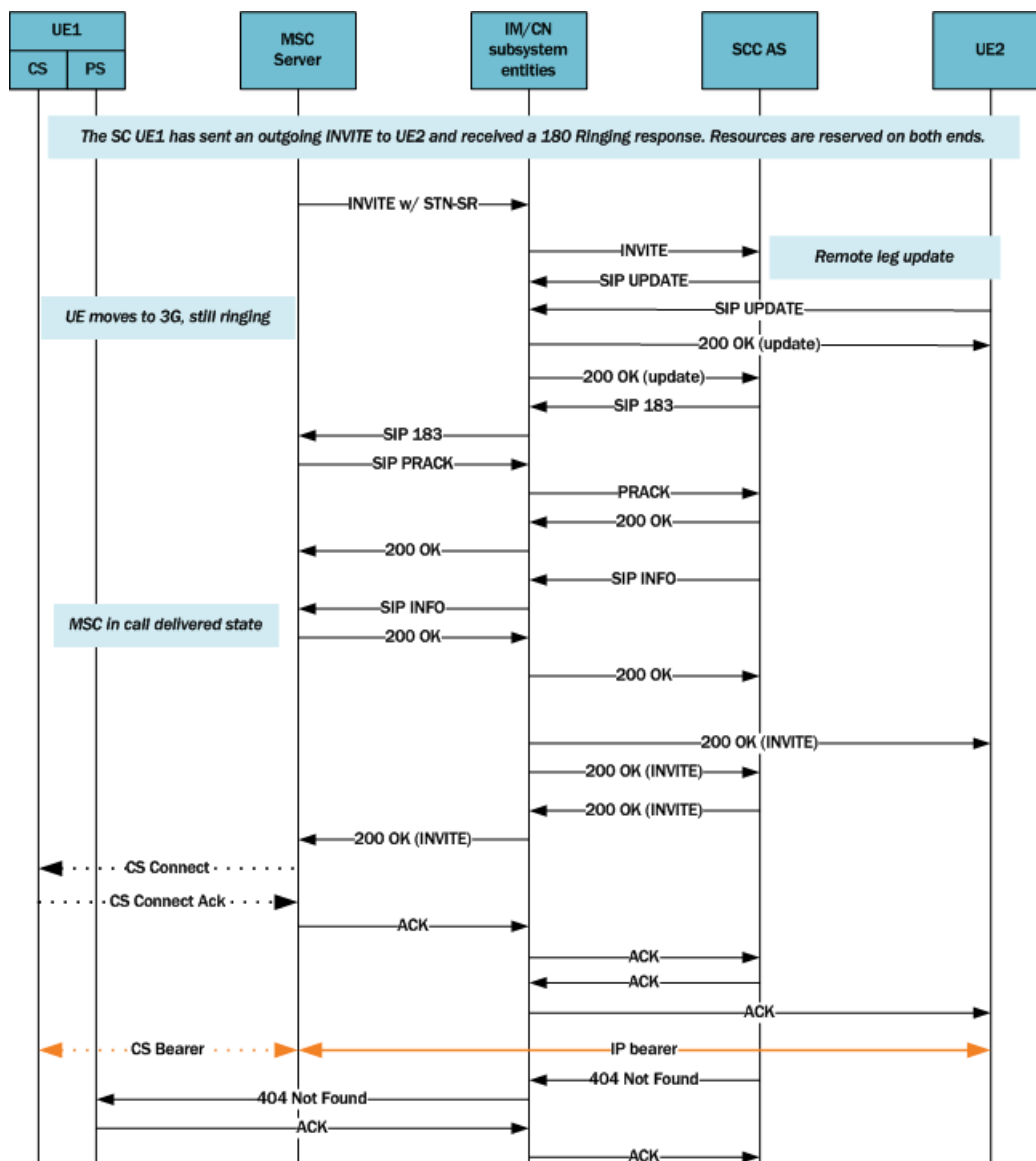


This diagram shows an incoming call session during the alerting phase.



This diagram shows an outgoing call session during the alerting phase.





## Failure and Cancellation

For cases of failure and/or cancellation, the ATCF behaves in accordance with TS 23.216 [3], Section 8:

- In the case of session failure before the MSC server initiates session transfer, the standard failover procedures (as defined in TS 23.401 [2]) apply and no further action is required by the UE.
- In the case of session failure after the UE receives the handover command, the UE attempts to return to the Universal Mobile Telecommunications System (UTRAN) or evolved Universal Mobile Telecommunications System (eUTRAN). The UE initiates signaling to transfer the session back, which the ATCF handles if the ATCF recognizes the session transfer.
- In the case when handover is cancelled, the UE—having received the handover cancellation message—begins the re-establishment procedure as though it needed to transfer the session to the eUTRAN/UTRAN. If the ATCF identifies this as a transfer session, it handles the session transfer back to the eUTRAN/UTRAN.

## Confirmed Dialogs

When an SC UE is engaged in one or more ongoing IMS sessions, it will send a re-INVITE containing:

- An SDP offer, including the media characteristics used in the existing dialog, and
- A Reason header field containing the protocol SIP and reason parameters cause with the value 487 (as specified in RFC 3326 [57]), with reason text reading either Handover cancelled or Failure to transition to CS domain.

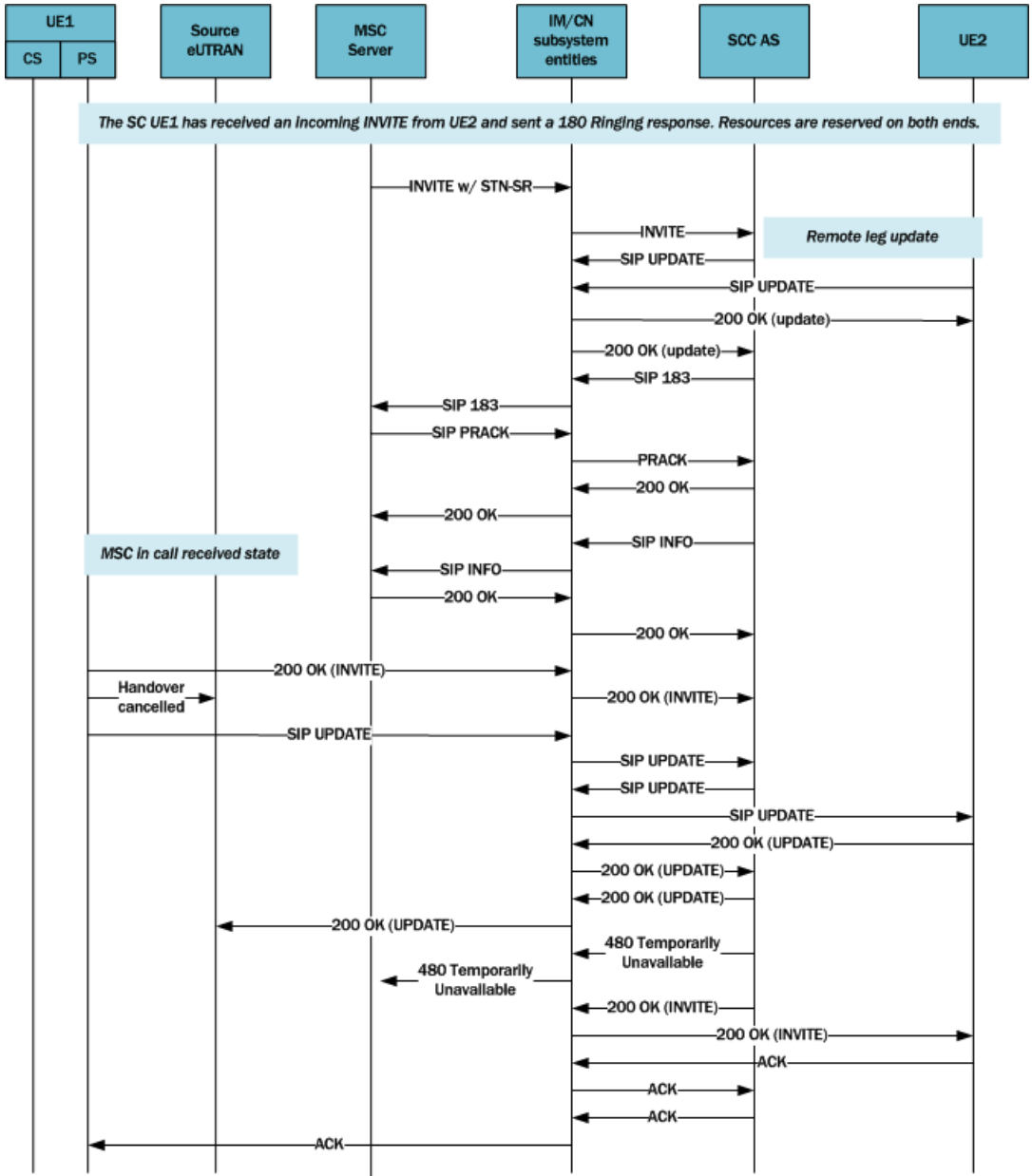
In all other ways, confirmed dialogs behave in accordance with TS 24.229 [2].

**Early Dialogs**

If the SC UE engages in a session which is in early dialog state, it will send a SIP UPDATE containing:

- An SDP offer, including the media characteristics used in the existing dialog, and
- A Reason header field containing the protocol SIP and reason parameters cause with the value 487 (as specified in RFC 3326 [57]), with reason text reading either Handover cancelled or Failure to transition to CS domain.

In all other ways, confirmed dialogs behave in accordance with TS 24.229 [2].



**SRVCC Handover Support in Alerting Phase**

Oracle now supports handovers between Packet-Switched (PS) and Circuit-Switched(CS) networks for calls in an alerting phase; that is, a 180 ringing response for the initial INVITE has been sent or received and the SIP final response has not been sent or received.

The behavior of the two anchoring points, ATCF and ATGW, is defined by 3GPP in Release 12 of Technical Specification TS 24.237. Oracle Communications developed these functional entities based on the initial version of

TS 24.237 Release 10, and has added the **sip-feature-caps** configuration element to align with Release 12. The element has three parameters:

- **state** — The value "enabled" triggers the feature and adds the Feature-Caps header to messages. The default value is "disabled".
- **atcf-management-uri** — identifies the feature capability indicator that will be used to transport the ATCF management URI. Possible values are "management" and "psi". The default value is "management". When the value is "management" and the value of state is "enabled", the Feature-Caps header "g.3gpp.atcf-mgmt-uri" is added and the value is the value of **atcf-psi-dn** in the **sip-config** configuration element. When the value is "psi" and the value of state is "enabled", the Feature-Caps header "g.3gpp.atcf-psi" is added and the value is the value of **atcf-psi-dn** in the **sip-config** configuration element.
- **atcf-alerting** — The value "enabled" adds the Feature-Caps header to messages and turns on the alerting feature. The default value is "disabled".

To ensure that calls in an alerting phase are transferred between PS and CS networks, set the value of **state** to "enabled", set the value of **atcf-management-uri** to "management", and set the value of **atcf-alerting** to "enabled". The Feature-Caps header is then added with:

- g.3gpp.atcf : value is configured STN-SR in **sip-config**
- g.3gpp.atcf-mgmt-uri: the value is **atcf-psi-dn** in **sip-config**
- g.3gpp.atcf-path: value is the ATCF URI for terminating requests
- g.3gpp.mid-call capability indicator
- g.3gpp.srvcc-alerting capability indicator

For more information, refer to the SIP Feature Capabilities section.

## Accounting

eSRVCC calls involve two SIP sessions, the accounting records for which need clear association. To correlate the records, the Oracle Communications Session Border Controller performs these actions for eSRVCC calls:

- When it detects a session handover, the system generates an Interim accounting record for the initial SIP session. This way, the records provide the exact time of the handover (i.e., the timestamp of the Interim record).
- The Oracle specific AVP Generic-ID will appear in Start, Interim, and Stop records for the handover SIP session. The Generic-ID contains the Call ID of the initial SIP session, which helps to correlate the two SIP sessions.
  - For RADIUS accounting records, refer to Oracle specific VSA 40.
  - For DIAMETER accounting records, refer to Oracle specific VSA 30.
- The Stop record for the initial SIP session will not have media flow information because the media session is considered part of the handover SIP session.
- If you are using QoS, the Stop records for the handover SIP session reflects the cumulative QoS statistics for both the initial SIP session and the handover SIP session.

## External Bandwidth Management

If you are using the Oracle Communications Session Border Controller's external bandwidth management for eSRVCC calls, note these considerations:

- At the time of handover and if the new handover realm has the same external policy server configured as the initial SIP session, the Rx will continue seamlessly with the same DIAMETER session identifier directed toward the policy server. From the perspective of the policy server, the same session is simply continuing.
- At the time of handover and if the new handover realm does not have an external policy server configured, policy server services will stop.

## ATCF Configuration

ATCF functionality requires configuration in the **sip-config** and **sip-interface** configuration elements.

1. Access the **sip-config** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

2. **atcf-stn-sr**—Enter the value of the Session Transfer Interface, Single Radio (STN-SR). Your entry will resemble this example: tel:+1-237-555-9999. This value will be included in the g.3gpp.atcf media feature tag that the ATCF allocates in the REGISTER message.
3. **atcf-psi-dn**—Enter the value to use for the Public Service Identity Domain Name (PSI-DN). Your entry will resemble this example: sip:atcf.visited2.com. If configured, this value will be included in the g.3gpp.atcf media feature tag that the ATCF allocates in the REGISTER message. If you leave this parameter blank, the Oracle Communications Session Border Controller will set this value to the SIP interface access.
4. **atcf-route-to-sccas**—If you leave this parameter set to disabled (default), the handover update, an INVITE, is routed to the IMS Core. If you set this parameter to enabled, the Oracle Communications Session Border Controller will route the handover update directly to the Service Centralization and Continuity Application Server (SCC-AS).
5. Type **done** to save your configuration.

Continue to and select the existing sip-interface configuration element targeted for this ATCF configuration:

```
ACMEPACKET(sip-config)# exit
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)# select
<RealmID>:
1: public

selection: 1
ACMEPACKET(sip-interface)#
```

6. **sip-atcf-feature**—Change this parameter from **disabled** (default) to **enabled** to turn on ATCF functionality for the ingress SIP interface.
7. Type **done** to save your configuration.

## LIR LIA Lookup to Home Subscriber Server from I-BCF

---

The Oracle Communications Session Border Controller, acting as an I-BCF, can perform a Location Information Lookup (LIR) to a Home Subscriber Server (HSS) to find the S-CSCF where to forward a request message. LIR/LIA messages are part of the Diameter Cx interface, which is used to communicate with HSS.

In an IMS environment, a user registers with S-CSCF and that S-CSCF assigns itself with an HSS as the serving registrar for that user. When a call needs to be routed to the user, an I-BCF performs a location information lookup to the HSS to determine the S-CSCF for that user and route the request toward that S-CSCF.

### Home Subscriber Server

The Oracle Communications Session Border Controller is configured with an HSS configuration element called home subscriber server. When the Oracle Communications Session Border Controller receives an INVITE from a user, a local policy routing lookup is performed.

The home subscriber server configuration element is configured with a name, which is referenced in a local policy next-hop parameter. It also includes, a state, IP address, port, and realm.

### P-Acme-Serving Parameter Creation

The Oracle Communications Session Border Controller can insert a P-Acme-Serving parameter into the Route header of a message. This parameter is inserted when the **add-lookup-parameter** parameter is enabled in the home subscriber server configuration element and the Oracle Communications Session Border Controller queried that HSS to find the message's destination.

The P-Acme-Serving parameter can be helpful for informing other Oracle Communications Session Border Controllers that this system is acting as the UA's I-BCF and that the HSS lookup has been done.

## HSS Watchdog Keepalives

The home subscriber server configuration element contains a **watchdog-ka-timer** parameter that sets a time to send DWR messages to the HSS. This is similar to external policy server Diameter Heartbeat functionality. When 3/4 of this timer elapses, the Oracle Communications Session Border Controlleri begins disconnection processes. DWR message creation is suspended when any other activity with the HSS occurs within watchdog-ka-timer seconds. DWR creation resumes watchdog-ka-timer seconds if all other traffic stops.

## Local Policy

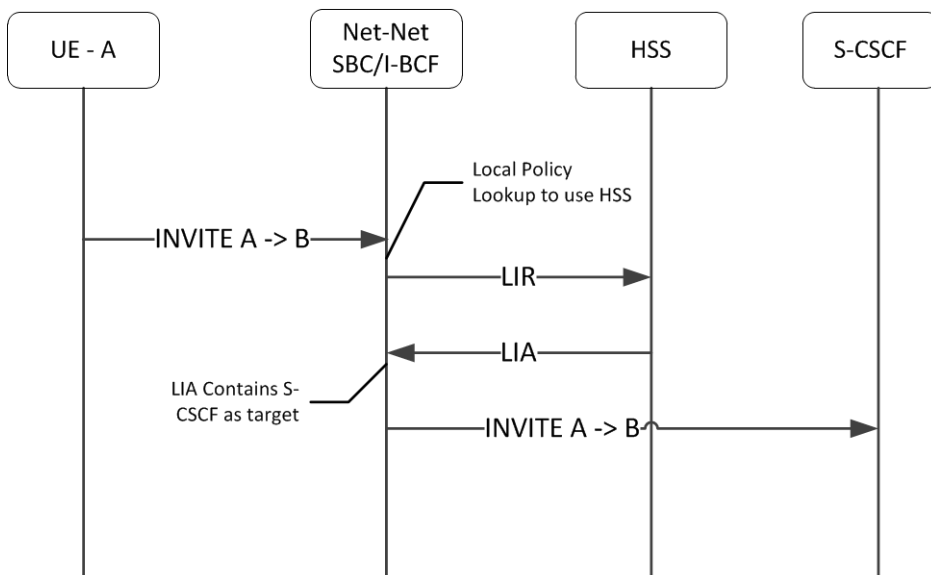
The Oracle Communications Session Border Controlleri performs all expected processing on an incoming message and begins a local policy lookup. When it encounters a policy attribute with a next hop in the form of `cx:element-name`, it will perform an LIR/LIA transaction over the Cx Diameter interface to the home subscriber server referenced as `element-name`.

The Oracle Communications Session Border Controlleri forwards the initial INVITE to the address in the Server-Name AVP returned by the HSS in the LIA message.

## LIR LIA Transaction

The LIR includes the Public-User-Identity AVP containing the request URI from the SIP request. The HSS responds with the assigned S-CSCF server (a Oracle Communications Session Border Controlleri) for this PUID. The answer is the form of a Location Info Answer (LIA) and is in the Server Name AVP. If the S-CSCF specified in this AVP is not the current Oracle Communications Session Border Controlleri, then the INVITE is forwarded to the address specified in the LIR.

If the S-CSCF returned in the LIR is this Oracle Communications Session Border Controlleri, then the AoR from the request URI is found in the registration cache and the message is forwarded to that endpoint. When the registration cache entry does not exist or is invalid, local policy processing continues with the next policy-attribute following "stop-recurse" rules. If there are no other routes, then a 404 is sent to the UA who sent the INVITE.



## LIR Format

The LIR contains the following information:

```

Vendor-Id: 9148
Supported-Vendor-Id: 10415
Supported-Vendor-Id: 13019
Vendor-Specific-Application-Id (grouped) :
  Vendor-ID: 10415
  Vendor-ID: 13019
  
```

```
Auth-Application-ID: 16777216
Origin-Host: IP Address:port of the system's source interface.
Origin-Realm: Realm-name where the system talks to this HSS.
Destination-Host: IP address of the HSS.
Public-Identity: Request URI of SIP Request
```

### LIA Format

The LIA is expected to contain:

```
Server-Name: S-CSCF address to forward this message
```

## Home Subscriber Server Configuration

To configure a home subscriber server (HSS):

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router path.

```
ACMEPACKET(configure)# session-router
```

3. Type **home-subscriber-server** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# home-subscriber-server
ACMEPACKET(home-subscriber-server)#
```

4. **name**—Enter the name for this home subscriber server configuration element to reference from other configuration elements.
5. **state**—Set this to **enabled** to use this configuration element.
6. **address**—Enter the IP address of this HSS.
7. **port**—Enter the port number on the HSS where to connect. The default value is 80.
8. **realm**—Enter the realm name where this HSS exists.
9. **watchdog-ka-timer**—Enter the period of time to send DWRs to the HSS.
10. **add-lookup-parameter**—Set this to enabled for the Oracle Communications Session Border Controller to add a P-Acme-Serving parameter to the Route header.
11. Type **done** when finished.

## Local Policy Configuration

To direct a policy attribute to retrieve a next hop from an HSS via an LIR lookup:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router path.

```
ACMEPACKET(configure)# session-router
```

3. Type **local-policy** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# local-policy
```

4. Choose an existing local-local object with the select command.

```
ACMEPACKET(local-policy)# select
<source-realm>:
1: realms 'realm01'; from *; to *
selection: 1
```

5. Type **policy-attributes** and press Enter. If you are adding this feature to a pre-existing local policy configuration, you will need to select and edit your local policy.

```
ACMEPACKET(local-policy) # policy-attributes
ACMEPACKET(policy-attributes) #
```

6. **next-hop**—Set this parameter to **cx:name-of-HSS-configuration** for this policy attribute to lookup routes by querrying the named HSS via an LIR.
7. Type **done** and save and activate your configuration.

## Statistics

The show home-subscriber command displays detailed information about HSS transactions. For example:

```
ACMEPACKET## show home-subscriber-server
17:54:58-186
HSS Status
-- Period -- ----- Lifetime -----
Active High Total Total PerMax High
Client Trans 0 0 0 12 4 1
Server Trans 0 0 0 1 1 1
Sockets 1 1 0 1 1 1
Connections 1 1 0 1 1 1
----- Lifetime -----
Recent Total PerMax
LIR 0 0 0
Sent Req Accepted 0 11 3
Sent Req Rejected 0 0 0
Sent Req Expired 0 0 0
Sent Req Error 0 0 0
Internal Errors 0 0 0
```

Note the following statistics provided for Recent and Lifetime periods:

- LIR—Number of LIR requests sent
- Sent Req Accepted—Number of requests for which we got success response (2xxx)
- Sent Req Rejected—Number of permanent failures (5xxx)
- Sent Req Expired—Number of requests for which there was no response
- Sent Req Error—Number of protocol errors/bad requests (1xxx, 3xxx, 4xxx)

## Emergency Access Transfer Function

The Emergency Access Transfer Function (EATF) is a logical, functional service defined in 3GPP TS 23.167, *IP Multimedia Subsystem (IMS) Emergency Sessions*, and TS 23.237, *IP Multimedia Subsystem (IMS) Service Continuity*; Stage 2. The EATF, essentially a special-purpose B2BUA, anchors emergency calls to enable access transfer between packet-switched and circuit-switched networks during eSR-VCC procedures when the LTE equipment is moving outside LTE coverage to either a 2G or 3G carrier network. Similar to the Access Transfer Control Function (ATCF) and ATGW (Access Transfer Gateway), the EATF is always located in the visited network when the user equipment is roaming.

Lacking this capability, the LTE equipment would be forced to re-establish the emergency session in the circuit-switched network through the legacy accesses (2G or 3G).

When the LTE equipment initiates a packet-switched emergency session, the INVITE is sent to the EATF thru the P-CSCF/E-CSCF (collocated on the SBC). This original INVITE is identified as an emergency session by the A-SBC/P-CSCF because it contains either an emergency short number (112, 991, and so forth) or an emergency service URN such as urn:service:sos.fire.

In the event that handoff to a circuit-switched network is required, the Mobile Switching Center (MSC) server initiates the transfer with a SIP INVITE containing an E-STN-SR (Emergency Session Transfer Number for Single Radio VCC) in the Request URI of the INVITE. Each network has a single E-STN-SR, essentially the telephone number of the EATF service, that is used exclusively for emergency session transfer access. The MSC directs the INVITE to the I-CSCF, which, in turn, which forwards the request directly to the EATF.

The EATF checks the E-STN-SR to determine that handoff to the circuit-switched network is requested and proceeds with the access transfer of the active session. The EATF associates the received SIP INVITE with an existing SIP session already anchored at the EATF using the instance-id feature tag. The EATF then sends a re-INVITE to the E-CSCF, which terminates the emergency session.

Once the session modification procedures are complete, as indicated by the reception of the SIP ACK request from the target access leg, the source access leg, previously established via IMS, is released.

### Enabling EATF Capability

Use the following procedure to enable EATF operations.

1. Use the following procedure to move to sip-config configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

2. Use the **eatf-stn-sr** parameter to enable EATF operation and to provide E.164 telephone number of the EATF service.

Provide the E.164 telephone number as either a tel: or a sip: URI. When confirming the E-STN-SR value, the match succeeds based solely on the E.164 portion regardless of the URI type.

```
ACMEPACKET(sip-config)# eatf-stn-sr tel:+1-237-555-9999
ACMEPACKET(sip-config)#
```

3. Use **done** and **exit** to complete configuration.

### Monitoring EATF ATCF Sessions

The existing **show sipd srvc** command has been enhanced to provide counts of successful and cancelled EATF sessions.

```
ACMEPACKET# show sipd srvc
SIP Status          -- Period --  ----- Lifetime -----
                   Active  High  Total      Total  PerMax   High
ATCF success        -     -     1          1      1
ATCF failure        -     -     -          -      -
ATCF cancellation   -     -     -          -      -
EATF success        -     -     1          1      1
EATF cancellation
```

### Restoration Procedures

---

When functioning as a P-CSCF within an IMS network, an SBC is required to conform to Restoration Procedures as defined in 3GPP TS 23.380, *IMS Restoration Procedures*, and 3GPP TS 24.229, *IP Multimedia Call Control Protocol Based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP)*.

TS 23.380 defines a set of standardized procedures for automatic restoration of connectivity after node or link failure. Section 4.4.3 of TS 23.380 imposes a specific injunction requiring that if the UE initiates an originating service request different from SIP REGISTER and the P-CSCF is unable to contact the S-CSCF in the Route, the P-CSCF shall return a specific error response to the UE to trigger a new registration. Section 5.2.6.3.2A of the related 3GPP TS 24.229 defines the structure and content of the required error response.

In the event of a P-CSCF failure to forward a request to the next hop device (whatever the reason), TS 24.229 mandates that the P-CSCF shall reject the call by returning a 504 (Server Time-out) response to the user endpoint. The 504 response must contain:

- a Content-Type header field with the value "application/3gpp-ims+xml"
- a P-Asserted-Identity header field set to the value of the SIP URI of the P-CSCF included in the Path header field during registration of the user whose UE sent the request causing this response



- an attached XML node-set (specified in Section 4.4.1 of 3GPP TS 24.615, *Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem*, containing a child element that conveys the reason for requiring a new registration

After issuing the 504 Server Time-out, the P-CSCF must reject any messages from the user endpoint except for registration requests.

## 3GPP IM CN Subsystem XML Body

The attached XML node-set is constructed as shown below.

```
<3gpp-ims version="1">
  <alternative-service>
    <type>
      <restoration/>
    </type>
    <reason>User configurable value</reason>
    <action>
      <initial-registration/>
    </action>
  </alternative-service>
</3gpp-ims>
```

## Enabling Restoration Procedures

Use the following procedure to enable compliance with IMS Restoration Procedures mandated by 3GPP TS 23.380 and TS 24.229. You can enable compliant behavior on individual SIP interfaces. By default, TS 23.30/TS 24.229 compliance is disabled on all SIP interfaces.

1. From superuser mode, enter the following ACLI command sequence to access sip-interface configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

2. Use the select ACLI command to select the target SIP interface, that is the SIP interface to be brought into compliance with TS 23.30/TS 24.229 requirements.

3. Use the pscf-restoration ACLI parameter to enable TS 23.30/TS 24.229 compliance.

Enable compliance by assigning a string value to this parameter. The string provides the contents of the <reason> element.

```
ACMEPACKET(sip-interface)# pscf-restoration "Restoration procedures in force"
ACMEPACKET(sip-interface)#
```

4. Use **done** and **exit** to complete enabling restoration procedures on the current SIP interface.
5. As required, repeat steps 1 through 4 to enable Restoration Procedures on other SIP interfaces.



---

# Transcoding

## Introduction

---

Transcoding is the ability to convert between media streams that are based upon disparate codecs. The Oracle Communications Session Border Controller supports IP-to-IP transcoding for SIP sessions, and can connect two voice streams that use different coding algorithms with one another.

This ability allows providers to:

- Handle the complexity of network connections and the range of media codecs with great flexibility
- Optimize bandwidth availability by enforcing the use of different compression codecs
- Normalize traffic in the core network to a single codec
- Enact interconnection agreements between peer VoIP networks to use approved codecs

By providing transcoding capabilities at the network edge rather than employing core network resources for the same functions, the Oracle Communications Session Border Controller provides cost savings. It also provides a greater degree of flexibility and control over the codec(s) used in providers' networks and the network with which they interconnect.

In addition, placing the transcoding function in the Oracle Communications Session Border Controller and at the network edge means that transcoding can be performed on the ingress and egress of the network. The Oracle Communications Session Border Controller transcodes media flows between networks that use incompatible codecs, and avoids back-hauling traffic to a centralized location, alleviating the need for multimedia resource function processors (MRFPs) and media gateways (MGWs) to support large numbers of codecs. This maximizes channel density usage for the MRFPs and MGWs so that they can reserve them for their own specialized functions.

## Transcoding Hardware

A transcoding Network Interface Unit (NIU) provides the DSP resources that enable transcoding on the Oracle Communications Session Border Controller.

- The Acme Packet 6300 can accept 1-24 transcoding modules to provide increasing transcoding capacity.
- The Acme Packet 3820 and the Acme Packet 4500 can accept 1-12 transcoding modules to provide increasing transcoding capacity.

### Transcoding Capacity

Transcoding capacity depends on the following:

- Codecs used for transcoding

## Transcoding

- Number of transcoding modules installed in the system. Capacity scales linearly with each extra transcoding module installed.

### Transcodable Codec Details

The following table lists the supported codecs, bit rates, RTP payload type, default ptime, and supported ptimes.

Codec	Supported Bit Rate (kbps)	RTP Payload Type	Default Ptime (ms)	Supported Ptime (ms)
G.711 PCMU	64	0	20	10, 20, 30, 40, 50, 60
G.711 PCMA	64	8	20	10, 20, 30, 40, 50, 60
G.722	48, 56, 64	9	20	10, 20, 30, 40
G.723.1	5.3, 6.3	4	30	30, 60, 90
G.726	16, 24, 32, 40	2, 96-127	20	10, 20, 30, 40, 50
iLBC	13.33	96-127	20	20, 30, 40, 60
	15.2	96-127	30	20, 30, 40, 60
G.729/A/B	8	18	20	10, 20, 30, 40, 50, 60, 70, 80, 90
AMR	4.75, 5.15, 5.90, 6.70, 7.40, 7.95, 10.2, 12.2	96-127	20	20, 40, 60, 80, 100
AMR-WB (G.722.2)	6.6, 8.85, 12.65, 14.25, 15.85, 18.25, 19.85, 23.05, 23.85	96-127	20	20, 40, 60, 80, 100
GSM FR	13	3	20	20
T.38	4.8, 9.6, 14.4	N/A		10, 20, 30

The following table lists the supported codecs, bit rates, RTP payload number, default ptime, and supported ptimes for codecs available only on the Acme Packet 6300.

Codec	Supported Bit Rate (kbps)	RTP Payload Type	Default Ptime (ms)	Supported Ptime (ms)
EVRC	0.8, 4.0, 8.55	96-127	20	20, 40, 60
EVRC0	0.8, 4.0, 8.55	96-127	20	20
EVRC1	4.0, 8.55	96-127	20	20, 40, 60, 80, 100
EVRCB	0.8, 2.0, 4.0, 8.55	96-127	20	20
EVRCB0	0.8, 2.0, 4.0, 8.55	96-127	20	20
EVRCB1	4.0, 8.55	96-127	20	20, 40, 60, 80, 100

### T.38 FAX Support

This release supports T.38 FAX relay (Version 0) conversion to T.30 over G.711 and supports FAX modulation schemes up to 14400 kbps V.17. The initial release does not support V.34 modulation.

## Session Licensing

AMR/AMR-WB and EVRC/EVRC-B audio codecs require extra licenses while all other codecs are implicitly licensed by installation of a transcoding NIU. These codecs are under royalty agreements which necessitates their own special licenses.

Licenses for AMR/AMR-WB and EVRC/EVRC-B transcoding sessions are installed by groups of 25.

## Transcoding Configuration

The Oracle Communications Session Border Controller performs transcoding functions— allowing the entities with incompatible codecs to communicate with each other— between two call legs. The two endpoints can be located in one or two realms or networks. The Oracle Communications Session Border Controller decides to transcode a call by evaluating messages in the SDP offer-answer transaction with respect to system configuration. An SDP offer can be in a SIP message such as an INVITE or a reINVITE, and contains information about the codecs the offerer would like to use. The answerer answers the SDP offer with its own set of supported codecs. reINVITEs are treated as new negotiations, with respect to the actual SDP offerer and answerer. The Oracle Communications Session Border Controller can manipulate an SDP message by reordering codec preference, and by adding and deleting codecs.

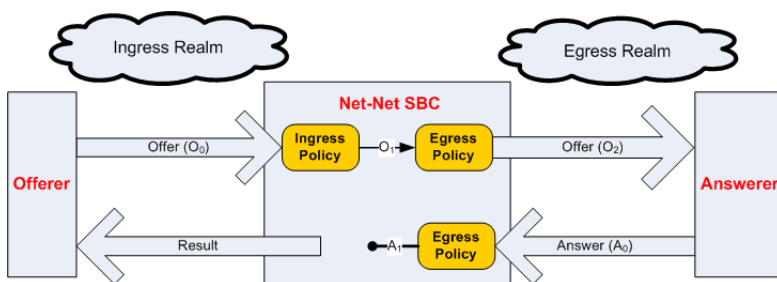
## Transcoding Processing Overview

Transcoding processing is viewed in terms of the ingress and egress realms. The ingress realm is where the SDP offer is received by the Oracle Communications Session Border Controller. The egress realm is where the SDP offer is sent, and where the SDP answer is expected to be received from (i.e., the answerer's realm). A call is defined as transcodable if an egress or ingress policy exists for the session and if the session is not subject to media release, as specified in the realm configuration.

To understand the details of transcoding processing, refer to the following diagram. An SDP offer, O0, is received by the Oracle Communications Session Border Controller in the ingress realm. The ingress codec policy is applied here and the SDP offer is transformed into O1. O1 is then passed to and processed by the egress codec policy. This SDP message is then forwarded to the answerer as O2. The answerer replies with A0 to the Oracle Communications Session Border Controller, which is subjected to the egress codec policy again and transformed into A1.

When policy dictates not to transcode the call, the Result SDP sent back to the offerer is based on the common codecs shared between A1 and O1. The Oracle Communications Session Border Controller first constructs the list of codecs that are present in both in O1 and A1. Then, the Oracle Communications Session Border Controller maintains the codec order from A1 for the Result as it is sent to the offerer.

When policy dictates to transcode the call, the top transcodable codec in O1 is used in the ingress realm and the top non-signaling codec in A1 is used in the egress realm.



## Defining Codec Policies

The following definitions are required for understanding transcoding processing:

**DTMFable Codecs**—Uncompressed codecs that are capable of properly transmitting a DTMF waveform. The only codecs designated as DTMFable are PCMU and PCMA.

**FAXable Codecs**—Uncompressed codecs that are capable of properly transmitting a T.30 waveform. The only codecs designated as FAXable are PCMU and PCMA.

Signaling Codecs—Non-audio codecs that are interleaved into a media stream but cannot be used on their own. The only codecs designated as Signaling Codecs by the Oracle Communications Session Border Controller are telephone-event and CN (comfort noise).

Disabling an m= line—This is in reference to an m= line in an SDP message. It means setting the m= line's port to 0 (RFC 3264). Enabling an m= line means it has a non-zero port. The m= line's mode attribute (sendrecv/inactive/rtponly, etc) is not considered.

A codec policy is created by configuring the following information:

- Which Codecs are allowed and which are denied in a realm.
- Which Codecs should be added to the SDP m= lines for an egress realm.
- The preferred order of codecs to indicate in an SDP m= line.
- The packetization time which should be enforced within a realm.

### Ingress Policy

Incoming SDP is first subject to the ingress codec policy. If no codec policy is specified in the realm config for the ingress realm, or the m= lines in the SDP offer are disabled (by a 0 port number), the SDP is transformed to O1 unchanged.


The ingress codec policy first removes all un-allowed codecs, as configured in the allow-codecs parameter by setting their port to 0 or removing the codecs from a shared m-line. For example, if two codecs share an m-line and one of them is un-allowed, the resulting m-line will not include the un-allowed codec and its attribute lines will be removed. If a single codec is used, the resulting m-line will include the codec, but its port will be set to 0 and its attribute lines will remain. Next, the remaining codecs are ordered with the **order-codecs** parameter. Ordering is when the codec policy rearranges the codecs in the SDP m= line. This is useful to suggest the codec preferences to impose within the egress realm. O1 is then processed by the egress codec policy after a realm is chosen as the destination.

In practical terms, the ingress policy can be used for filtering high-bandwidth codecs from the access realm. It can also be used for creating a suggested, prioritized list of codecs to use in the ingress realm.

### Egress Policy

The Oracle Communications Session Border Controller applies egress codec policy to the SDP that has already been processed by ingress policy. The egress policy is applied before the SDP exits the system into the egress realm. If no egress codec policy is defined, or the SDP's m= lines are disabled (with a 0 port), the SDP is passed untouched from the ingress policy into the egress network.

The egress codec policy first removes all un-allowed codecs in the **allow-codecs** parameter (<codec>:no). Codecs on the **add-codecs-on-egress** list are not removed from the egress policy regardless of the how the **allow-codecs** parameter is configured. If the result does not contain any non-signaling codecs, theptime attribute is removed from the SDP. Codecs not present in O1 that are configured in the **add-codecs-on-egress** parameter are added to the SDP, only if O1 contains one or more transcodable codecs.

 **Note:** Transcoding can only occur for a call if you have configured the add-codecs-on-egress parameter in an egress codec policy.

If codecs with dynamic payload types (those between 96 and 127, inclusive) are added to the SDP, the lowest unused payload number in this range is used for the added codec.

The following rules are also applied for egress policy processing:

If O1 contains at least one transcodable codec, the codecs listed in the Egress policy are added to the SDP.

- telephone-event, as configured in add-codecs-on-egress will only be added if O1 contains at least one DTMFable codec.
- T.38, as configured in add-codecs-on-egress will only be added if there is no T.38 and there is at least one FAXable codec (G711Fall Back (FB)) in O1. T.38 will then be added as a new m= image line to the end of SDP.

If G711FB is not allowed in the egress policy, the Oracle Communications Session Border Controller disables the m= line with the FAXable codec. Otherwise if G711FB is allowed, pass it through the regular offer processing allowing/adding only FAXable codecs.

- G711FB, as configured in `add-codecs-on-egress` will only be added if there is no G711FB and there is T.38 in O1. G711FB will then be added as a new `m=` audio line to the end of SDP.

If T.38 is not allowed in the egress policy, the Oracle Communications Session Border Controller disables the `m=` image line. Otherwise if T.38 is allowed, pass it through the regular offer processing.

If the result of the egress policy does not contain any non-signaling codecs, audio or video, the `m=` line is disabled, by setting the port number to 0.

The `m=` line is next ordered according to rules for the `order-codecs` parameter.

Finally, all attributes, `a=` lines, `ptime` attribute, and all other unrecognized attributes are maintained from O1. Likewise, appropriate attributes for codecs added by the `add on egress` parameter are added to SDP. Finally, `rtpmap` and `fntp` parameters are retained for codecs not removed from the original offer. The result of all this is O2, as shown in the overview diagram.

In practical terms, codec policies can be used to normalize codecs and `ptime` in the core realm where the network conditions are clearly defined.

Codec policies can also be used to force the most bandwidth-conserving codecs anywhere in the network.

### Post Processing

If any errors are encountered during the Ingress and Egress policy application, or other violations of RFC3264 occur, the call is rejected. If O2 does not contain any enabled `m=` lines at the conclusion of the initial call setup, the call is rejected. If O2 does not contain any enabled `m=` lines at the conclusion of a `reINVITE`, the `reINVITE` is rejected and the call reverts back to its previous state.

## Codec Policy Definition

Codec policies describe how to manipulate SDP messages as they cross the Oracle Communications Session Border Controller. The Oracle Communications Session Border Controller bases its decision to transcode a call on codec policy configuration and the SDP. Each codec policy specifies a set of rules to be used for determining what codecs are retained, removed, and how they are ordered within SDP.

### Syntax

The following parameters are used to create a codec policy. Their syntax is described inline.

#### **allow-codecs**

**allow-codecs**—The `allow-codecs` parameter configures the codecs that are allowed and/or removed from the SDP. A blank list allows nothing, `*` allows all codecs, **none** removes all codecs, the `:no` designation blocks the specific codec or class of media, and the `:force` designation is used to remove all non-forced codecs.

The **allow-codecs** parameter is configured in the following way:

- `<codec>:no`—blocks the specific codec
- `*`—allow all codecs.
- `<codec>:force`—If any forced codec is present in an SDP offer, all non-forced codecs are stripped from the `m=` line.
- **audio:no**—audio `m=` line is disabled
- **video:no**—video `m=` line is disabled

For example, if you configure PCMU in the `allow-codecs` parameter, the PCMU codec, received in an SDP message is allowed on to the next step of transcoding processing, and all other codecs are removed.

The order of precedence is for removing codecs according to codec policy is:

1. `<codec>:no`—Overrides all other `allow-codecs` parameter actions.

## Transcoding

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2. **audio:no / video:no**. An allow-codecs line like “allow-codecs PCMU audio:no” disables the PCMU m= line because audio:no has a higher precedence than the specific codec.
3. **<codec>:force**
4. **<codec>** Specific codec name and those codecs configured in the add-codecs-on-egress list.
5. **\*** has the lowest precedence of all flags. For example "**allow-codecs \* PCMU:no**" allows all codecs except PCMU.

### order-codecs

**order-codecs**—The order-codecs parameter is used to re-order the codecs in the m= line as the SDP is passed on to the next step. This parameter overwrites the order modified by the **add-codecs-on-egress** command, when relevant. The following is valid syntax for this parameter:

- **<blank>**—Do not re-order codecs
- **\***—You can add a **<codec>** before or after the **\*** which means to place all unnamed codecs before or after (the position of the **\***) the named codec. For example:
- **<codec> \***—Puts the named codec at the front of the codec list.
- **\* <codec>**—Puts the named codec at the end of the codec list.
- **<codec1 > \* <codec2>**—Puts **<codec1>** first, **<codec2>** last, and all other unspecified codecs in between them.
- **<codec>**—When the **\*** is not specified, it is assumed to be at the end.

Any codec name is allowed in the **order-codecs** parameter, even those not defined or not transcodable. An **\*** tells the order-codecs parameter where to place unspecified codecs with respect to the named codecs. Refer to the examples below.

- **<blank>**—do not reorder m= line
- **PCMU \***—Place PCMU codec first, all others follow
- **\* PCMU**—Place PCMU codec last, all others proceed PCMU
- **G729 \* PCMU**—Place G729 codec first, PCMU codec last, all others remain in between
- **PCMU**—If **\*** is not specified, it is assumed to be at the (PCMU **\***).

### Add on Egress

**add-codecs-on-egress**—This parameter adds a codec to the SDP’s m= line only when the codec policy is referenced from an egress realm (except in one 2833 scenario). Codecs entered for this parameter are added to the front of the m= line. Signaling codecs are added to the end of the m= line.

Transcoding can only occur if this parameter is configured. There is a special case for 2833 support where the add-codecs-on-egress parameter is configured for an ingress realm. See [RFC 2833 Scenario 2](#) for details.

### Packetization Time

**packetization-time**—This parameter specifies a media packetization time in ms to use within the realm referencing this codec policy. Packetization time You must also enable the force ptime parameter to enable transrating in conjunction with configuring the packetization time. See the [Transrating](#) section for more information.

## Answer Processing and Examples

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### Unoffered Codec Reordering

According to RFC 3264, the answerer can add codecs that were not offered to the Answer. The answerer may add new codecs as a means of advertising capabilities. RFC 3264 stipulates that these unoffered codecs must not be used. The RFC does not dictate where in the m= line these codecs can appear and it is valid that they may appear as the most preferred codecs.

In order to simplify the answer processing, the Oracle Communications Session Border Controller moves all unoffered codecs in A0 to the back of the SDP answer before any other answer processing is applied.



## Non-transcoded Call

The decision to transcode is based on the top non-signaling codec in A1. If the top A1 codec is present in O1, the call proceeds, non-transcoded. This is the rule for non-signaling codecs (i.e., not RFC 2833 nor FAX).

## Transcoded Call

The following two conditions must then be met to transcode the call's non-signaling media:

- The top A1 codec is not present in the O1 m= line
- The top A1 codec was added by the egress policy

If these rule are met, the Oracle Communications Session Border Controller will transcode between the top A1 codec and the top transcodable, non-signaling O1 codec.


## Voice Transcoding

The following examples use the ingress and egress codec policies listed at the top of each scenario. The examples use changing SDP offers and answers, which contribute to unique results, per example. The effects of the SDP offers and answers are explained in each example.

### Voice Scenario 1

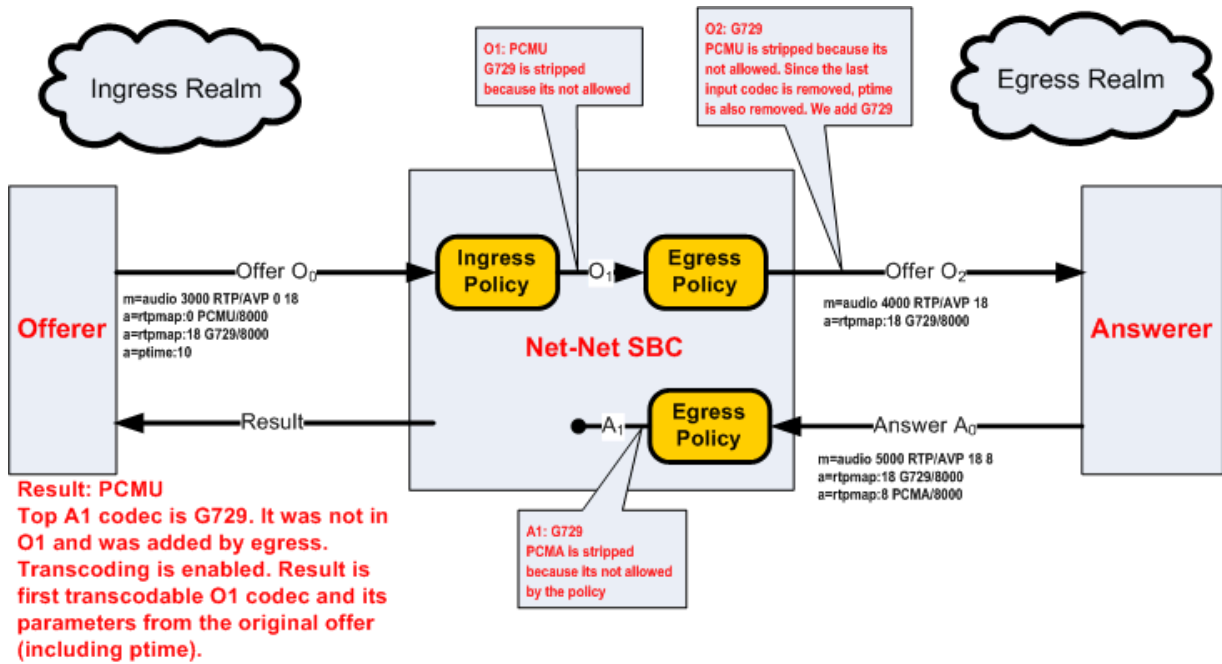
The following ingress and egress policies are used for scenario 1.

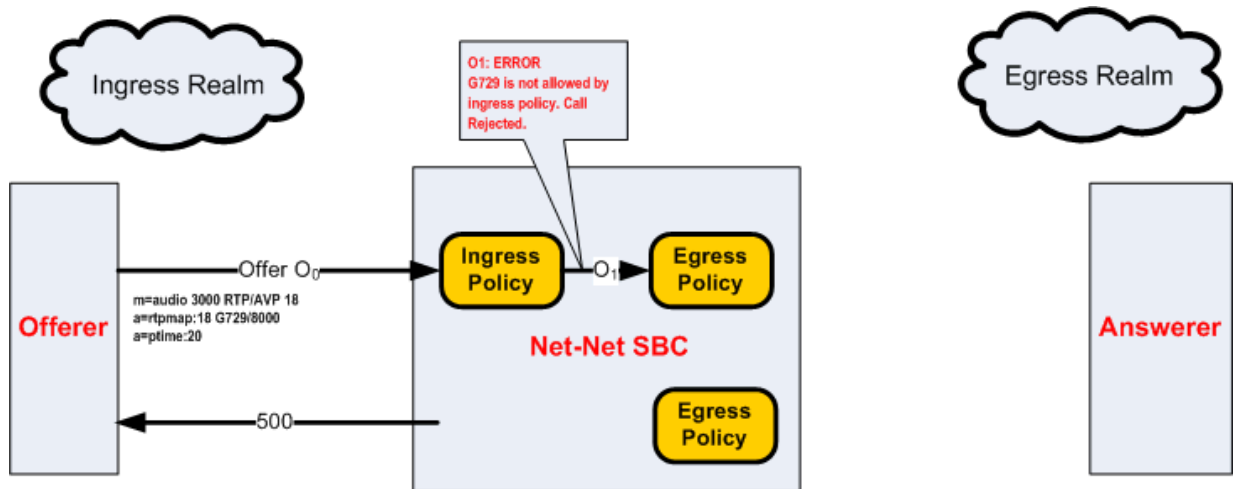
Ingress Policy		Egress Policy	
allow-codecs	PCMU GSM	allow-codecs	G729 GSM G722
add-codecs-on-egress	PCMU	add-codecs-on-egress	G729
order-codecs		order-codecs	
force-ptime	disabled	force-ptime	disabled
packetization-time		packetization-time	

 **Note:** The codec in the ingress policy's add-codecs-on-egress parameter has no effect in the following examples. Its presence would have an effect if a reINVITE was initiated from egress realm, effectively reversing the roles of the codec policies.

1. In the following diagram, PCMU and G729 are offered. Ingress policy removes G729 and allows PCMU. The egress policy adds G729 and strips PCMU from offered SDP and forwards it on to the answerer (ptime is also removed because the last codec was removed).

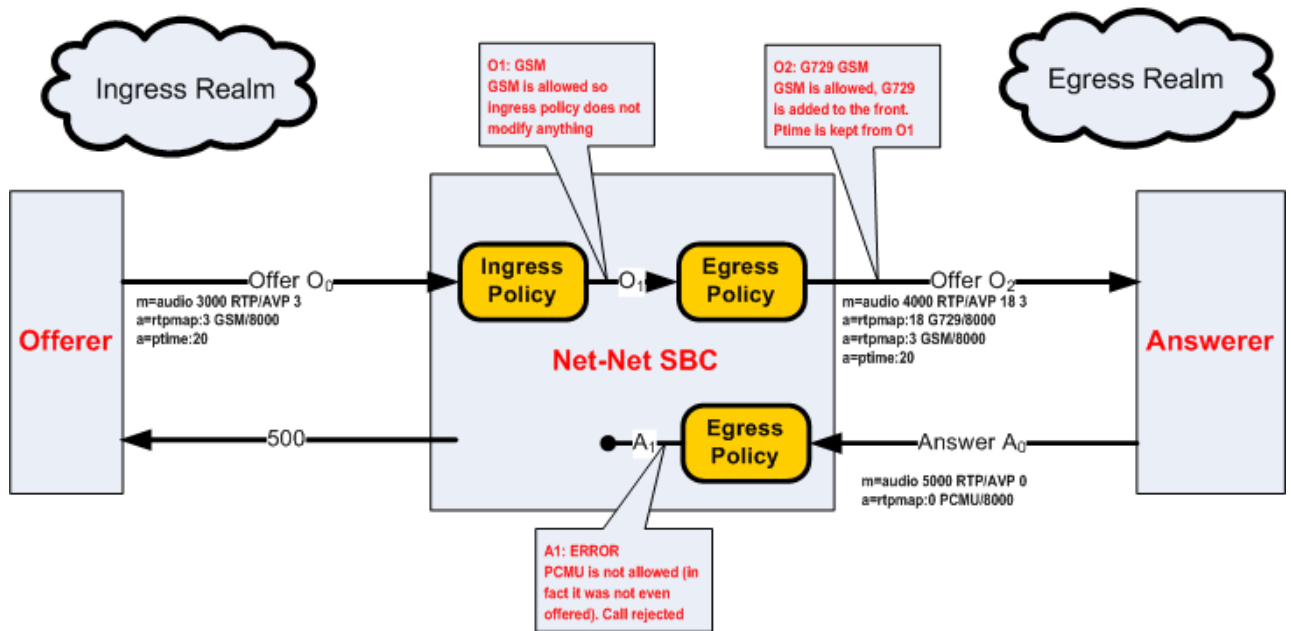
The SDP answer agreed to use G729 and adds PCMA. The egress policy then strips PCMA from the SDP answer. At this point, the top codec in A1, G729 is checked against O1. Since G729 is not present in O1, it is transcoded to PCMU.





- In the following diagram, GSM is in the original SDP offer. It is then passed through to O1. Egress policy adds G729 and retains ptime from O1 and sends this to the answerer as O2.

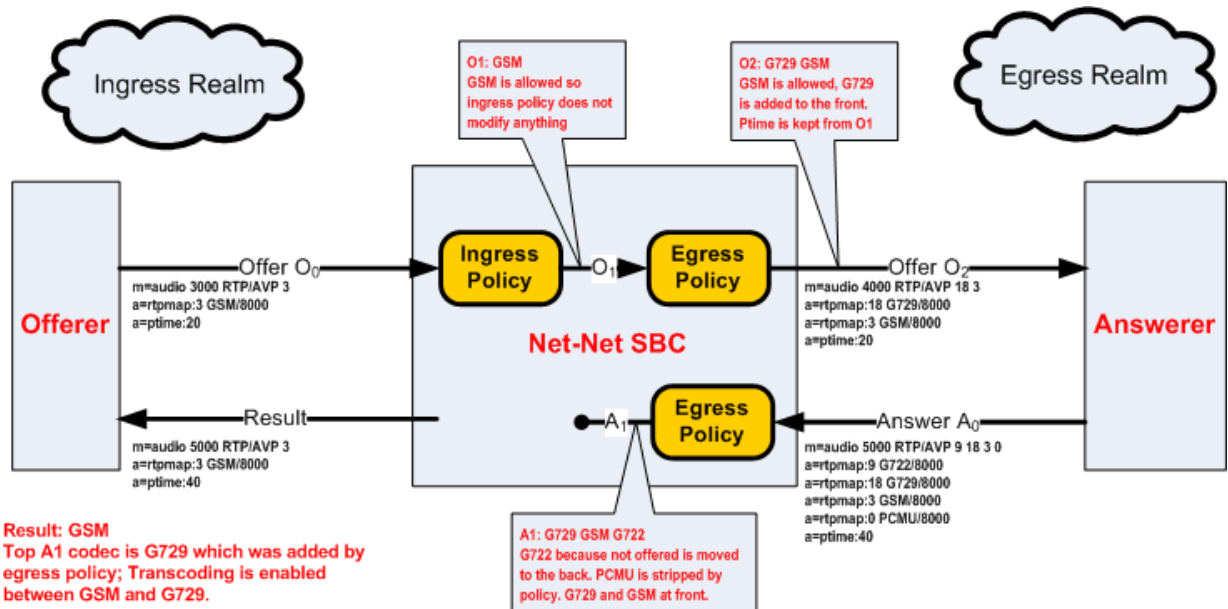
The SDP answer states that the answerer wants to use PCMU. This is a violation of the RFC3264. Therefore the call is rejected.



In this example, when the negotiation fails, the Oracle Communications Session Border Controller sends a 500 message to the offerer and a BYE message to the answerer.

- In the following diagram, GSM is in the original SDP offer. It is then passed through to O1. Egress policy adds G729 and retains ptime from O1 and sends this to the answerer as O2.

The SDP answer replies with G722 G729 GSM and PCMU. PCMU is stripped by policy, G722 is moved to the back of the answer because it was not offered. The top A1 codec was not in O1, and was added by egress policy, therefore the call is transcoded between GSM and G729.

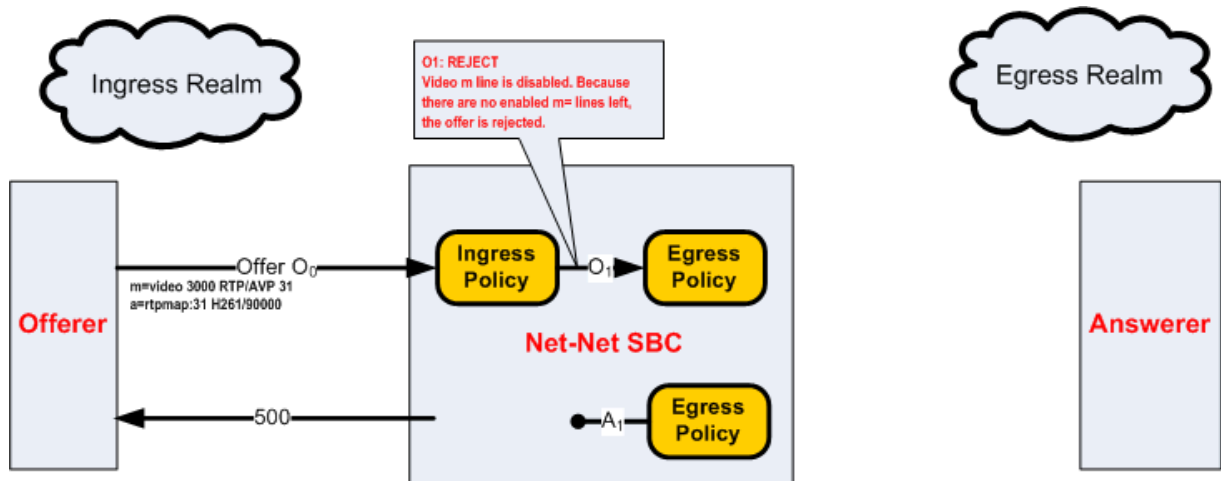


### Voice Scenario 2

The following ingress and egress policies are used for scenario 2.

Ingress Policy		Egress Policy	
allow-codecs	Video:no PCMU:force * PCMA:force	allow-codecs	* PCMA:no
add-codecs-on-egress		add-codecs-on-egress	iLBC G726-16
order-codecs		order-codecs	G726-16 * PCMU
force-ptime	disabled	force-ptime	disabled
packetization-time		packetization-time	

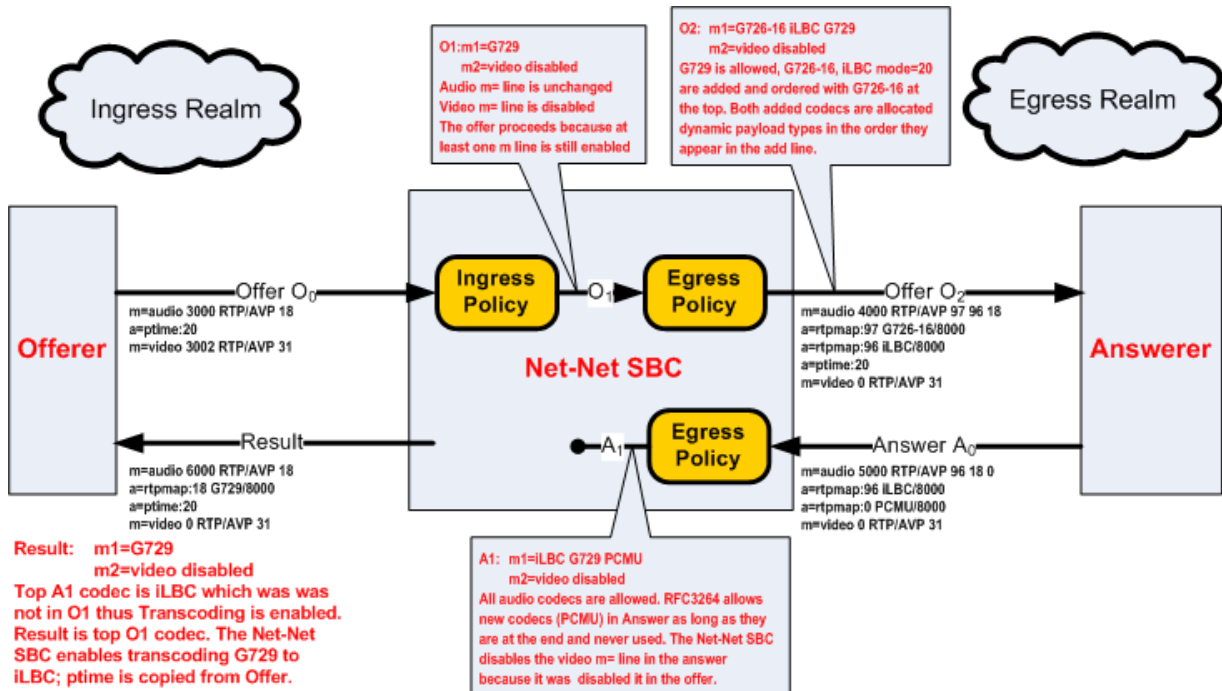
1. In the following diagram, a video m= line is offered. The ingress policy disables the video m= lines. With no enabled m= lines left, the call is rejected.



2. In the following diagram, G729 and video are offered to the Oracle Communications Session Border Controller. Ingress policy allows G729 and disables the video m= line. The egress policy adds iLBC and G726-16, and then orders the codecs according to the order-codecs parameter. The ptime is maintained between  $O_0$  and  $O_2$ . Both

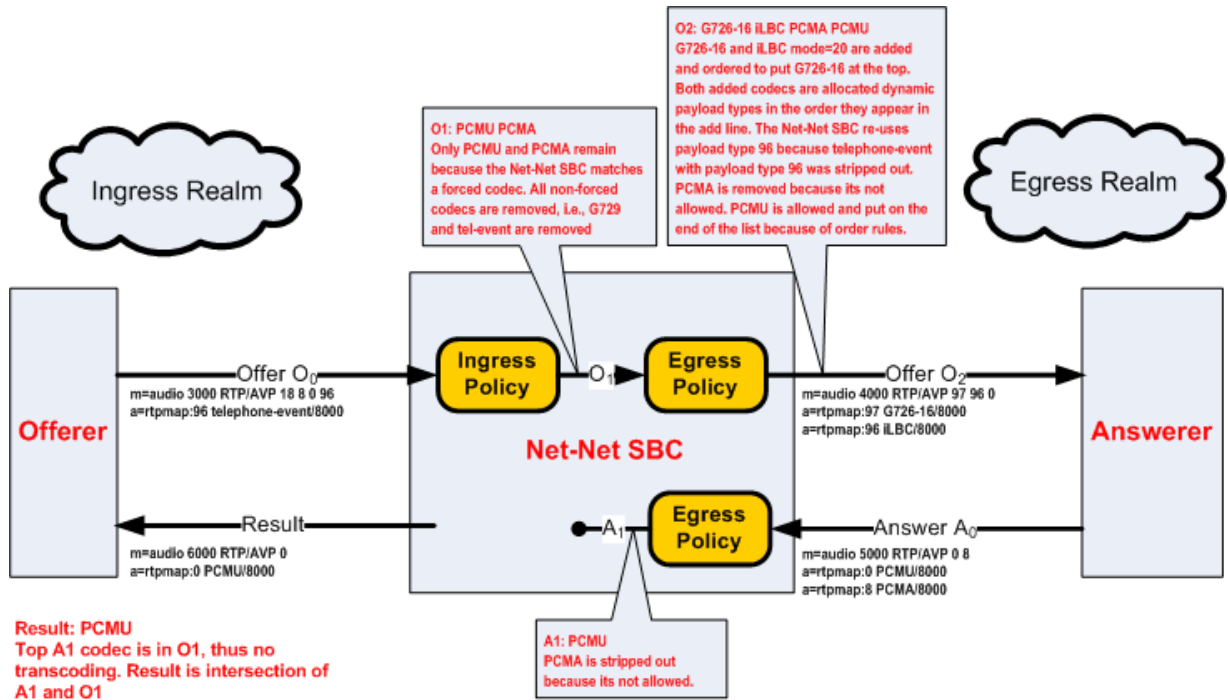
added codecs are allocated dynamic payload types in the order they appear in their m= line. A disabled Video m= line is passed on to the answerer.

The SDP answer agreed to use iLBC, G729, and adds PCMU, and reorders them as stated. The disabled video m= line is maintained. At this point, the top codec in A1, iLBC is used and transcoded with the top codec in O1, G729.



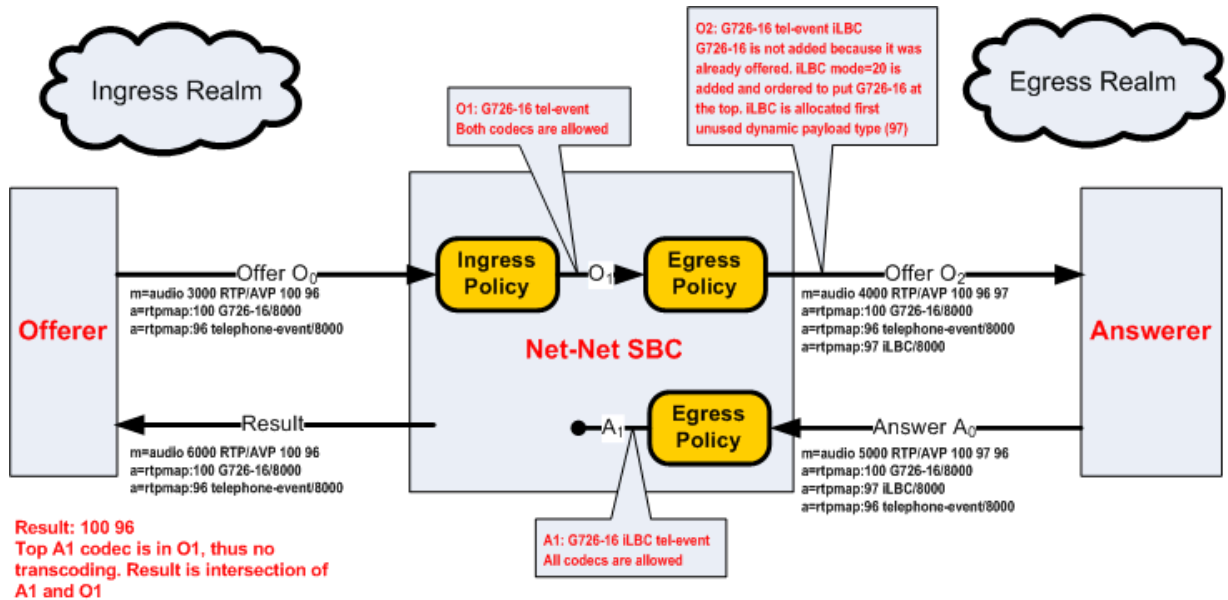
- In the following diagram, G729 and video are offered to the Oracle Communications Session Border Controller. Ingress policy allows G729 and disables the video m= line. The egress policy adds iLBC and G726-16, and then orders the codecs according to the order-codecs parameter. The ptime is maintained between  $O_0$  and  $O_2$ . Both added codecs are allocated dynamic payload types in the order they appear in their m= line.

The SDP answer only wants to use PCMU and PCMA. The egress policy removes PCMA and passes only PCMU to the offerer. Because PCMU was in  $O_1$  and is now the only codec in  $A_1$ , it is used, and no transcoding is used between the endpoints.



- In the following diagram, G726-16 and telephone-event are offered to the Oracle Communications Session Border Controller. Ingress policy allows both codecs. The egress policy adds iLBC, and then orders the codecs according to the order-codecs parameter.

The SDP answer agreed to use all codecs, but reorders them with G726-16 in the top position. Because G726-16 is the top codec in A1, and it is also present in O1, it is used for this call without any transcoding.



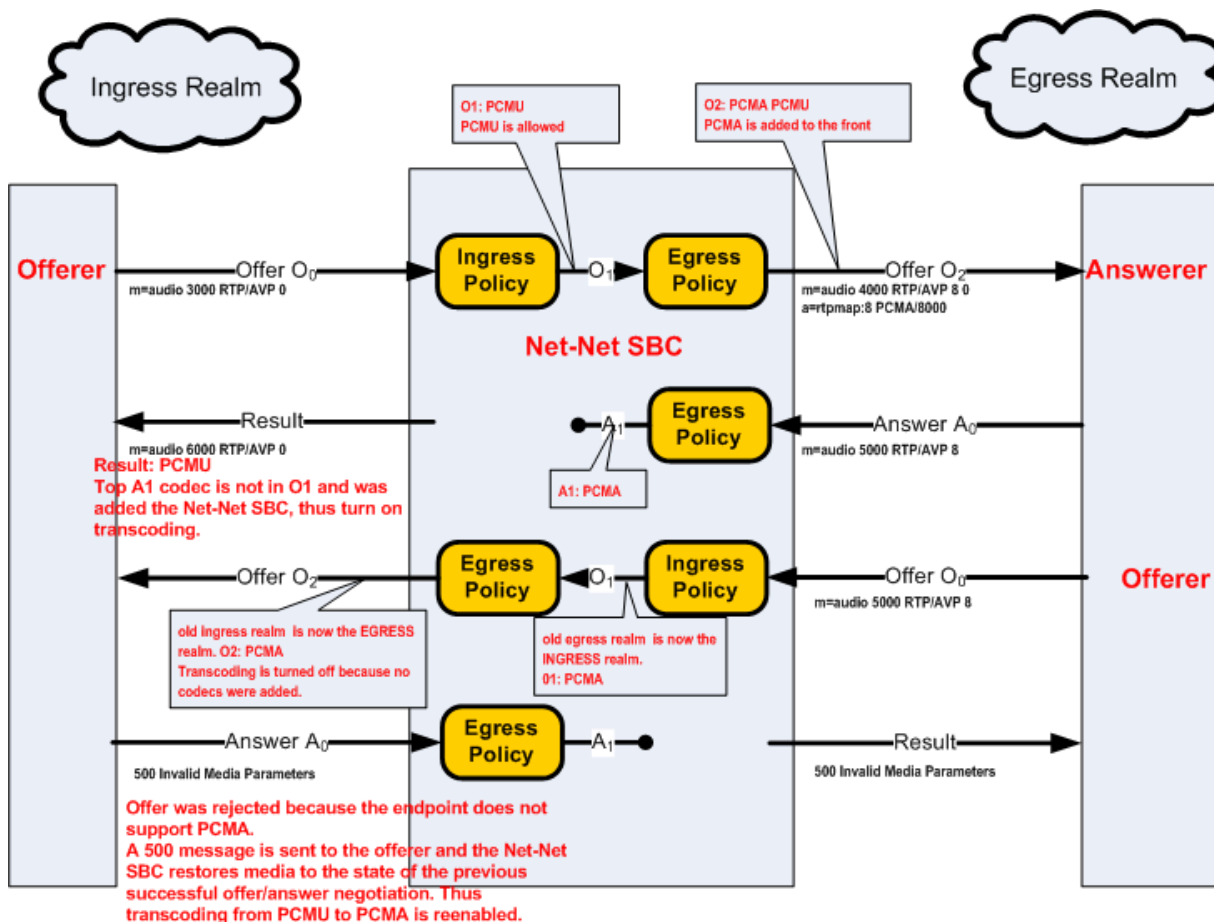
### Voice Scenario 3

Voice scenario 3 involves reINVITES. The following ingress and egress policies are used for scenario 3.

Ingress Policy		Egress Policy	
allow-codecs	PCMU G729	allow-codecs	*
add-codecs-on-egress		add-codecs-on-egress	PCMA

Ingress Policy		Egress Policy	
order-codecs		order-codecs	
force-ptime	disabled	force-ptime	disabled
packetization-time		packetization-time	

In the following diagram, the answerer sends a reINVITE after a previous transcoding session was established. The original offerer and answerer swap roles. The new offerer rejects the SDP offer and the call reverts to the state negotiated in the original SDP negotiation.



## RFC 2833 Transcoding

RFC 2833 defines an RTP payload that functions interchangeably with DTMF Digits, Telephony Tones and Telephony Signals. The Oracle Communications Session Border Controller can monitor audio stream for in-band DTMF tones and then can convert them to data-based telephone-events, as sent in RFC2833 packets. This section explains how the Oracle Communications Session Border Controller transcodes between these RTP-based telephone events and in-band DTMF tones carried by G711. DTMF tones can only be transported in non compressed codecs. The Oracle Communications Session Border Controller supports two DTMFable non-compressed codecs: PCMU (G711μ) and PCMA (G711A).

**Note:** The following line is added to SDP whenever telephone-event is added on egress: a=fmtp:101 0-15

The following two scenarios describe when telephone-event to DTMF transcoding takes place:

### RFC 2833 Scenario 1

The following ingress and egress policies are used for scenario 1.

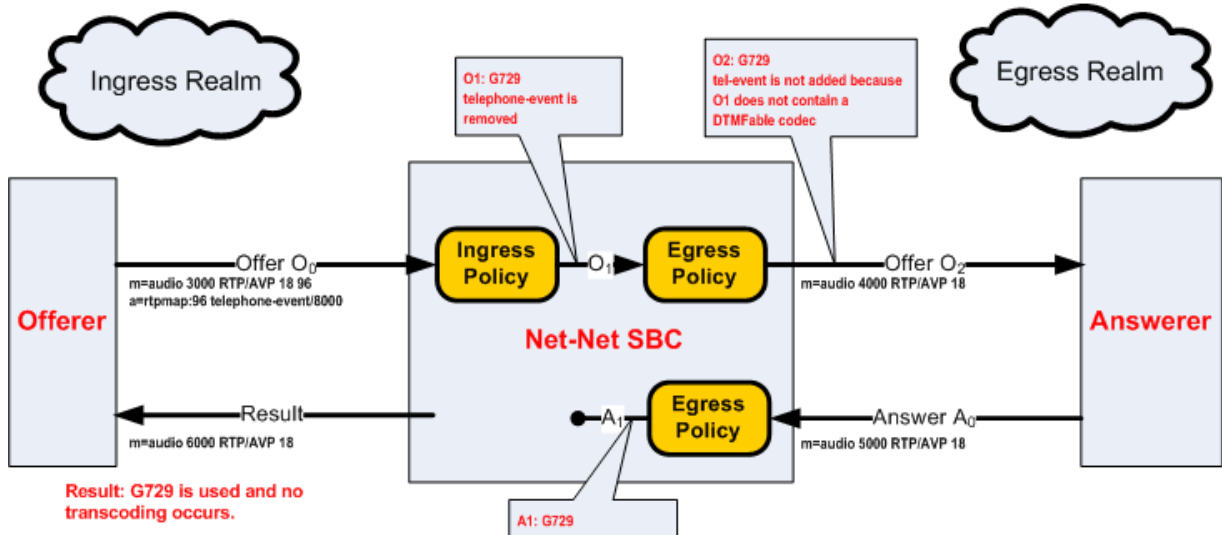


## Transcoding

Ingress Policy		Egress Policy	
allow-codecs	* telephone-event:no	allow-codecs	* PCMA:no
add-codecs-on-egress		add-codecs-on-egress	telephone-event
order-codecs		order-codecs	
force-ptime	disabled	force-ptime	disabled
packetization-time		packetization-time	
dtmf-in-audio	preferred	dtmf-in-audio	preferred

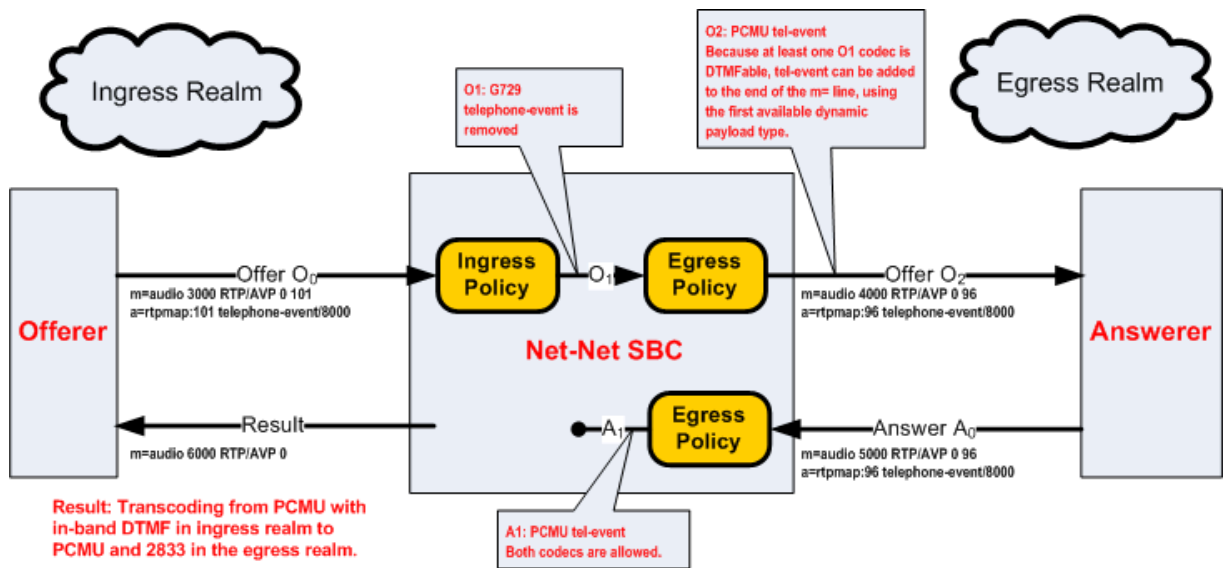
1. In the following diagram, telephone event was offered by the offerer but was stripped by ingress policy. telephone-event was not added by the egress policy because the remaining audio codec in O1 was not DTMFable. G729 was the only codec forwarded on to the answerer.

The SDP answer agreed to use the remaining audio codec, G729. A0 is unaltered by egress policy, and forwarded as the Result to the offerer. Therefore, G729 is used in both the ingress and egress realms, the call does not support RFC 2833, and the call is not transcoded.



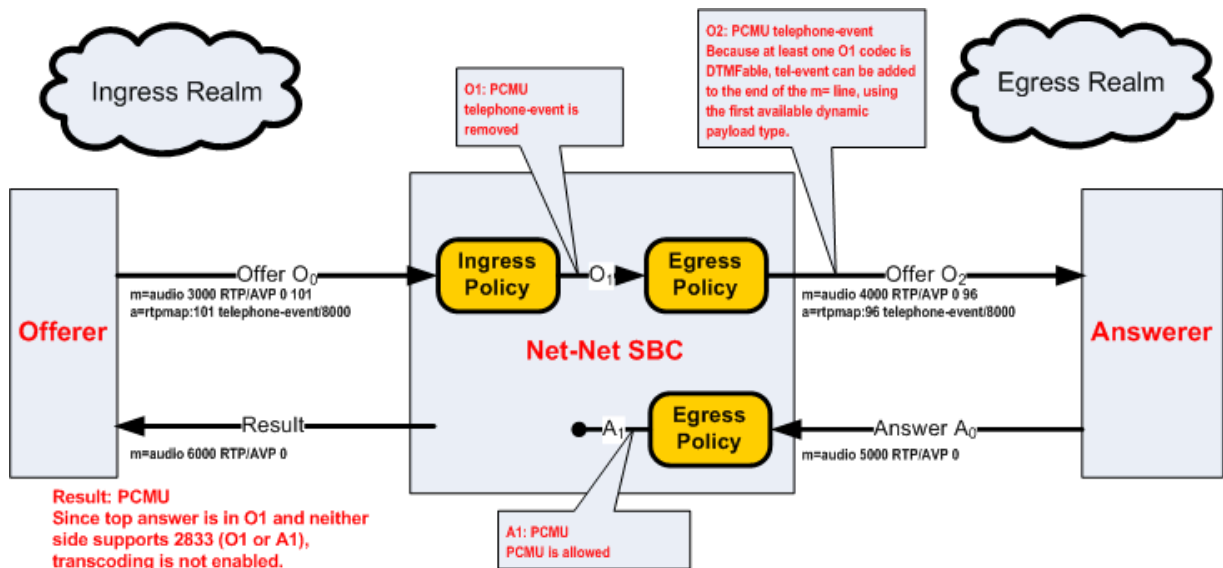
2. In the following diagram, telephone event was offered by the offerer but was stripped by ingress policy. telephone-event was added by the egress policy because the remaining audio codec in O1 was DTMFable. PCMU and telephone-event are then forwarded on to the answerer. Note that the telephone-event payload type is added with the lowest available dynamic type number.





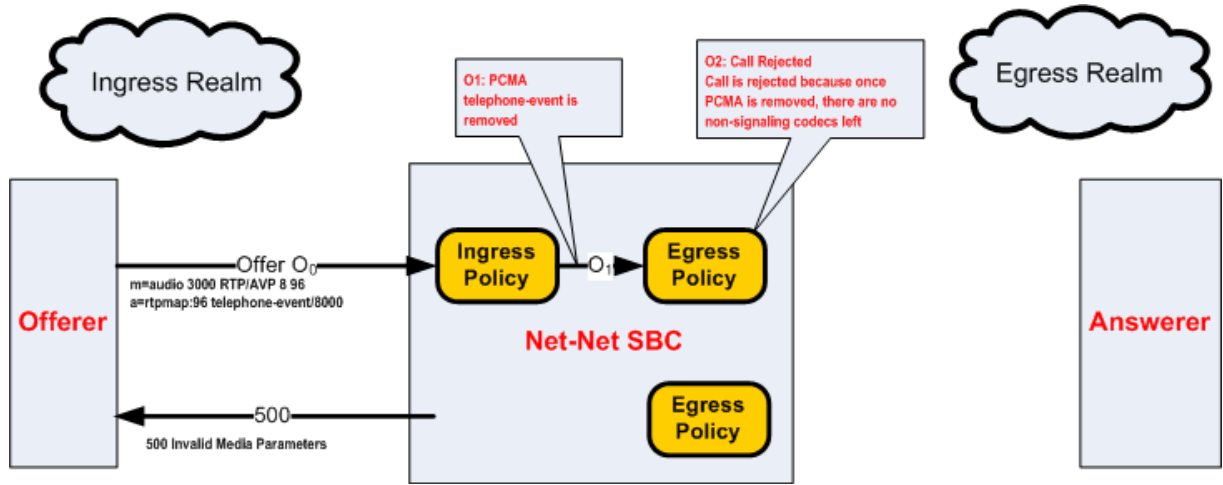
This case illustrates when the answerer supports audio and RFC 2833, but the offerer supports audio with inband DTMF. The Oracle Communications Session Border Controller transcodes between RFC2833 in the egress realm to in-band DTMF on the ingress realm.

3. In the following diagram, telephone event was offered by the offerer but was stripped by ingress policy. telephone-event is added by the egress policy because the remaining audio codec in O1 was DTMFable. PCMU and telephone-event are then forwarded on to the answerer. Note that the telephone-event payload type is added with the lowest available dynamic type number.



The SDP answer only agreed to use PCMU. When A0 reaches the egress policy, it is passed along through the Oracle Communications Session Border Controller to the offerer. Because telephone-event was not answered by the answerer and not supported in O1, it can't be used. Transcoding is therefore not used for this call.

4. In the following diagram, telephone event was offered by the offerer and was stripped by ingress policy. Since PCMA was also stripped by the egress policy, leaving no non-signaling codecs, the call is rejected. A 500 message is sent back to the offerer.

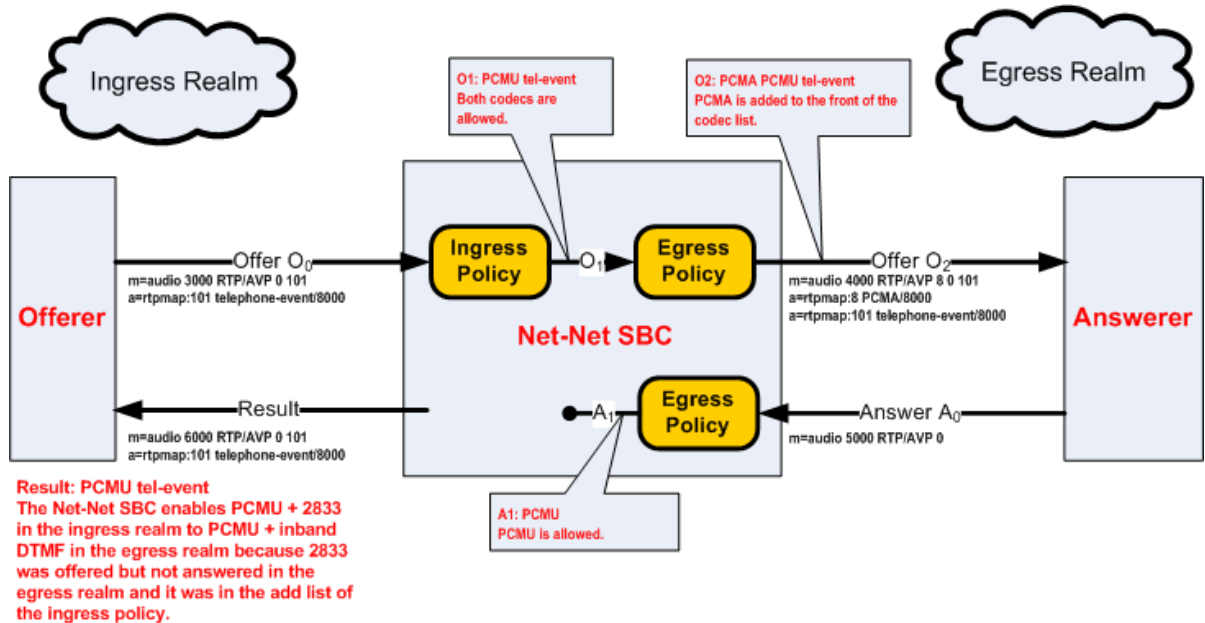


**RFC 2833 Scenario 2**


The following ingress and egress policies are used for RFC2833 scenario 2.

Ingress Policy		Egress Policy	
allow-codecs	*	allow-codecs	*
add-codecs-on-egress	telephone-event	add-codecs-on-egress	PCMU
order-codecs		order-codecs	
force-ptime	disabled	force-ptime	disabled
packetization-time		packetization-time	
dtmf-in-audio	preferred	dtmf-in-audio	preferred

- In the following diagram, telephone event and PCMU are offered by the offerer. They are both passed to O1, and PCMA is added as it is sent to the answerer. The SDP answer, A0 disables all codecs but PCMU.

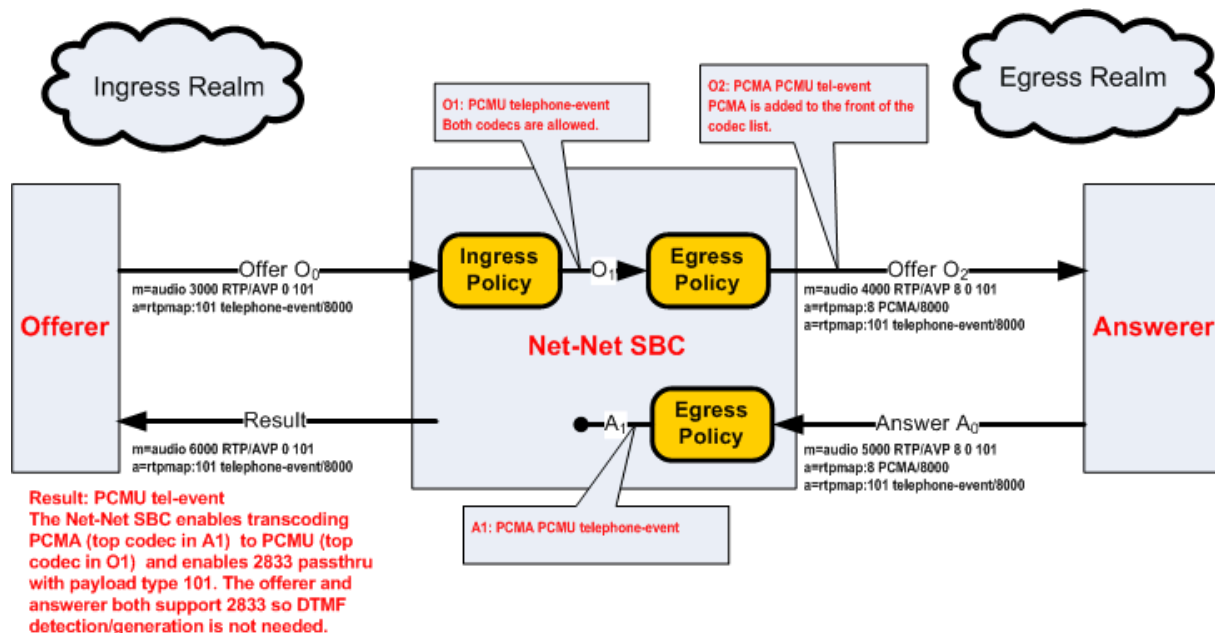


The Oracle Communications Session Border Controller adds telephone-event to the result because it is listed on the ingress policy's add-codecs-on-egress parameter and present in the offerer's SDP.

 **Note:** This is the only time the add list of an ingress policy is utilized as a check.

The result SDP includes PCMU and telephone-event in the ingress realm, which is transcoded to PCMU with in-band DTMF in the egress realm.

- In the following diagram, telephone event and PCMU are offered by the offerer. They are both passed to O1, and PCMA is added as it is sent to the answerer. The SDP answer supports all three codecs offered, with PCMA added on top.



The answerer responds with PCMA as the preferred codec in A0. The Oracle Communications Session Border Controller compares A1 to O1 to make the transcoding decision. PCMA is the top codec in A1 and is transcoded to PCMU, the top codec in O1. Also, because telephone-event is supported by both sides of the call, it is passed through without any transcoding necessary.

This case illustrates when both endpoints are capable of sending and receiving telephone-event. Regardless of whether the audio portion of the call is transcoded, the telephone-event messages are passed through the system untouched, thus not requiring transcoding resources. This is known as telephone-event pass-through.

## FAX Transcoding

FAXes are transmitted in a call as either T.30 and T.38 media. T.30 FAX is binary in-band media carried over G.711. The Oracle Communications Session Border Controller can transcode between T.38 and a faxable codec. The supported faxable codecs are PCMU and PCMA.

T.30 can only be transported in non-compressed codecs. The two non-compressed codecs supported by the Oracle Communications Session Border Controller are PCMU (G711 $\mu$ ) and PCMA (G711A). If a transcoding realm does not support an uncompressed codec, T.30 can not be supported in that realm. Alternatively, G711FB may be allowed specifically for FAX only.

The Oracle Communications Session Border Controller uses an internal codec called G711FB (G711 - Fall Back) that is an umbrella codec of all FAXable codecs. G711FB will default to PCMU for the purpose of offering a faxable codec. You can remap G711FB to PCMA by configuring the media-profile for it appropriately. G711FB's only use is for FAX transcoding.

FAX transcoding is triggered when you configure the add on egress parameter with either T.38 or G711FB. In a FAX scenario, when the codec policy adds either T.38 or G711FB, a new m= line is added to the SDP. When adding T.38,

## Transcoding

the new m= line specifies the T.38 codec. When adding G711FB, the new m= line specifies PCMU (or alternatively PCMA).

Once added, m= lines can not be deleted in the context of a call. The Oracle Communications Session Border Controller maintain all m= lines between itself and an endpoint throughout the course of call. All m= lines not in use can be disabled by setting their receive port to 0, but they can not be removed from the SDP.

### Defining G711FB

G711 Fall Back (G711FB) is an internal codec that encompasses PCMU and PCMA for carrying fax information FAXable codecs. The G711FB codec must be configured either way for when the Oracle Communications Session Border Controller inserts a FAXable codec in SDP. G711FB is only used for FAX transcoding scenarios.

To define G711 FB, create a media profile configuration element named g711fb and set the payload-type to 0 or 8.

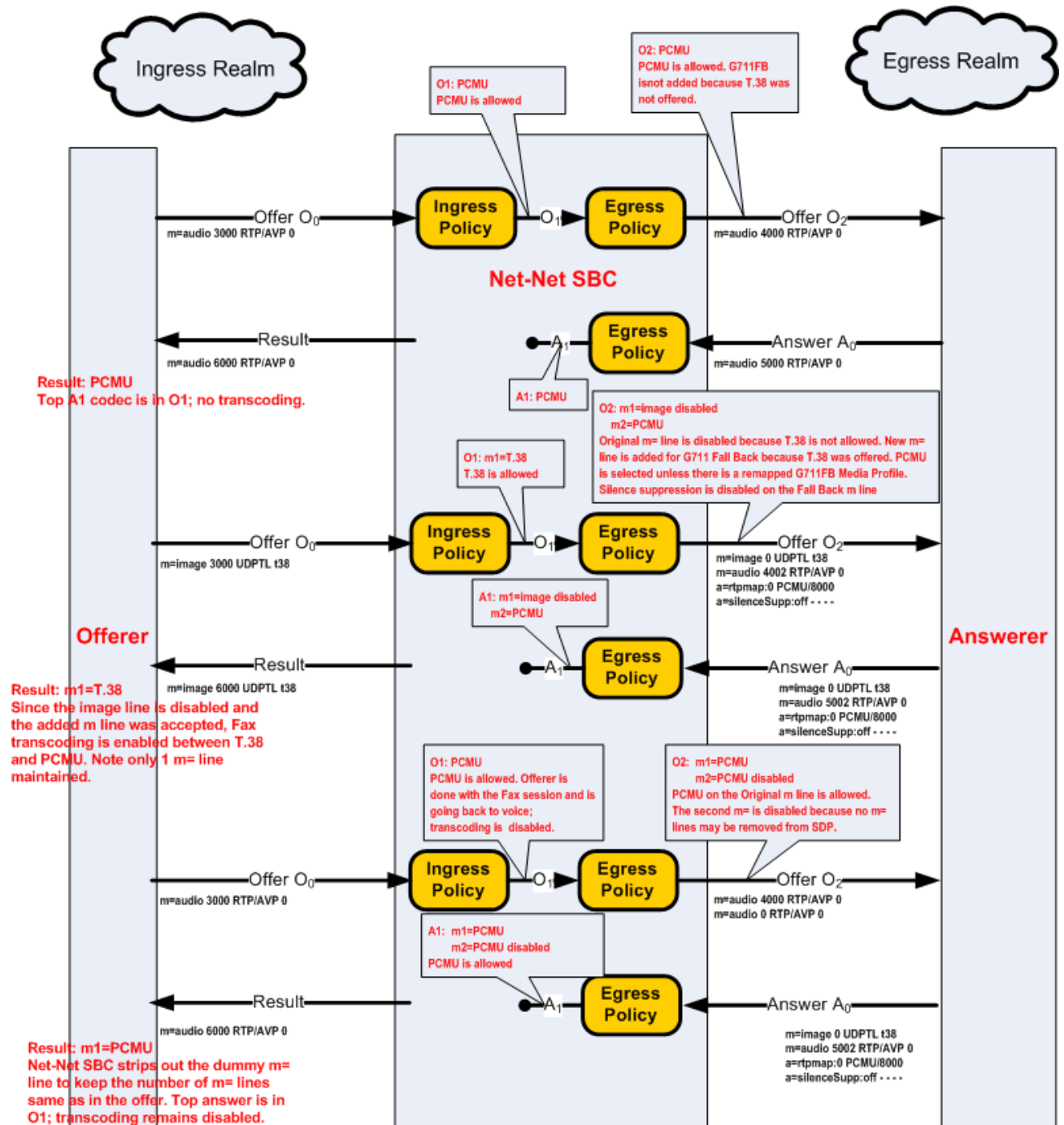
Codec (supported bit rates)	RTP Payload Type	Default Ptime (ms)	Supported Ptime (ms)
T.38	N/A	30	10, 20, 30
G711FB (64 kbps)	0, 8	30	10, 20, 30

### FAX Scenario 1

The following ingress and egress policies are used for this FAX scenario.

Ingress Policy		Egress Policy	
allow-codecs	*	allow-codecs	T.38:no
add-codecs-on-egress		add-codecs-on-egress	G711FB
order-codecs		order-codecs	
force-ptime	disabled	force-ptime	disabled
packetization-time		packetization-time	

In the following diagram, there are three offer-answer exchanges. Initially a PCMU-to-PCMU session is negotiated. Next, a T.38 to PCMU session is negotiated. Finally, the session reverts to non-transcoded PCMU to PCMU state.



### FAX Scenario 2

The following ingress and egress policies are used for this FAX scenario.

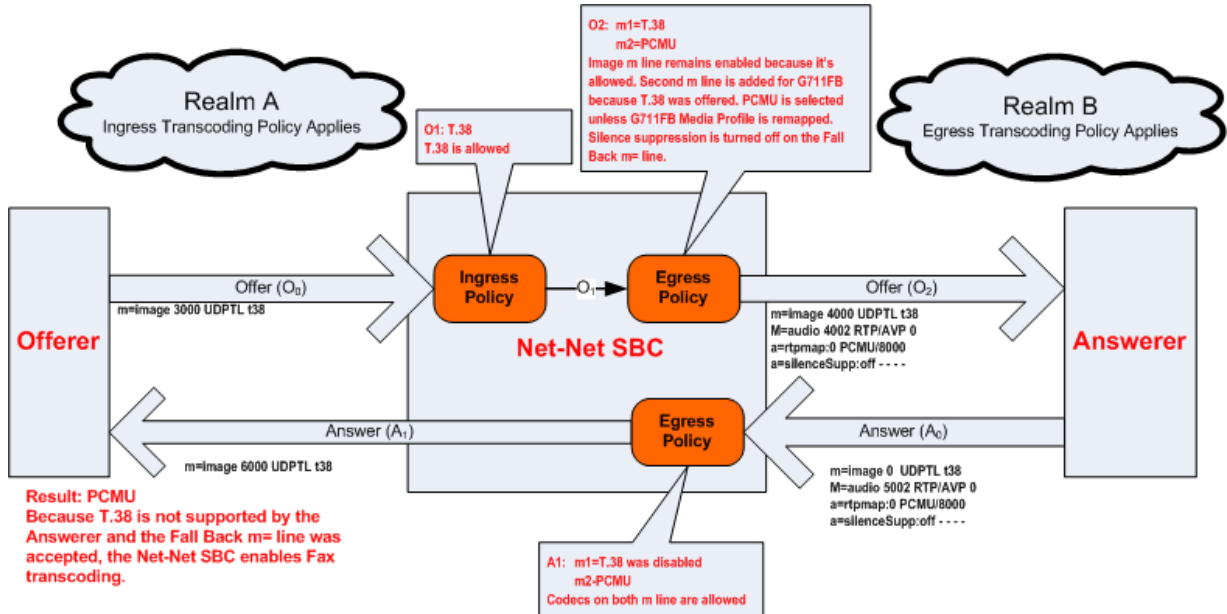
Ingress Policy		Egress Policy	
allow-codecs	*	allow-codecs	*
add-codecs-on-egress		add-codecs-on-egress	G711FB
order-codecs		order-codecs	
force-ptime	disabled	force-ptime	disabled

## Transcoding

Ingress Policy		Egress Policy	
packetization-time		packetization-time	

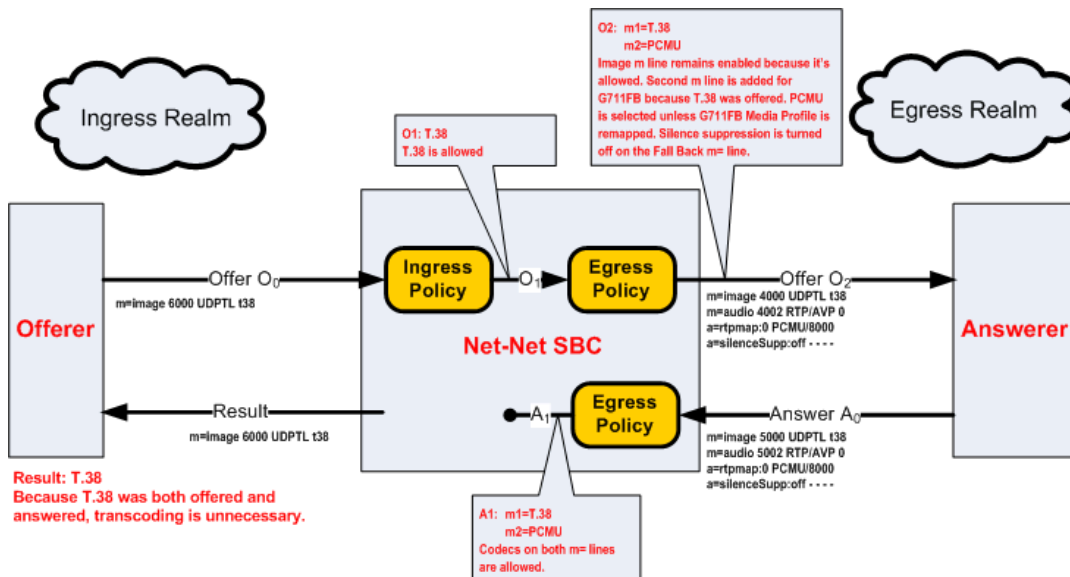
- In the following diagram, T.38 is offered to the Oracle Communications Session Border Controller. A second m= line was added to O1 that included a G711FB codec (PCMU).

The SDP answer agreed to PCMU, but disabled T.38. When the Oracle Communications Session Border Controller forwarded the SDP in A1 to the answerer, it stripped the second m= line. Because A1 rejects T.38 m= line, but accepts the PCMU m= line, FAX transcoding is enabled.



- In the following diagram, T.38 is offered to the Oracle Communications Session Border Controller. A second m= line was added to O1 that included a G711FB codec (PCMU).

The SDP answer agreed to PCMU and T.38. Because both O1 and A1 support T.38, the call proceeds without transcoding.



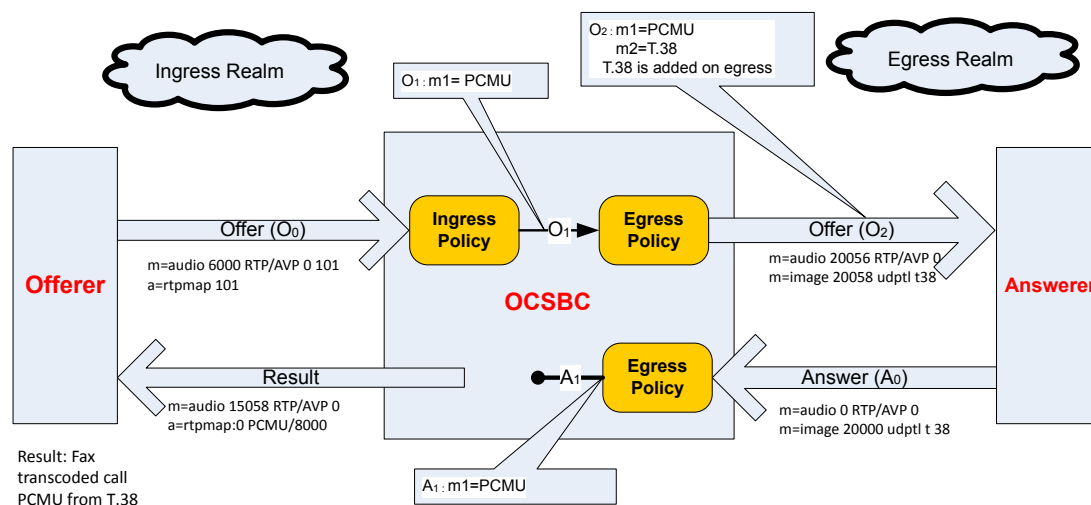
### FAX Scenario 3

The following ingress and egress policies are used for this FAX scenario.

Ingress Policy		Egress Policy	
allow-codecs	*	allow-codecs	*
add-codecs-on-egress		add-codecs-on-egress	PCMU, G729 ,T.38
order-codecs		order-codecs	
force-ptime	disabled	force-ptime	disabled
packetization-time		packetization-time	

- In the following diagram, PCMU and telephone-event codecs are received by Oracle Communications Session Border Controller .The egress codec policy has PCMU, G729 and T.38 **add-codecs-on-egress**. Since there is a faxable codec in the SDP offer and T.38 in **add-on-egress**, the non-faxable codec G729 is stripped and T.38 is added to egress offer.

In the A0 answer the audio m-line is zero indicating disabled and the port for the image m-line is non-zero (20000) so the called party has selected T.38. Hence PCMU in realm A is transcoded to T.38 in realm B.

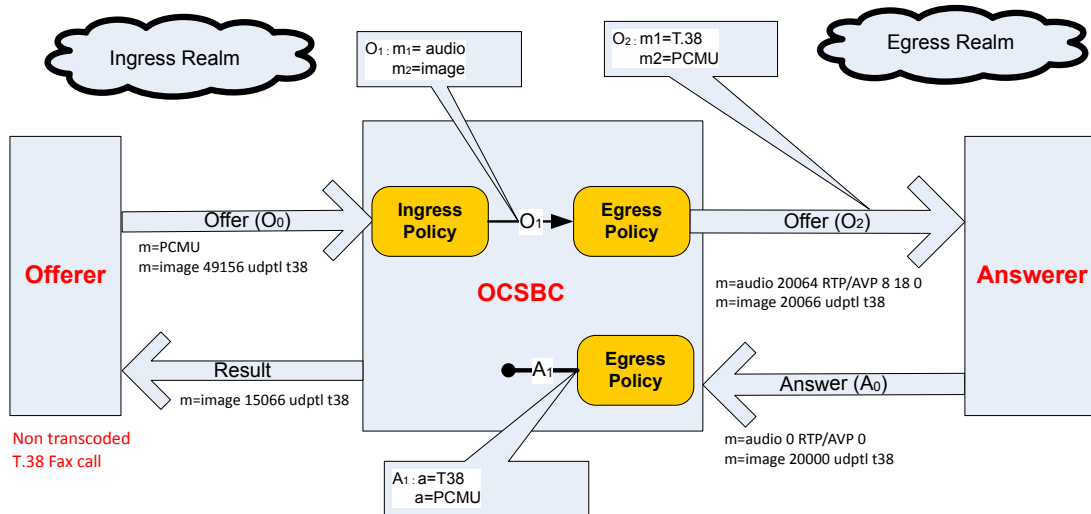


- In the following diagram, PCMU and T38 are received by the Oracle Communications Session Border Controller. The egress codec policy has PCMU, G729, and T.38 **add-codecs-on-egress**.

T.38 is present in the O<sub>0</sub> offer, and therefore will not be explicitly added by the egress codec policy to the O<sub>2</sub> Offer. The logic to remove non-faxable codecs is only invoked when T.38 is added by the **add-codecs-on-egress** parameter. In this example, T.38 is not added since is already present in SDP. With PCMU and T.38 received, G729 as a non-faxable codec is not removed. Therefore, PCMU, T.38, and G729 are all present in the O<sub>2</sub> SDP offer sent to the answerer.

In the A<sub>0</sub> answer, the audio m-line port is 0 indicating that audio is disabled, and the port for the image m-line is nonzero (20000), thus the called party has selected T.38. In the A<sub>1</sub> answer the OCSBC send the audio m-line port as 0 indicating that audio is disabled, and the port for the image m-line is nonzero (20000). This set-up results in a non-transcoded T.38 -OCSBC -T.38 fax call.





## Transrating

The Oracle Communications Session Border Controller can transrate media as it exits the Oracle Communications Session Border Controller into the network. Transrating is also known as forced packetization time (ptime), and is used to enforce a configured ptime within a realm. Transrating is often desirable when devices in a realm can only accept media with a specific ptime, or to optimize bandwidth.

If this feature is configured, the media portion of a call is transrated regardless of which codecs are ultimately chosen for each realm as long as they are transcodable. This allows realms that have devices that can only use a single packetization interval to interwork with devices that may or may not have the same packetization capabilities.

You must enable force-ptime in the egress codec policy and then specify the packetization time to force. When force ptime is enabled, it implicitly masks all codecs not of the specified packetization time that are listed in that codec policy's allow codecs and add codecs on egress parameters. For example, if force ptime is enabled with a packetization time of 20 ms, then no G723 codecs (which are only available at 30, 60, and 90 ms) may be active via codec policy in that realm.

Transrating occurs when forced-ptime is enabled and the offered and answered ptimes do not match and the top non-Signaling codec of A1 and top non Signaling codec of O1 are Transcodable.

**Note:** Answered ptime A1 does not have to be equal to the ptime inserted into the outgoing offer O2, it just has to be different than the offer the Oracle Communications Session Border Controller received (O1).

Please refer to [Transcodable Codecs](#) for current list of ptimes per codec supported by the DSPs.

### Transrating Scenario 1

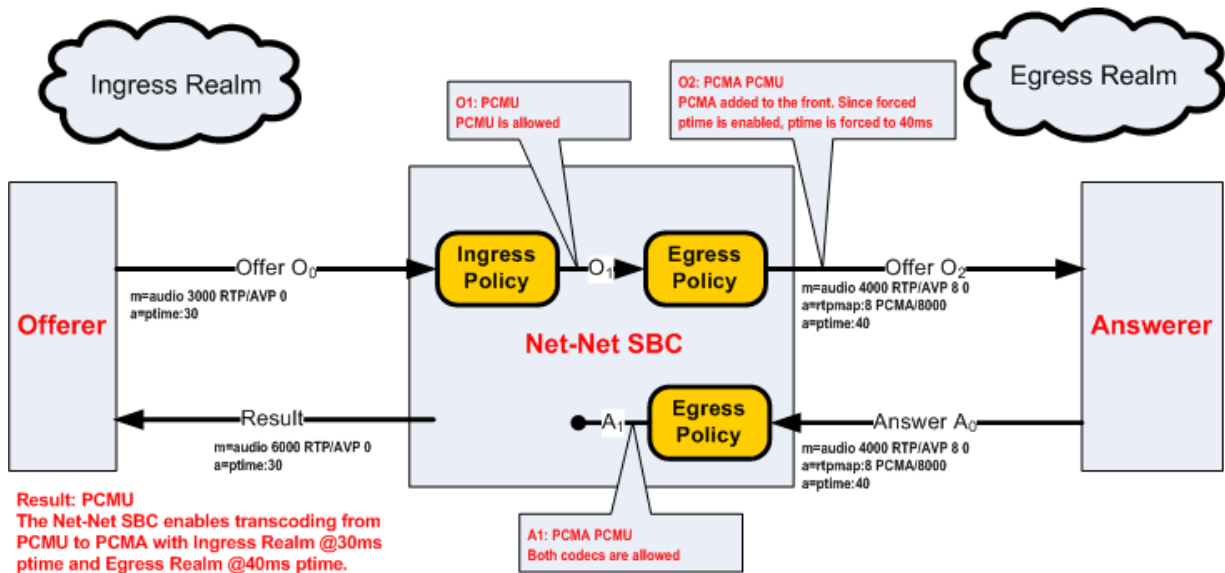
The following ingress and egress policies are used for this FAX scenario.

Ingress Policy		Egress Policy	
allow-codecs	*	allow-codecs	*
add-codecs-on-egress		add-codecs-on-egress	PCMA
order-codecs	G723 *	order-codecs	
force-ptime	disabled	force-ptime	enabled
packetization-time		packetization-time	40

1. In the following diagram, PCMU is offered in the ingress realm with 30ms ptime, and the egress realm is forced to use 40ms ptime. PCMA is added as the top codec for the egress realm.

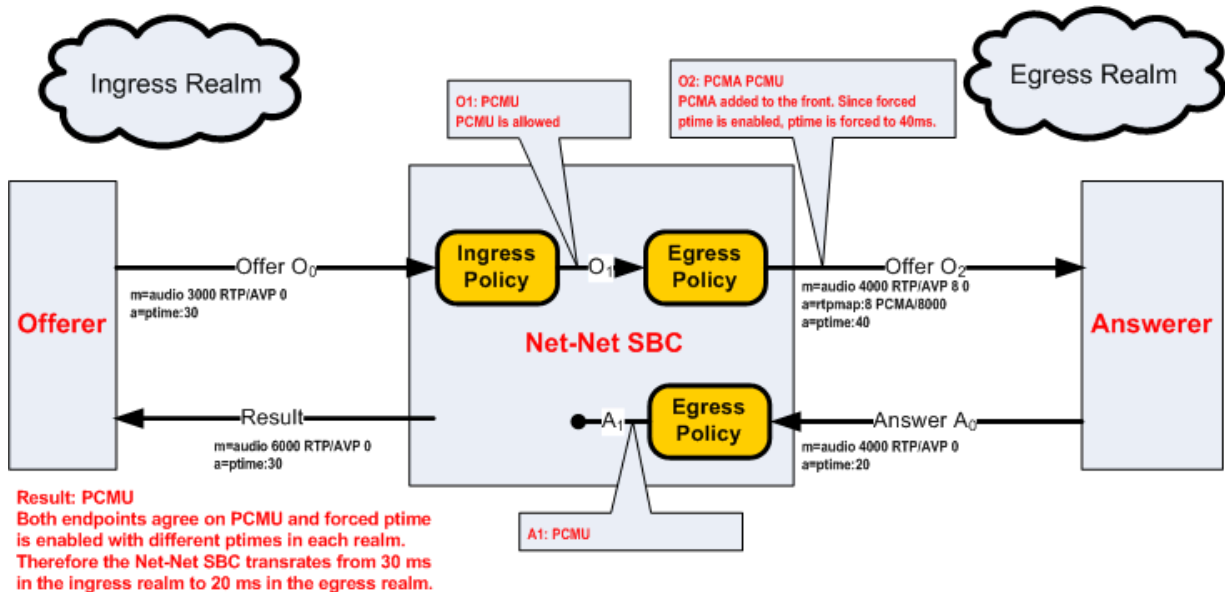


The Oracle Communications Session Border Controller enables transcoding between the ingress realm (PCMU) and the egress realm (PCMA) and the ptimes as negotiated are also maintained.



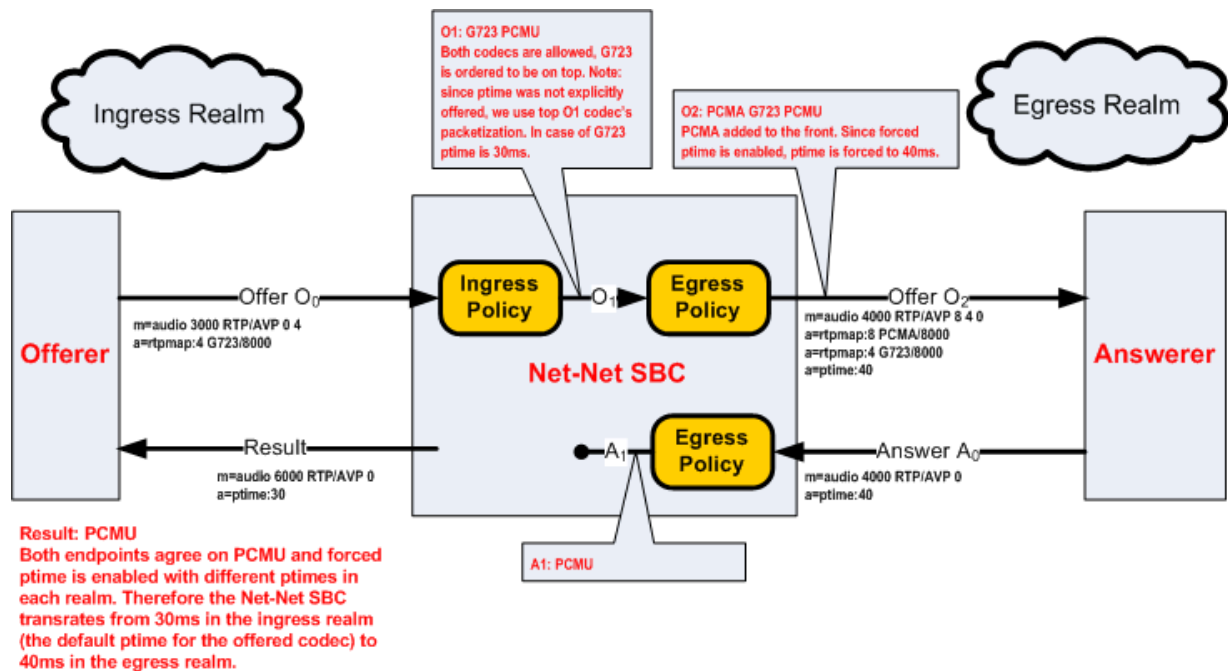
- In the following diagram, PCMU is offered in the ingress realm with a ptime of 30ms, and forced to 40 ms in the egress realm by policy.

The answerer chooses to use PCMU with a 20 ms ptime. Thus the call is not transcoded, but it is transrated from 30ms in the ingress realm to 20ms in the egress realm.



- In the following diagram, PCMU and G723 are offered in Realm A. The top codec's ptime (30ms) is implied as the one for the ingress realm. The Oracle Communications Session Border Controller adds PCMA to the SDP offer with a 40ms ptime.

The answerer chooses to use PCMU with a 40 ms ptime. Thus the call is transrated from 30ms in the ingress realm to 40ms in the egress realm.



## Default Media Profiles

The Oracle Communications Session Border Controller contains a set of default media profiles that define characteristics of well-known IANA codecs. You can not view the default media profiles' configurations, but you can override them by configuring identically-named media profile configuration elements.

Transcodable codecs are a subset of the default media profiles which the Oracle Communications Session Border Controller can transcode between.

## Transcodable Codecs

The following list shows the transcodable codecs which the Oracle Communications Session Border Controller can add to SDP. These codecs all reflect default media profiles for their given names. Enter codecs in the configuration exactly as below.

- PCMU
- PCMA
- G729
- G729A
- iLBC
- telephone-event
- T.38
- G711FB
- G726
- G726-16
- G726-24
- G726-32
- G726-40
- G722
- G723
- GSM
- AMR

- AMR-WB
- EVRC0
- EVRC
- EVRC1
- EVRCB0
- EVRCB
- EVRCB1

When creating an override media profile from the previously listed codec, case is ignored. Also, GSM is GSM-FR.

## Preferred Default Payload Type

When the Oracle Communications Session Border Controller adds a codec with a dynamic payload type to SDP, it uses the lowest unused payload number. You can configure a preferred payload type for a dynamic codec by creating an override media profile. This makes the Oracle Communications Session Border Controller use your preferred payload type for insertion into SDP. If you configure a dynamic codec to use a preferred payload type, and that payload type is already in use, the codec will still be inserted into SDP, but with the first available dynamic payload type.

For example, you create a media profile for telephone-event with a payload type of 101. If telephone-event is added to SDP, and payload type 101 is already in use in the SDP, the Oracle Communications Session Border Controller will use the first available payload type in the 96-127 range when adding telephone-event.

## Redefining Codec Packetization Time

You can configure a media profile with a packetization time (ptime) that overrides the codec's default ptime. Transcoding functions look up and use default ptimes when not specified in offered or answered SDP. Default ptime for most audio codecs is 20ms; some however are 30ms. See the [Transcodable Codecs](#) list for default values.

To change the default ptime for a codec, you must create a media profile that overwrites the default ptime parameter with your new packetization time. When SDP is received with no 'a=ptime' attribute or when adding the codec to egress SDP, the newly configured ptime is used.

New default ptime for a media profile is entered by typing "ptime=<x>" in the parameters parameter, where <x> is the new default packetization time.

## mptime Support for Packet Cable

The SDP specification lacks the ability to specify unique packetization times per codec when more than one codec is listed in an m= line. The ptime attribute is not related to a specific codec but to the entire m= line. When multiple codecs appear on a single m= line, the PacketCable mptime attribute can specify different packetization times for each codec.

The Oracle Communications Session Border Controller adheres to PKT-SP-NCS1.5-I01-050128 and PKT-SP-EC-MGCP-I06-021127 for processing and generating mptime. The mptime line uses an integer to indicate the packetization time for each corresponding codec in the m= line. The dash character, "-", on an mptime line is used for non-packetized codecs, such as CN or telephone-event.

If the Oracle Communications Session Border Controller receives an invalid mptime, it is ignored and removed. If a valid mptime is received in the incoming SDP, its values will be used for packetization times of each corresponding codec and a valid mptime line will be sent in the outgoing SDP.

Valid:

```
m=audio 10000 RTP/AVP 0 96 8
a=mptime:20 - 30
a=rtptime:96 telephone-event/8000
```

Valid: 'ptime' attribute is ignored

## Transcoding

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```
m=audio 10000 RTP/AVP 0 8
a=mptime:20 30
a=ptime:30
```

Invalid: dash cannot be first mptime value

```
m=audio 10000 RTP/AVP 96 0
a=mptime: - 20
```

When Oracle Communications Session Border Controller includes an mptime in an outgoing SDP, it will also always add a ptime attribute with the value of the most preferred codec. This is done to increase the interoperability with devices that do not support mptime.

### AMR-NB and AMR-WB Specifications

The Oracle Communications Session Border Controller supports Adaptive Multi-Rate Narrow Band & Wide Band codecs. All configurations of this codec, as indicated by SDP, are transcodable except when the following SDP parameters are enabled:

- robust-sorting
- interleaving

When AMR is configured in a codec policy's add-codecs-on-egress parameter, it is forwarded from the Oracle Communications Session Border Controller with the following default settings:

- 12.2 kbps (AMR-NB)
- 23.85 kbps (AMR-WB)
- RTP/IF1 format
- No redundant packets
- bandwidth efficient default payload
- No CRC frame
- 20ms default ptime



**Note:** AMR and AMR-WB each require a separate license.

## EVRC Family of Codecs

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The Acme Packet 6300 supports 2 codecs in the EVRC family. Each codec has has three variants.

- EVRC-A is also commonly referred to as EVRC. The following EVRC-A packet formats are supported:

Header-free packet format = EVRC0

Bundled/interleaved packet format = EVRC

Compact Bundled packet format = EVRC1

- The following EVRC-B packet formats are supported:

Header-free packet format = EVRCB0

Bundled/interleaved packet format = EVRCB

Compact Bundled packet format = EVRCB1

### EVRC-A Codec for Transcoding

The Acme Packet 6300 supports the Enhanced Variable Rate codec (EVRC) for transcoding. Three subtypes of the EVRC codec are supported as media-profiles:

- EVRC0—header-free EVRC format. This codec is transcodable.
- EVRC—non-header-free EVRC format. This codec is transcodable.
- EVRC1—fixed rate EVRC format. This codec is transcodable.

To transcode to and from EVRC, the EVRC license must be installed.

 **Note:** The Acme Packet 6300 handles optional parameters according to RFC 4788 unless otherwise specified.

### **EVRC0 Supported Options**

Required Parameters: none

Optional Parameters:

- `silencesupp`—If this parameter is not present in a DTX session, the default value 1 MUST be assumed.
- `dtxmax`—If this parameter is not present in a DTX session, the default value is 32.
- `dtxmin`—If this parameter is not present in a DTX session, the default value is 12.
- `hangover`—If this parameter is not present in a DTX session, the default value 1 MUST be assumed.

The payload type is dynamic for this codec. EVRC0 specifies the header-free format of the EVRC codec. Only a single frame (20 ms) is allowed in header-free mode.

### **EVRC Supported Options**

Required Parameters: none

Optional parameters:

- `ptime`
- `maxptime`
- `maxinterleave`—If not signaled, the `maxinterleave` length is set to 5.
- `silencesupp`—If this parameter is not present, the default value 1 MUST be assumed.
- `dtxmax`—See `dtxmax` in EVRC0 section.
- `dtxmin`—See `dtxmin` in EVRC0 section.
- `hangover`—See `hangover` in EVRC0 section.

The payload type is dynamic for this codec. The default `ptime` is 20 ms. The `ptimes` 20, 40, and 60 ms are supported for transcoding.

### **EVRC1 Supported Options**

Required parameters: none

Optional parameters:

- `ptime`—See RFC 4566
- `maxptime`
- `fixedrate`—Valid values are 0.5 or 1. If this parameter is not present, 0.5 is assumed.
- `silencesupp`—If this parameter is not present, 1 MUST be assumed.
- `dtxmax`—See `dtxmax` in EVRC0 section.
- `dtxmin`—See `dtxmin` in EVRC0 section.
- `hangover`—See `hangover` in EVRC0 section.

The payload type is dynamic for this codec. Only the 20 ms `ptime` is supported for transcoding. The media rate will be fixed to either full or half rate depending on the `fixedrate` parameter (half rate is default).

### **Default settings for EVRC encoding**

- CDMA Rate change for Dim and Burst disabled
- CDMA DTX control enabled

### **EVRC Configuration**

In the CLI, refer to EVRC codecs exactly as follows:

EVRC0  
EVRC  
EVRC1

### EVRC-B Codec for Transcoding

The Acme Packet 6300 supports the Enhanced Variable Rate codec, extension B (EVRCB) for transcoding. Three subtypes of the EVRCB codec are supported as media-profiles:

- EVRCB0—This codec is transcodable.
- EVRCB—This codec is transcodable.
- EVRCB1—This codec is transcodable.

To transcode to and from EVRCB, the EVRCB license must be installed.

 **Note:** The Acme Packet 6300 handles optional parameters according to RFC 4788 unless otherwise specified.

#### EVRCB0 Supported Options

EVRCB0 is the header-free format of the EVRCB codec. Only a single frame (20 ms) is allowed in header-free mode.

Required Parameters: none

Optional Parameters:

- `silencesupp` — If this parameter is not present in a DTX session, the default value 1 MUST be assumed.
- `dtxmax` — If this parameter is not present in a DTX session, the default value is 32.
- `dtxmin` — If this parameter is not present in a DTX session, the default value is 12.
- `hangover` — If this parameter is not present in a DTX session, the default value 1 MUST be assumed.

The payload type is dynamic for this codec. EVRCB0 specifies the header-free format of the EVRCB codec. Only a single frame (20 ms) is allowed in header-free mode.

#### EVRCB Supported Options

Required Parameters: none

Optional parameters:

- `ptime`
- `maxptime`
- `maxinterleave`—If not signaled, the `maxinterleave` length is set to 5.
- `silencesupp`—If this parameter is not present, the default value 1 MUST be assumed.
- `dtxmax`
- `dtxmin`
- `hangover`

The payload type is dynamic for this codec. The default `ptime` is 20 ms. Only the 20 ms `ptime` is supported for transcoding.

#### EVRCB1 Supported Options

Required parameters: none

Optional parameters:

- `ptime`—See RFC 4566
- `maxptime`
- `fixedrate`—Valid values are 0.5 or 1. If this parameter is not present, 0.5 is assumed.
- `silencesupp`—If this parameter is not present, 1 MUST be assumed.
- `dtxmax`
- `dtxmin`

- hangover

The payload type is dynamic for this codec. Only the 20 ms ptime is supported for transcoding. The media rate will be fixed to either full or half rate depending on the fixedrate parameter (half rate is default).

### Default fixed settings for EVRCB encoding

- CDMA Rate change for Dim and Burst disabled
- CDMA DTX control enabled

### EVRCB Configuration

In the ACLI, refer to EVRCB codecs exactly as follows:

```
EVRCB0
EVRCB
EVRCB1
```


## Configuring Transcoding

### Codec Policy Configuration

Transcoding is configured by creating codec policies and referencing them from a realm configuration.

#### ACLI Configuration Instructions

The parameters that you can configure are name, allow-codecs, add-codecs-on-egress, order-codecs, and ptime. The following section provides brief explanations of how these parameters work, and how you configure each of them.

 **Note:** A single codec policy can be reused for any number of realms.

To access the configuration parameters for codec policies:

1. In Superuser mode, type configure terminal and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type media-manager and press Enter.

```
ACMEPACKET (configure) # media-manager
```

3. Type **codec-policy** and press Enter.

```
ACMEPACKET (media-manager) # codec-policy
```

From this point, you can start configuring your codec policy.

#### Naming Codec Policies

The codec policy's name is important not only because it uniquely identifies the policy, but because it is the name you will enter into your realm configuration's **codec-policy** parameter. It is important to apply the correct policy to the appropriate realm.

To set the codec policy's name:

**name**—Set the name for this codec policy, and note it for future reference when you apply codec policies to realms. This parameter is required, and has no default.

```
ACMEPACKET (codec-policy) # name private
```

#### Removing Allowing and Adding Codecs

The Oracle Communications Session Border Controller removes and allows codecs using the **allow-codecs** parameter. Refer to the [Codec Policy Definition](#) section of this chapter for configuration information.

## Transcoding

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- **allow-codecs**—The **allow-codecs** parameter takes a list of codecs that you want to pass through the Oracle Communications Session Border Controller and can explicitly allow them to remain in the SDP for the next step; codecs not matching the items on this list are removed. This parameter is required.
- **add-codecs-on-egress**—The **add-codecs-on-egress** parameter sets the codecs that the Oracle Communications Session Border Controller adds to an offer if that codec is not already there. This parameter applies only to the egress policy.

For **allow-codecs**, **order-codecs**, and **add-codecs** to codec policies:

You can configure and edit these two transcoding parameters as ACLI lists, meaning that there are **add** and **delete** commands associated with each. You type the name of the parameter, choose the operation you want to perform on the list (adding or deleting), and then specify the data that you want to add or remove.

The examples in the procedure that follows show you how to add to the lists you are configuring. To remove items from the **allow-codecs** list, simply replace the **add** command you see in these example with **delete** and the items you want to remove.

If you want to overwrite previous values, you can enter the command, a Space and the items in the list enclosed in quotes (“”).

1. **allow-codecs**—Enter a list of codecs that are allowed to pass through the Oracle Communications Session Border Controller. Use the syntax in the [Transcodable Codecs](#) section of this chapter. To allow all codecs, enter an asterisk (\*).

```
ACMEPACKET(codec-policy) # allow-codecs *
```

When multiple items are added, enclose them in quotes. For example:

```
ACMEPACKET(codec-policy) # allow-codecs G729 G711 AMR
```

2. **add-codecs-on-egress**—Enter the codecs that you want added to the SDP offer for the egress codec policy. If you leave this parameter blank, then the Oracle Communications Session Border Controller will not add codecs to the SDP answer. This parameter cannot be wildcarded; other possible values are listed in the [Transcodable Codecs](#) section of this chapter.

```
ACMEPACKET(codec-policy) # add-codecs-on-egress G729
```

If you need to modify the list of configured codecs, you must enter the complete list at once.

### Ordering Codecs

Codec policy can specify the order that codecs appear in the SDP offer or answer. Refer to the [Codec Policy Definition](#) section of this chapter for configuration information.

To configure an order which codecs appear in the offer:

**order-codecs**—Enter the order in which you want codecs to appear in the SDP offer or answer. You can enter them in any of the ways described in the preceding explanation.

```
ACMEPACKET(codec-policy) # order-codecs G711 * G729
```

### Transrating Configuration

The following procedure explains how to configure transrating for a codec policy. This codec policy must be applied as an egress codec policy.

To configure forcedptime for a codec policy:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET(configure) # media-manager  
ACMEPACKET(media-manager) #
```

3. Type **network-parameters** and press Enter.



```
ACMEPACKET(media-manager) # codec-policy
ACMEPACKET(codec-policy) #
```

- If you are adding support for this feature to a pre-existing configuration, then you must select the specific configuration instance, using the ACLI **select** command.

```
ACMEPACKET(codec-policy) # select 1
```

You can now configure forced ptime.

- force-ptime**—Set this parameter to **enabled** to enable forced ptime for this codec policy.
- packetization-time**—Enter the ptime in ms to use in the realm where this codec policy is active. Valid values are 10, 20, 30, 40, 50, 60, 70, 80, 90 ms. The default value is 20 ms.
- Save your work using the ACLI **done** command.

## Applying a Codec Policy to a Realm

Once you have configured a codec policy, you apply it to a realm by configuration name.

To apply a codec policy to a realm:

- In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

- Type **media-manager** and press Enter.

```
ACMEPACKET(configure) # media-manager
```

- Type **realm-config** and press Enter.

```
ACMEPACKET(media-manager) # realm-config
```

- codec-policy**—Enter the name of the codec policy that you want to apply to this realm. This value is the same as the one you entered in the name parameter for the codec policy you want to use for this realm. There is no default for this parameter.

```
ACMEPACKET(realm-config) # codec-policy private
```

## Media Profile Configuration


Media profiles must be created and then defined when you want to override the Oracle Communications Session Border Controller's default media profiles.

### ACLI Configuration Instructions and Examples

The parameters that you can configure are name, allow-codecs, add-codecs-on-egress, order-codecs, and ptime. The following section provides brief explanations of how these parameters work, and how you configure each of them.

#### Creating User-Defined Ptime per Codec

To change the Oracle Communications Session Border Controller's default ptime for a specific codec, you must create a media profile configuration element. In the **parameter** parameter, you set the ptime to the value of your choosing.

 **Note:** The frames-per-packet parameter in the media profile configuration element is NOT used for setting a user defined ptime for that codec.

To configure a new ptime value for a codec:

- In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

- Type **session-router** and press Enter.

```
ACMEPACKET(configure) # session-router
ACMEPACKET(session-router) #
```

- Type **media-profile** and press Enter.

## Transcoding

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```
ACMEPACKET(session-router) # media-profile
ACMEPACKET(media-profile) #
```

If you are adding ptime to a pre-existing media profile, then you must select (using the ACLI select command) the configuration that you want to edit. If you are adding ptime to an undefined media profile, you must create it first.

4. **name**—Type the name of the codec for which you are creating a new default ptime.

```
ACMEPACKET(media-profile) # name pcmu
```

5. **payload-type**—Enter the well-known payload type for this codec.

```
ACMEPACKET(media-profile) # payload-type 0
```

6. **parameter**—Set the ptime by typing **parameter**, a Space, **ptime=**, the new ptime value. Then press Enter. For example:

```
ACMEPACKET(media-profile) # parameter ptime=40
```

7. Save your work using the ACLI **done** command.

## Media Type Subnames

---

You can define multiple versions of a media profile for a single codec by using the subnames feature. You can then reference the new media profile by a combination of the media profile name and media profile subname.

Some media types are not unique per just their value in an SDP **m=** line, they must be uniquely identified by looking at additional SDP parameters. For example, you can define a media profile for G729, when only the parameter and value **annexb=yes** is present in the SDP. By creating a media profile + subname that defines both a media type and parameter, you can perform various operations on G729 only when **annexb=yes** is encountered.

Some applications of media type subnames are:

- maintaining different versions of the same codec with different bandwidth ceilings
- maintaining different versions of the same codec with different ptimes
- grouping codecs by using customer as a subname
- grouping codecs by using realm as a subname

## SDP Parameter Matching

This feature matches parameters in the **a=fmtp**, codec-specific SDP **a=** line. It does not try to match a global **m=** line attribute like **a=ptime**.

## Using Subnames with Codec Policies

Media profiles are defined and referenced in the ACLI by a name and subname in the following format

```
<name>::<subname>
```

If no subname has been created for a media profile, you may continue using the media profile name without any subname specifier.

For example, to remove a media profile and subname configured as PCMU::customer1 from all SDP entering the egress realm, you would configure the codec policy **allow-codecs** parameter as follows:

```
allow-codecs PCMU::customer1:no
```

## Subname Syntax and Wildcarding

You can wildcard one or both portions (name and subname) of a media type and subname pair:

- When you wildcard the **name** portion of the value, you can provide a specific subname that the Oracle Communications Session Border Controller uses to find matching media profiles.
- When you wildcard the subname portion of the value, you can provide a specific **name** that the Oracle Communications Session Border Controller uses to find matching media profiles.

The following table defines and explains subname wildcarding and syntax:

Syntax	Example Value	Description
<name>	PCMU	Matches any and all media profiles with the name value configured as PCMU. This entry has the same meaning as a value with this syntax: <name>::*.
<name>::	PCMU::	Matches a media profile with the name with the name value configured as PCMU with an empty subname parameter.
<name>::*	PCMU::*	Matches any and all media profiles with the name value configured as PCMU with any and all subname configured.
<name>:: <subname&gt;< td=""> <td>PCMU::64k</td> <td>Matches a media profiles with the name with the name value configured as PCMU with the subname parameter set to 64k.</td> </subname&gt;<>	PCMU::64k	Matches a media profiles with the name with the name value configured as PCMU with the subname parameter set to 64k.
*	*	Matches anything, but does not have to be a defined media profile.
*::*	*::*	Matches any and all media profiles, but requires the presence of media profile configurations.
*:: <subname&gt;< td=""> <td>*::64k</td> <td>Matches all media profiles with this subname. You might have a group of media profiles with different names, but the same subname value.</td> </subname&gt;<>	*::64k	Matches all media profiles with this subname. You might have a group of media profiles with different names, but the same subname value.
*::	*::	Matches any media profiles with an empty subname parameter.
::	::	Invalid
::*	::*	Invalid

### Wildcarding add-codecs-on-egress

It is important to note that you may not configure **add-codecs-on-egress** with a wildcarded subname in a codec policy. You may only add a specific instance of a media type.

Valid:

```
add-codecs-on-egress PCMU
add-codecs-on-egress PCMU::customer1
```

Invalid:

```
add-codecs-on-egress PCMU::*
```

## Media Type and Subname Configuration

To use media type subnames with a codec policy, you must first configure a media profile and subname. Then you can configure a codec policy with a media type and subname pair for your application

To use configure a media type and subname:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **media-profile** and press Enter.

```
ACMEPACKET(session-router)# media-profile
ACMEPACKET(media-profile)#
```

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4. **name**—Type the name of the codec for which you are creating a new default ptime.

```
ACMEPACKET (media-profile) # name g729
```

5. **subname**—Enter a description for the use of this subname

```
ACMEPACKET (media-profile) # subname annexb=yes
```

You may now configure this subname's unique attributes. PCMU is created with ptime of 30 in this example.

6. **parameters**—Set the ptime by typing **parameter**, a Space, **ptime=**, the new ptime value. Then press Enter. For example:

```
ACMEPACKET (media-profile) # parameter annexb=yes
```



**Note:** Remember to configure all additional, required media profile parameters, or they will inherit default values.

7. Save your work using the ACLI **done** command.

## Codec Policy Configuration with a Media Type with a Subname

To configure a codec policy with a media type with subname:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
```

3. Type **codec-policy** and press Enter.

```
ACMEPACKET (media-manager) # codec-policy
```

4. Use the ACLI **select** command to select a codec policy.

```
ACMEPACKET (codec-policy) # select 1
```

You may now enter a media profile with subname to any parameter in the codec policy that accepts a media profile.

5. **allow-codecs**—Enter a list of codecs that this codec policy allows or denies from passing through the Oracle Communications Session Border Controller. To allow all codecs, enter an asterisk (\*).

```
ACMEPACKETcodec-policy# allow-codecs g729::annexb=yes:no
```

6. Save and activate your configuration.

## Codec and Conditional Codec Policies for SIP

---

The Oracle Communications Session Border Controller has the ability to add, strip, and reorder codecs for SIP sessions. This builds on the Oracle Communications Session Border Controller's pre-existing abilities to route by codec and re-order one codec in an SDP offer by allowing you to configure the order of multiple codecs and to remove specific codecs within the media descriptions in SDP offers.

You can enable the Oracle Communications Session Border Controller to perform these operations on SDP offers by configuring codec policies. Codec policies are sets of rules that specify the manipulations to be performed on SDP offers. They are applied on an ingress and egress basis using the realm and session agent configurations.

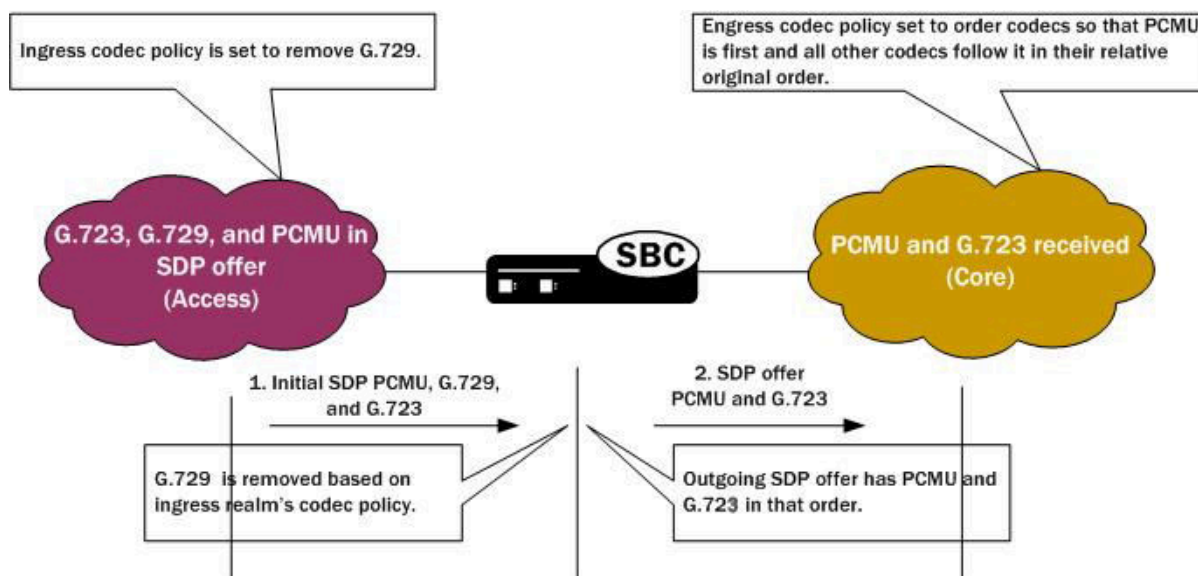
Oracle Communications Session Border Controller supports three types of codec policies:

- **Ingress policy**—Codec policy that the Oracle Communications Session Border Controller applies to the SDP offer for incoming traffic
- **Egress policy**—Codec policy that the Oracle Communications Session Border Controller applies to the SDP offer for traffic leaving the Oracle Communications Session Border Controller
- **Conditional policy**—Codec policy that the Oracle Communications Session Border Controller applies to the SDP offer for traffic leaving the Oracle Communications Session Border Controller. A conditional policy differs from

an egress policy in providing the capability to perform standard codec manipulations (add, strip and re-order) dynamically, based on the codec list and associated parameters contained in the original SDP offer. Refer to [Conditional Codec Policies](#) for specific details regarding the use and construction of these policies.

The Oracle Communications Session Border Controller applies codec policies during the offer phase of media format negotiation. If codec manipulation is enabled, then the Oracle Communications Session Border Controller performs the modification according to the specific policy and forwards on the traffic.

For example, when the Oracle Communications Session Border Controller receives a SIP INVITE with SDP, it refers to the realm through which the INVITE arrived and performs any manipulations specified by an ingress codec policy that may have been assigned to the ingress realm. With the media description possibly changed according to the ingress codec policy, the Oracle Communications Session Border Controller passes the SDP offer to the outgoing realm so that an egress codec policy can be applied. Note that the SDP to be evaluated by the egress codec policy may match the original SDP, or it may have been changed during transit through the ingress realm. After applying the egress coded policy, the Oracle Communications Session Border Controller forwards the INVITE.



Since the offer-answer exchange can occur at different stages of SIP messaging, the assigned ingress and egress roles follow the media direction rather than the signaling direction. It might be, for example, that the offer is in an OK that the Oracle Communications Session Border Controller modifies.

You can apply codec policies to realms and to session agents; codec policies configured in session agents take precedence over those applied to realms. However, it is not required that there be both an ingress and an egress policy either for realms or for session agents. If either one is unspecified, then no modifications take place on that side. If neither ingress nor egress policies specified, SDP offers are forwarded as received.

## Relationship to Media Profiles

For each codec that you specify in a codec policy, there must be a corresponding media profile configuration on the Oracle Communications Session Border Controller. You configure media profiles in the ACLI via the session-router path. In them, you can specify codec type, transport protocol, required bandwidth, and a number of constraints.

## Manipulation Modes

You can configure a codec policy to perform several different kinds of manipulations:

- Allow—List of codecs that are allowed for a certain codec policy; if a codec does not appear on this list, then the Oracle Communications Session Border Controller removes it. You can wildcard this list with an asterisk (\*) so that all codecs are allowed. Further, you can create exceptions to a wildcarded allow list.

## Transcoding

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- You make an exception to the wildcarded list of codecs by entering the codec(s) that are not allowed with a **no** attribute. This tells the Oracle Communications Session Border Controller to allow all codecs except the one(s) you specify.

```
ACMEPACKET(codec-policy) # allow-codecs (* PCMA:no)
```

- You can also create exceptions to allow lists such that audio or video codecs are removed. However, when the allow list specifies the removal of all audio codecs and an INVITE arrives at the Oracle Communications Session Border Controller with only audio codecs, the Oracle Communications Session Border Controller behaves in accordance with RFC 3264. This means that the resulting SDP will contain one attribute line, with the media port for the media line set to 0. The terminating side will need to supply new SDP in its reply because the result of the manipulation is the same as an INVITE with no body.

```
ACMEPACKET(codec-policy) # allow-codecs (* audio:no)
```

- Order—List of the codecs where you specify their preferred order in the outgoing media offer. The Oracle Communications Session Border Controller arranges matching codecs according to the rule you set, and any remaining ones are added to the list in the same relative order they took in the incoming media offer. If your list specifies a codec that is not present, then the ordering proceeds as specified but skips the missing codec.

You can use an asterisk (\*) as a wildcard in this list, too. The placement of the asterisk is key, as you can see in the following examples:

- For an order rule set this way

```
ACMEPACKET(codec-policy) # order (A B C *)
```

codecs A, B, and C will be placed at the front of the codec list in the order specified; all other codecs in the offer will follow A, B, and C in the same relative order they had in the original SDP offer.

- For an order rule set this way:

```
ACMEPACKET(codec-policy) # order (* A B C)
```

codecs A, B, and C will be placed at the end of the codec list in the order specified; all other codecs in the offer will come before A, B, and C in the same relative order they had in the original SDP offer.

- For an order rule set this way

```
ACMEPACKET(codec-policy) # order (A * B C)
```

codec A will be placed at the beginning of the codec list, to be followed by all other codecs in the offer in the same relative order they had in the original SDP offer, and then B and C will end the list.

- Force—An attribute you can use in the allow list with one codec to specify that all other codecs should be stripped from the outgoing offer. You can specify multiple forced codecs in your rules.
  - If you set multiple codecs in the allow list and one of them is forced, then the outgoing offer will contain the forced codec.
  - If you set multiple codecs in the allow list and the one that is forced is not present in the offer, then the Oracle Communications Session Border Controller will select a non-forced codec for the outgoing offer.

```
ACMEPACKET(codec-policy) # allow (PCMU G729:force)
```

You cannot use the force attribute with a wildcarded allow list.

- No—An attribute that allows you to strip specified codecs or codec types from a wildcarded allow list.

```
ACMEPACKET(codec-policy) # allow (* PCMA:no)
```

## In-Realm Codec Manipulation

In addition to being able to apply codec policies in realms, the realm configuration supports a setting for determining whether codec manipulation should be applied to sessions between endpoints in the same realm.

In-realm codec manipulation can be used for simple call flows that traverse two realms. If the originating and terminating realms are the same, the Oracle Communications Session Border Controller checks to see if you have enabled this capability. If you have enabled it, then the Oracle Communications Session Border Controller performs the specified manipulations. If this capability is not enabled, or if the realm's media management in realm (**mm-in-**

**realm**) setting is disabled, then the Oracle Communications Session Border Controller does not perform codec manipulations.

For more complex calls scenarios that involve call agent or reinitiation of a call back to the same realm, the Oracle Communications Session Border Controller does not perform in-realm codec manipulation.

## Conditional Codec Policies

A codec policy performs actions conditionally when any of its parameters includes a conditional value. A conditional value includes a target codec paired with a requirement for executing the action. The Oracle Communications Session Border Controller manipulates SDP according to this value pair if the ingress SDP or a previous manipulation to the SDP meets the condition's criteria. The user configures this "conditional manipulation" by extending upon the syntax of three core parameters in the codec-policy configuration element, including:

- `allow-codecs`,
- `add-codecs-on-egress`, and
- `order-codecs`.

The system establishes conditions on a codec policy as a sequence of allowing, adding, and re-ordering. Each step in this sequence can occur with or without conditions. Allowing is required. Any applied policy, whether or not it is conditional, without an allow blocks all traffic. Allow all, using the wildcard asterisk character is a typical setting. Adding only applies to egress policies.

To establish conditions, the user configure the parameter with pairs that consist of target codecs followed by the condition(s) that trigger the action. Each policy parameter can include one or more of these pairs. When configuring a parameter with multiple values, the user encloses them within parenthesis, whether or not there are conditions.

The system processes all policies serially, regardless of whether any includes a condition. Specifically, the system first determines which codecs to allow, then which to add (none on ingress), then the codec order. The system also processes parameter values serially. The user, therefore, must configure all parameter values based on what may have been changed previously. This is particularly important when using both ingress and egress codec policies. Consistent with this serial concept, egress policies operate on what the system presents to them, which includes the results of any ingress policies.

Note that the use of conditional **order-codecs** is often done in conjunction with **add-codecs-on-egress** to define the location of user-added codecs to the list presented at egress. When **order-codecs** is not used, the system places all added codecs at the beginning of the list in the order they were added. It is often desirable to place an added codec in a different position, using **order-codecs**.

## Conditional Codec Lists

Conditional codec policies are constructed using existing ACLI configuration commands — **allow-codecs**, **add-codecs-on-egress**, and **order-codecs** — in conjunction with keywords and operators. Conditions are defined by a continuous character string (no spaces allowed) that starts with the `:(` character sequence and is terminated by a closing parenthesis. For example,

```
ACMEPACKET(codec-policy) # allow-codecs PCMU:(!G729)
```

which can be interpreted as — allow PCMU if the G729 codec is not in the SDP offer after ingress codec policy processing.

An example of using **add-codecs-on-egress** is:

```
ACMEPACKET(codec-policy) # add-codecs-on-egress PCMU:(PCMA)
```

which can be interpreted as — add PCMU if PCMA codec is in the SDP offer after ingress codec policy processing.

If PCMA is in the SDP offer after ingress codec policy processing, the **add-codecs-on-egress** ACLI command is treated as **add-codecs-on-egress PCMU**. If PCMA is not in the SDP offer after ingress codec policy processing, **add-codecs-on-egress** is treated as empty.

Both the conditioned codec and/or the condition itself can contain subnames. For example,

```
ACMEPACKET(codec-policy) # add-codecs-on-egress AMR::ONE:(AMR::TEST0)
```

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which can be interpreted as — add AMR::ONE if AMR::TEST0 codec is in the SDP offer after ingress codec policy processing.

Codecs contained in the condition can be wildcarded. For example,

```
ACMEPACKET(codec-policy) # add-codecs-on-egress AMR::ONE:(AMR::*)
```

which can be interpreted as — add AMR::ONE if any AMR codec is in the offer after ingress codec policy processing.

An example of using **order-codecs** is:

```
ACMEPACKET(codec-policy) # order-codecs (PCMU:(PCMU) *)
```

which can be interpreted as — set the codec order to PCMU followed by the list after ingress codec policy processing if PCMU is not present. When the system adds a codec, the codec goes to the end of the offered list. This conditional format may be used to place an added codec at the front of list.

### Conditional Codec Operators

Three logical operators are available to construct conditional lists

the OR operator (|)

```
ACMEPACKET(codec-policy) # add-codecs-on-egress AMR::ONE:(AMR::*|PCMU)
```

which can be interpreted as — add AMR::ONE if any AMR:: codec is in the SDP offer after ingress codec policy processing, or if PCMU is in the SDP offer after ingress codec policy processing.

the AND operator (&)

```
ACMEPACKET(codec-policy) # add-codecs-on-egress AMR::ONE:(AMR::*&PCMU)
```

which can be interpreted as — add AMR::ONE if any AMR:: codec is in the SDP offer after ingress codec policy processing, and if PCMU is in the SDP offer after ingress codec policy processing.

the NOT operator (!)

```
ACMEPACKET(codec-policy) # add-codecs-on-egress AMR::ONE:(!AMR::*)
```

which can be interpreted as — “add AMR::ONE if no AMR:: codec is in the SDP offer after ingress codec policy processing.

Each operator applies only to the codec immediately following it. Operators are processed left to right until all conditions have been tested. The condition result is accumulated as each of the conditions is processed. For example:

```
ACMEPACKET(codec-policy) #add-codecs-on-egress AMR::ONE:  
(!AMR::TEST1&AMR::TEST0|AMR::TEST2)
```

- Test SDP offer for AMR::TEST1 (a NOT operation).  
If AMR:TEST1 is NOT present, set accumulated result to TRUE.  
If AMR:TEST1 is present, set accumulated result to FALSE.
- Test SDP offer for AMR::TEST0 (an AND operation).  
If AMR is present, no change to accumulated result.  
If AMR is not present, set accumulated result to FALSE.
- Test SDP offer for AMR::TEST2 (an OR operation).  
If AMR is present, set accumulated result to TRUE.  
If AMR is not present, no change to accumulated result.

Multiple conditions can be concatenated; in this case, individual conditions are separated by SPACE characters and the ACLI command argument is bracketed with double quotation marks ...). For example:

```
ACMEPACKET(codec-policy) #add-codecs-on-egress (PCMU G729:(G726) G723:(PCMA))
```

- PCMU is unconditionally added to the egress codec list.



- Process the first condition — G729:(G726)  
If G726 is present, the result is TRUE; add G729 to the egress codec list.  
If G726 is not present, the result is FALSE; do not add G729 to the egress codec list.
- Process the second condition — G723:(PCMA)  
If PCMA is present, the result is TRUE; add G723 to the egress codec list.  
If PCMA is not present, the result is FALSE; do not add G723 to the egress codec list.

## ACLI Instructions and Examples

This section gives instructions and examples for how to configure codec policies and then apply them to realms and session agents. It also shows you how to configure settings for in-realm codec manipulation.

### Creating a Codec Policy

To create a codec policy:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the signaling-related configurations.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **codec-policy** and then press Enter.

```
ACMEPACKET (media-manager) # codec-policy
ACMEPACKET (codec-policy) #
```

4. **name**—Enter the unique name for the codec policy. This is the value you will use to refer to this codec policy when you apply it to realms or session agents. This parameter is required and is empty by default.
5. **allow-codecs**—Enter the list of media format types (codecs) to allow for this codec policy. In your entries, you can use the asterisk (\*) as a wildcard, the force attribute, or the no attribute so that the allow list you enter directly reflects your configuration needs. Enclose entries of multiple values in parentheses ( ( ) ). For more information, refer to the [Manipulation Modes](#) section above.

The codecs that you enter here must have corresponding media profile configurations.

**allow-codecs** can be used to construct ingress, egress, or conditional codec policies. For details of conditional codec policies, refer to [Conditional Codec Policies](#).

6. **add-codecs-on-egress**—Enter the codecs that the Oracle Communications Session Border Controller adds to an egress SDP offer if that codec is not already there. This parameter applies only to egress offers. For more information, refer to the [Manipulation Modes](#) section above.

The codecs that you enter here must have corresponding media profile configurations.

**add-codecs-on-egress** can be used to construct ingress, egress, or conditional codec policies. For details of conditional codec policies, refer to [Conditional Codec Policies](#).

7. **order-codecs**—Enter the order in which you want codecs to appear in the outgoing SDP offer. Remember that you can use the asterisk (\*) as a wildcard in different positions of the order to directly reflect your configuration needs. Enclose entries of multiple values in parentheses ( ( ) ). For more information, refer to the [Manipulation Modes](#) section above.

The codecs that you enter here must have corresponding media profile configurations.

**order-codecs** can be used to construct ingress, egress, or conditional codec policies. For details of conditional codec policies, refer to [Conditional Codec Policies](#).

8. Save and activate your configuration.

Your codec policy configuration will resemble the following example:

```
codec-policy
  name          private
  allow-codecs  g723:no pcmu video:no
  order-codecs  pcmu *
```

### Applying a Codec Policy to a Realm

Note that codec policies defined for session agents always take precedence over those defined for realms.

To apply a codec policy to a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm-config** and press Enter.

```
ACMEPACKET (media-manager) # realm-config
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI **select** command) the realm that you want to edit.

4. **codec-policy**—Enter the name of the codec policy that you want to apply to this realm. By default, this parameter is empty.
5. Save and activate your configuration.

### Applying a Codec Policy to a Session Agent

Note that codec policies that are defined for session agents always take precedence over those that are defined for realms.

To apply a codec policy to a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter.

```
ACMEPACKET (session-router) # session-agent
```

If you are adding support for this feature to a pre-existing session agent, then you must select (using the ACLI **select** command) the realm that you want to edit.

4. **codec-policy**—Enter the name of the codec policy that you want to apply to this realm. By default, this parameter is empty.
5. Save and activate your configuration.

### In-Realm Codec Manipulations

To enable in-realm codec manipulations:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
ACMEPACKET (configure) # media-manager
```

3. Type **realm-config** and press Enter.

```
ACMEPACKET (media-manager) # realm-config
```

If you are adding support for this feature to a pre-existing realm, then you must select (using the ACLI `select` command) the realm that you want to edit.

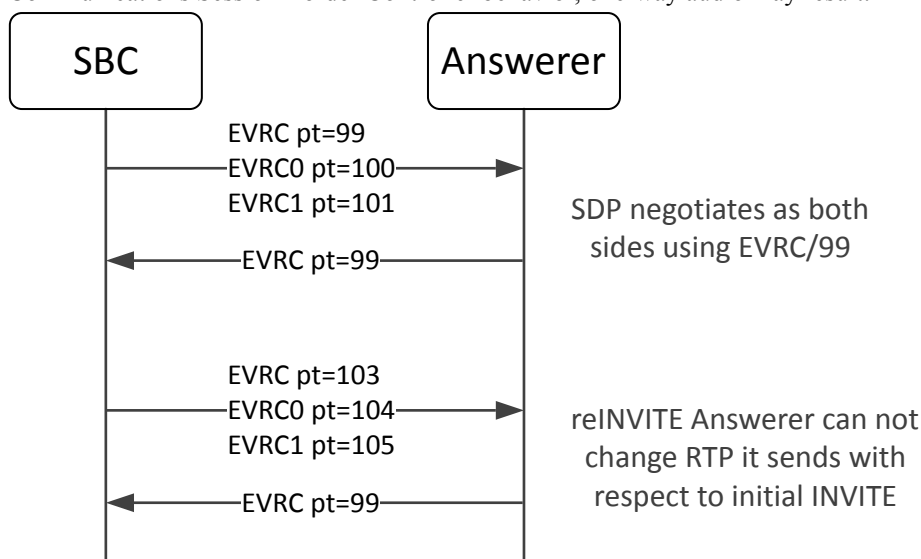
4. **codec-manip-in-realm**—Enter the name of the codec policy that you want to apply to this realm. The default value is `disabled`. The valid values are:
  - enabled | disabled
5. Save and activate your configuration.

## Asymmetric Dynamic Payload Types Enablement

Transcoding Support for Asymmetric Dynamic Payload Types supports calls with asymmetric payload types such that a codec offered with one payload type and answered with another payload type is acceptable to the Oracle Communications Session Border Controller. This behavior requires transcoding resources.

The Oracle Communications Session Border Controller's default behavior when an endpoint answers an SDP offer with a different payload type than offered is to use what the endpoint replied with as if that were what the Oracle Communications Session Border Controller included in its SDP offer. The Oracle Communications Session Border Controller would then expect to receive the same payload type that the endpoint offered. This is referred to as symmetric payload type mapping, whereby the payload type number is the same for both media flows.

The Oracle Communications Session Border Controller needs to support the asymmetric case when codecs with dynamic payload types are used. For example, a call may be set up with the dynamic codec using one payload type value, and a reINVITE may use a different payload type value for the same codec. Because of the range of remote UE equipment's behavior in a reINVITE case, the Oracle Communications Session Border Controller can support this via its Asymmetric Dynamic Payload Type feature. That is, some devices do not necessarily use the currently negotiated payload type, they may use previously negotiated payload types. As such devices interact with the default Oracle Communications Session Border Controller behavior, one-way audio may result.



For transcoded calls, enabling this feature places no additional load on the system. But for calls that are not transcoded, this feature consumes transcoding resources.

## Configure Transcoding for Asymmetric Dynamic Payload Types

Transcoding support for asymmetric dynamic payload types enables the Oracle Communications Session Border Controller to perform transcoding when the Real-time Transport Protocol (RTP) is offered with one payload type and is answered with another payload type. Enable transcoding for asymmetric dynamic payload types from the command line.

Before You Begin

## Transcoding

- Confirm that you are in Superuser mode.

### Procedure

1. Access the **media-manager-config** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# media-manager
ACMEPACKET(media-manager-config)#
```

2. Type **select** to begin editing.

```
ACMEPACKET(media-manager-config)# select
ACMEPACKET(media-manager-config)#
```

3. Use the **options +audio-allow-asymmetric-pt** command to enable support for asymmetric payload types.

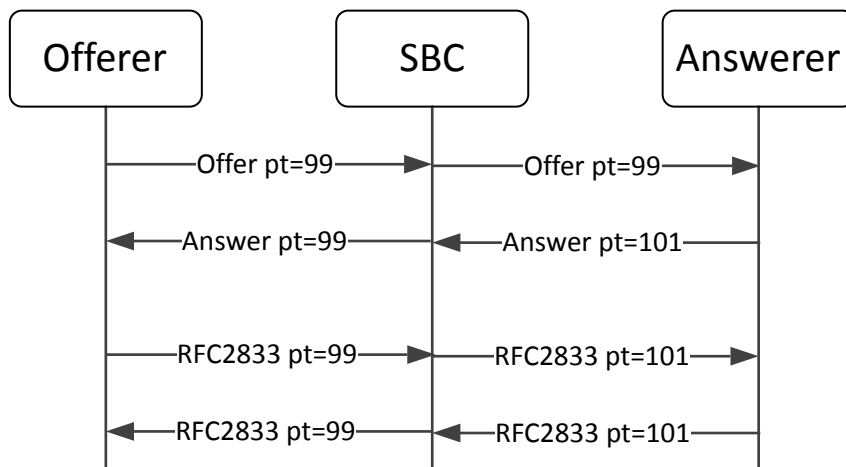
```
ACMEPACKET#(media-manager) options +audio-allow-asymmetric-pt
ACMEPACKET#(media-manager)
```

4. Type **done** to save your configuration.

## Asymmetric Payload Type Support for RFC2833 Interworking

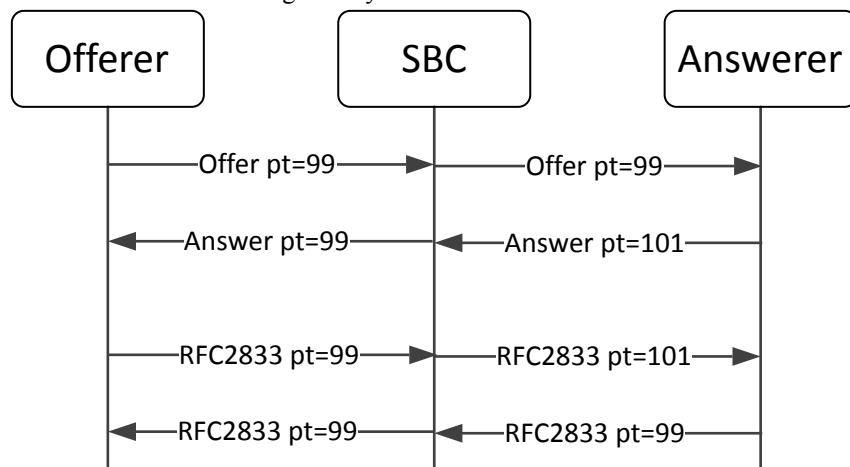
RFC3264 describes when an endpoint sends a different payload-type value that what it receives for a given codec as is asymmetric behavior. Otherwise, when an endpoint can only send the same payload-type value for what it receives, the behavior is symmetric. The Oracle Communications Session Border Controller can be configured to behave in an RFC compliant manner which is to support asymmetric cases. Symmetric cases are covered by baseline functionality. For asymmetric cases, the Oracle Communications Session Border Controller can receive the a different payload-type that was offered.

The Oracle Communications Session Border Controller's default behavior when an endpoint answers an SDP offer with a different payload type value for RFC 2833 telephone-events is to use what the endpoint replied with in subsequent media exchanges, and accept that payload type for received traffic. This is referred to as symmetric payload type mapping, whereby the endpoint replies with the same payload type it was offered for a given media type.



The Oracle Communications Session Border Controller can also handle the asymmetric case by configuration. This is when the SDP media is sent with a different payload type value for the same codec as sent. When the payload-types differ between the offer and answer, then it is asymmetric. If not specifically configured to handle this case, then the

media will not flow through the system.



Add the **rfc2833-allow-asymmetric-pt** option to the sip interface or session agent to enable support of asymmetric payload types for 2833 interworking.

## DTMF Indication over HD Audio Codecs

When performing DTMF transcoding while HD Audio codecs are present, Oracle Communications Session Border Controller accounts for telephone-event tone indication at clock rates that match those of the HD audio codecs.

The telephone-event tone indication's clock rate, when sent alongside an HD codec, must match the audio codec's clock rate. While many non-HD codecs use 8000 Hz clock rates, HD codecs can use clock rates of 16000 Hz. Transcoding processing takes these two telephone-event clock rates into account when performing SDP manipulation. If the wrong clock rate is used, RFC2833 telephone-event indications may be relayed incorrectly.

On the ingress side of the call, if a telephone-event is received, and no audio codec with a matching clock rate is received, the ingress-side SDP response will have the unmatched telephone-event removed.

If the **allow-codec** parameter uses the `:no` tag to remove the last of an audio codec that matches a telephone-event (8000 or 16000) from SDP, then the telephone-event of that clock rate (if present) will be removed from the SDP too.

When codec policy dictates to add AMR-WB, and the received SDP contains PCMU/PCMA and telephone-event (8000), the SDP offer sent from the egress interface will include telephone-event (16000)

If **rfc2833-mode** is set to **preferred**, the Oracle Communications Session Border Controller adds telephone-event 16000 to outbound SDP if AMR-WB is present in the outbound SDP.

On the egress side of the call if the SDP contains a telephone-event without an audio codec with a matching clock rate, an appropriate audio codec will be added. If codec policy adds a telephone-event, then the SBC analyzes the audio codecs in the outbound SDP and ensures that matching telephone events (8000 and/or 16000) are present.

Once the Oracle Communications Session Border Controller determines the SDP to forward, the order of telephone-event clock rates will be modified to match the order of audio codec rates. Thus if AMR-WB is the top codec followed by codecs with 8000 Hz clock rates, the telephone-event with a clock rate of 16000 Hz will be listed above telephone-events with 8000 Hz clock rates.

## RFC2833 and KPML Interworking

Keypad Markup Language (KPML) is used to indicate DTMF tones in SIP messaging. KPML is used by the Key Press Stimulus Package as a SIP Event Notification Package. Endpoints will send DTMF tone indications via NOTIFY messages. The Oracle Communications Session Border Controller can engage in the Event Notification with an endpoint on one call leg and perform digit translation to RFC 2833 telephone-events on the other call leg.

**KPML to RFC2833**

KPML to RFC2833 interworking requires that the INVITE request or response's SDP does not contain telephone-event, and is received from

- A session agent with **kpml-2833-interworking** set to enabled
- A session agent with **kpml-2833-interworking** set to inherited from the SIP interface (with **kpml-2833-interworking** set to enabled)
- A previous hop that is not a session agent, and the SIP interface the message received on has **kpml-2833-interworking** set to enabled.

The egress side of the call must have **rfc2833-mode** set to **preferred** in either the SIP interface or session agent, so that the Oracle Communications Session Border Controller inserts telephone-event in the SDP. Setting **rfc2833-mode** to dual is unsupported. The answerer must respond to the invite accepting of telephone-event media. The Allow-Event: kpml header is removed from the INVITE request or response when KPML-interworking is not set to enabled on the next hop.

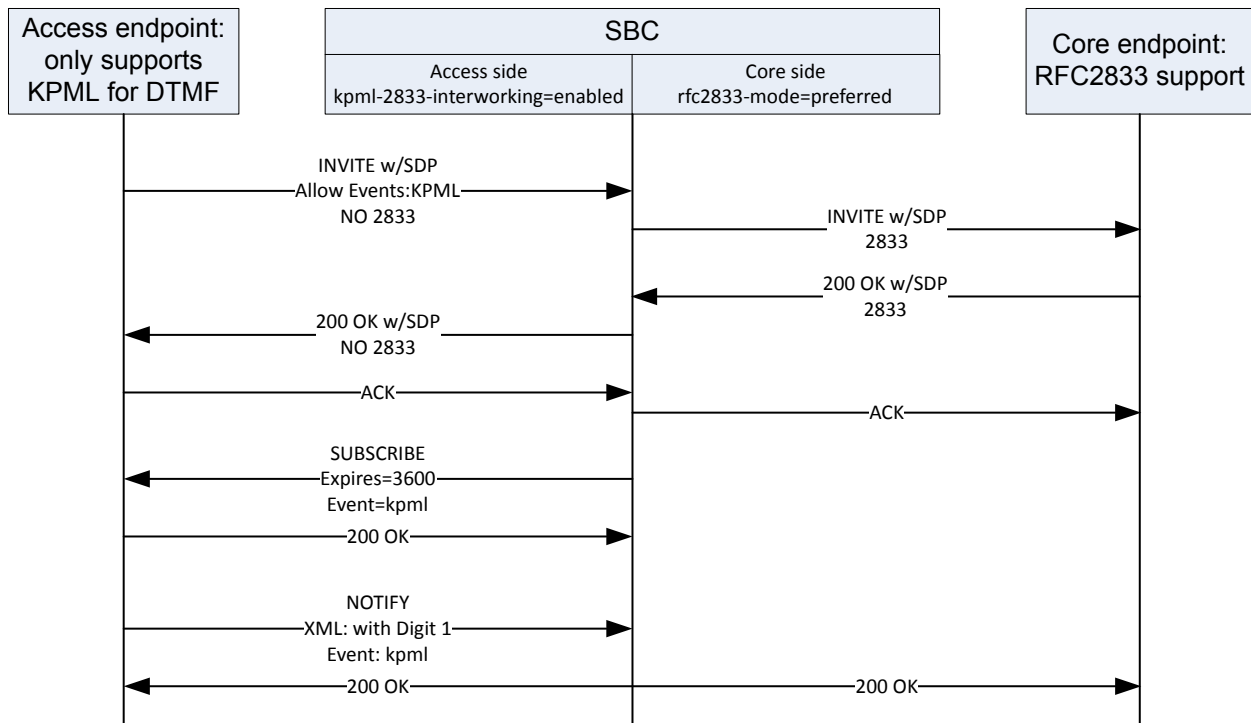
If the INVITE succeeds, the Oracle Communications Session Border Controller replies with a SUBSCRIBE request for the KPML event to the caller.

If the caller replies to the SUBSCRIBE with a 2xx, all subsequent NOTIFY request received on the dialog are processed and replied to with a 200 OK. Each digit in the NOTIFY request will generate a telephone-event on the egress side of the call.

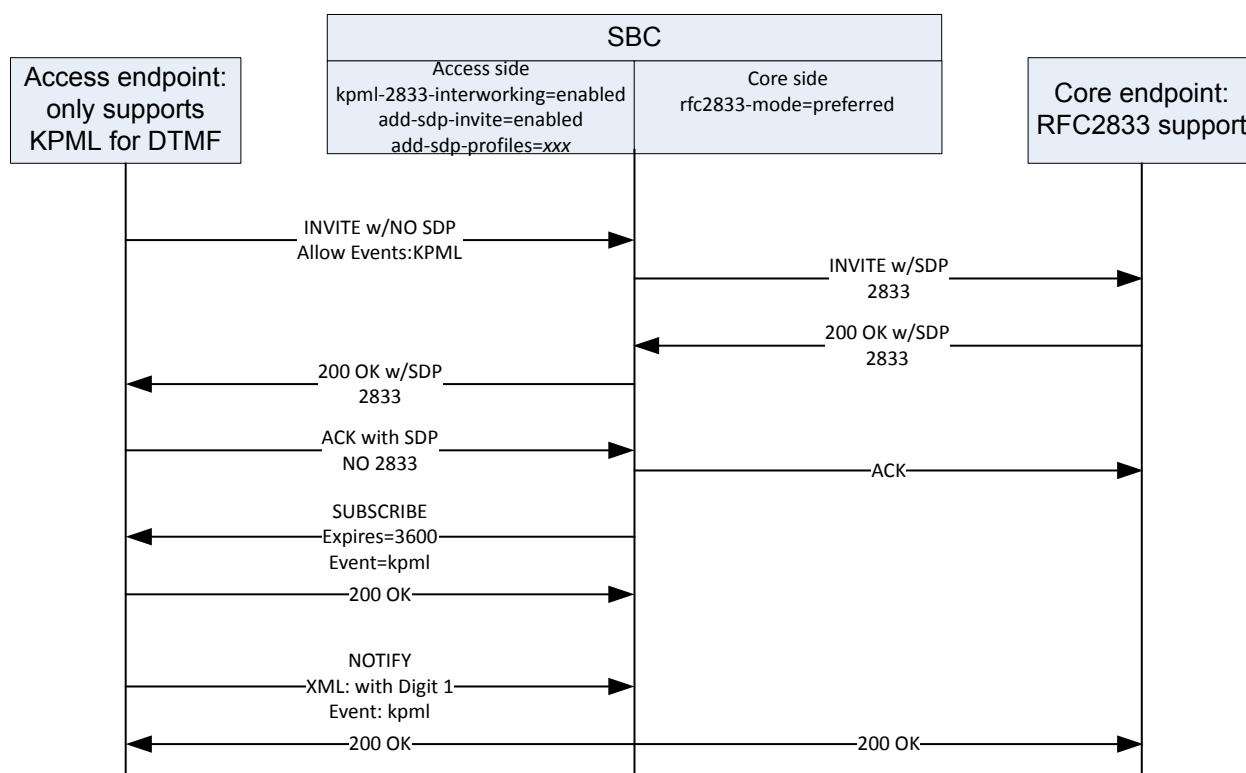
The Oracle Communications Session Border Controller generates a refresh SUBSCRIBE before the subscription expires.

If the call negotiates to RFC 2833 to RFC 2833, no KPML interworking will be performed.

The following diagram shows the standard case where KPML to RFC 2833 interworking is performed.



KPML to RFC 2833 translation is also supported when used in conjunction with SDP insertion on the KPML side of the call. For example:



## RFC 2833 to KPML

RFC2833 to KPML interworking requires that the INVITE request or response's does not contain Allow-Event: kpml. When sending an INVITE when RFC2833 interworking applies, an Allow-Event: kpml header is added to the INVITE when the message's next hop on egress is either :

- A session agent with **kpml-2833-interworking** set to enabled
- A session agent with **kpml-2833-interworking** set to inherited from the SIP interface (with **kpml-2833-interworking** set to enabled)
- Not a session agent, and the SIP interface the message is sent from has **kpml-2833-interworking** set to enabled.

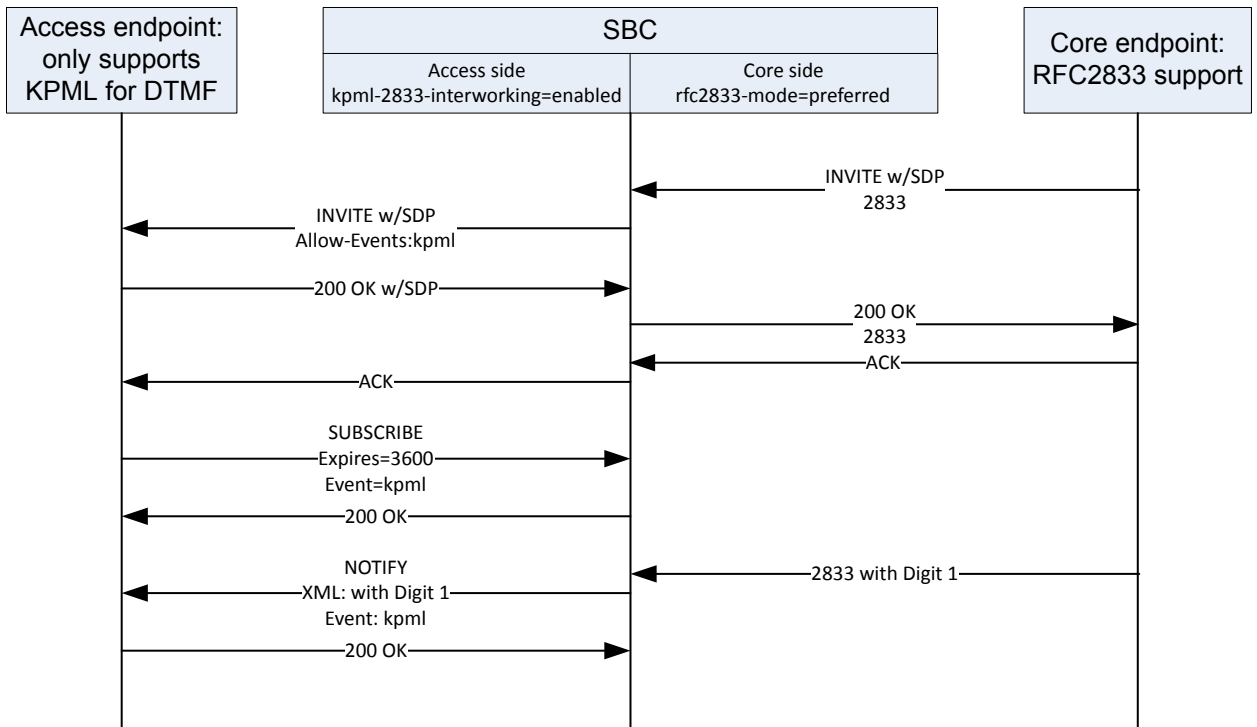
If the scenario passes the above test and the Allow-Event: kpml header is added to the INVITE:

When the Oracle Communications Session Border Controller subsequently receives a SUBSCRIBE request within the INVITE created dialog for the kpml event, it will accept the subscription and respond with a 200 OK. At this point, the Oracle Communications Session Border Controller can generate a KPML NOTIFY request with a KPML digit corresponding to each 2833 telephone-event received from the far end until

- the INVITE dialog is terminated due to a BYE
- the subscription expires
- the subscription is terminated with a subscribe request Expires: 0

The following diagram shows a typical RFC2833 to KPML interworking call flow. Note the following:

- While the example has SDP in the INVITE, delayed offer scenarios where the SDP exchange occurs in the 200OK and the ACK are supported
- The **rfc2833-mode** parameter in the in either the SIP interface or session agent, is considered in the interworking. See next section:
- If the call negotiates to RFC 2833 to RFC 2833, no KPML interworking will be performed.
- In RFC 2833 to KPML scenarios, if the SUBSCRIBE is not received quickly enough, RFC 2833 digits will be dropped.



**Table 1: RFC2833 to KPML interaction with rfc2833-mode**

Originator	Terminator	SBC Behavior
rfc2833-mode=transparent, offers 2833 SDP	rfc2833-mode=transparent, answers NO 2833 SDP	Forwards 2833 offer, Adds Allow-Event: kpml, passes non-2833 capability to Originator (no 2833 support)
rfc2833-mode=preferred, offers 2833 SDP	rfc2833-mode=transparent, answers NO 2833 SDP	Forwards 2833 offer, Adds Allow-Event: kpml, inserts 2833 capability to Originator, performs 2833 to KPML translation
rfc2833-mode=transparent, offers 2833 SDP	rfc2833-mode=preferred, answers NO 2833 SDP	Forwards 2833 offer, Adds Allow-Event: kpml, passes non-2833 capability to Originator (no 2833 support)
rfc2833-mode=preferred, offers 2833 SDP	rfc2833-mode=preferred, answers NO 2833 SDP	Forwards 2833 offer, Adds Allow-Event: kpml, inserts 2833 capability to Originator, performs 2833 to KPML translation

### KPML-2833 Interworking Configuration

To configure KPML - 2833 interworking:

1. In Superuser mode, type `configure terminal` and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type and `session-router` press Enter.

```
ACMEPACKET(configure)# session-router
```



3. Type **sip-interface** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. **kpml-interworking**—Set this parameter to enabled to use KPML-2833 interworking.
5. Save and activate your configuration.

## KPML-2833 Interworking on a Session Agent Configuration

To configure KPML - 2833 interworking on a session agent:

Enter the prerequisites here (optional).

Enter the context of your task here (optional).

1. Access the **session-agent** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)
```

2. Select the **session-agent** object to edit.

```
ACMEPACKET(session-agent)# select
<hostname>:
1: 192.168.100.101:1813

selection: 1
ACMEPACKET(session-agent)#
```

3. **kpml-interworking**—Set this parameter to enabled for the Oracle Communications Session Border Controller to interwork RFC2833 from the other call leg to the call leg running through this session agent. This parameter may be set to inherit to use the **kpml-interworking** value configured on the receiving SIP interface.
4. Type **done** to save your configuration.

## Maintenance and Troubleshooting

### show mbcd errors

The **show mbcd errors** command displays statistics related to MBCD task errors. The following fields are explained:

- XCode Internal Errors—Number of uncategorized errors due to Transcoding session error.
- XCode Alloc Errors—Number of times that buffer allocation failed for transcoding tasks.
- XCode Update Errors—Number of errors encountered when attempting to update an entry in the Transcoding table upon receipt of the first packet for a media flow.
- XCode Delete Errors—Number of errors encountered when attempting to delete an entry in the Transcoding table.
- XCode Over Cap Errors—Number of Transcoding sessions denied once session capacity is reached.
- XCode Over License Cap—Number of Transcoding sessions denied once license capacity is reached.

```
ACMEPACKET# show mbcd errors
13:22:50-126
MBC Errors/Events          Recent      Total      PerMax
Client Errors              0           0           0
Client IPC Errors          0           0           0
Open Streams Failed        0           0           0
Drop Streams Failed        0           0           0
Exp Flow Events            0           0           0
Exp Flow Not Found         0           0           0
```

## Transcoding

Transaction Timeouts	0	0	0
Server Errors	0	0	0
Server IPC Errors	0	0	0
Flow Add Failed	180	180	180
Flow Delete Failed	0	0	0
Flow Update Failed	0	0	0
Flow Latch Failed	0	0	0
Pending Flow Expired	0	0	0
ARP Wait Errors	0	0	0
Exp CAM Not Found	0	0	0
Drop Unknown Exp Flow	0	0	0
Drop/Exp Flow Missing	0	0	0
Exp Notify Failed	0	0	0
Unacknowledged Notify	0	0	0
Invalid Realm	0	0	0
No Ports Available	0	0	0
Insufficient Bandwidth	0	0	0
Stale Ports Reclaimed	0	0	0
Stale Flows Replaced	0	0	0
Telephone Events Gen	0	0	0
Pipe Alloc Errors	0	0	0
Pipe Write Errors	0	0	0
Not Found In Flows	0	0	0
XCode Internal Errors	0	0	0
XCode Alloc Errors	0	0	0
XCode Update Errors	0	0	0
XCode Delete Errors	0	0	0
XCode Over Cap Errors	180	180	180
XCode Over License Cap	0	0	0
SRTP Capacity Exceeded	0	0	0

### show xcode api-stats

The **show xcode api-stats** command shows the client and server side message counts for the XClient and XServer software components. The main messages are allocate, update, and free of the transcoding resource. The command uses a 100 second window to show recent counts within the sliding window as well as the total and per max (maximum in a sliding window interval). This command is useful for comparing the client and server side counts and seeing where errors may have occurred with the transcoding resources.

```
ACMEPACKET#show xcode api-stats
----- Client -----
Message/Event      Recent      Total      PerMax
-----
Allocs             0           5197      4897
Updates            0           1776      1676
Frees              0           6355      6015
Error-Allocs       0            0          0
Error-Updates      0            0          0
Error-Frees        0            0          0
Total              0           13328     12588
----- Server -----
Recent      Total      PerMax
-----
Allocs      0           6355      6055
Updates     0            888       788
Frees       0           6355      6015
Error-Allocs 0            45         45
Error-Updates 0            888        888
Error-Frees  0             0          0
Total       0           14531     13791
```

### show xcode dbginfo

The debug information command shows the packet API statistics for the host to DSP path. There is one session/connection opened with each DSP. The command displays the total packet counts as well as the round trip time statistics for the packets. The recent field shows the count since the last time the command was executed

```
ACMEPACKET#show xcode dbginfo
Startup Time      : 2006-09-08 01:11:50.522
Last Clear Time   : 2006-09-08 01:11:50.522
Last Read Time    : 2006-09-08 17:14:52.351
Current Time      : 2006-09-08 17:14:52.351
Up Time           : 0 Days, 16 Hours 3 Minutes 2 Seconds
-- Life Time --   -- Recent --
```

```

PktApiStats:
  OpenConnectionCnt      =          2
  OpenSessionCnt         =          2
  TotalPktSentCnt        =    21051      21051
  TotalPktRecvCnt        =    21041      21041
  TotalPktRecvEventCnt   =          0          0
  TotalPktRecvDataCnt    =          0          0
  TotalPktRejectCnt      =          5          5
  TotalPktTimeoutCnt     =          0          0
  TotalPktInvalidCnt     =          0          0
  TotalPktDropCnt        =          0          0
  TotalPktDropEventCnt   =          0          0
  TotalPktDropDataCnt    =          0          0
  TotalPktLateRspCnt     =          0          0
  LowestRoundTripMs      =          1
  HighestRoundTripMs     =         2010
  LowestExtractTimeMs    =          1
  HighestExtractTimeMs   =        13320
  HighestTransportRxTimeMs =          0
  ulHighestTransportNoRxTimeMs =          0

```

## Active and Period Statistics for EVRC and other Codecs

Codec use per-realm statistics include more detail by breaking out the EVRC codecs into their specific variants. That is, the system keeps track of EVRC0, EVRC1, EVRCB, EVRCB0, and EVRCB1 codec use per-realm on an explicit basis. These 5 counts are also available for SNMP query and are added to the `ap-codec.mib` file. Additionally, The Oracle Communications Session Border Controller now maintains counts for all codecs that appear in the **show sipd codecs <realm>** command for Active, Recent High, and Lifetime High periods.

## show sipd codecs

The **show sipd codecs <realm ID>** command displays media-processing statistics per SIP traffic. This command displays statistics per realm and requires a realm argument.

### Session Based Statistics

The first 3 statistics listed by the **show sipd codecs** are session based. These statistics are titled Transcoded, Transrated, and Transparent.

- **transcoded**—counts of sessions that use the Transcoding NIU's TCUs to transcode between two or more codes.
- **transrated**—counts of sessions that use the Transcoding NIU's TCUs to change the packetization interval among dialogs in the session.
- **transparent**—counts of sessions that require no TCU hardware intervention (all end-to-end media uses the same codec)

A value of "none" which is not counted in the statistics is set when there is no media at all or media is not yet negotiated. Sessions within the same realm are counted only once.

These are meter type counters, and thus have an "active" count as well as total lifetime values. The media-processing state of the session only can increase in precedence (highest=transcoded, transrated, transparent, lowest=none). Thus, if a session begins as transcoded, and then is re-negotiated to transparent later by a re-INVITE, it is still considered transcoded. However, if a session begins as transparent, it can go to transcoded by a re-INVITE. In such a case, the total counts for both transparent and transcoded would be incremented. If there are several media lines, the highest precedence is used for the session.

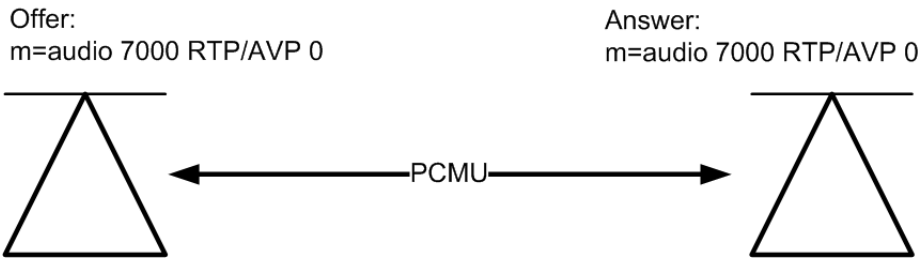
### Flow Based Statistics

The remaining lines of the **show sipd codecs** command track the number of codecs in established sessions. The 'Other' type refers to unknown codecs. Only the Recent-Total and other lifetime columns are populated; the Active and Recent High are not applicable. These counts represent each SDP m= line emanating in the queried realm. Refer to the following examples:

## Transcoding

### Single audio stream example

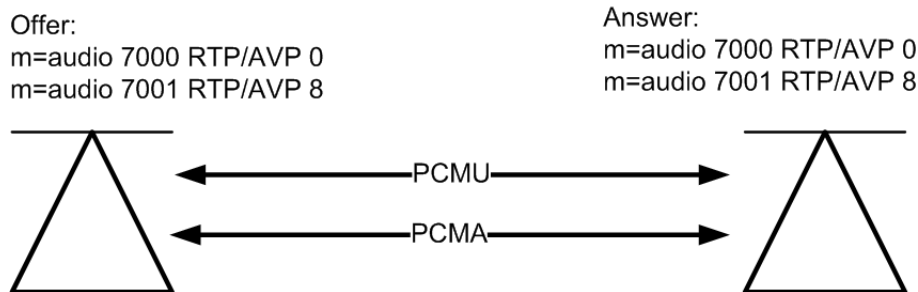
The following diagram shows an intra-realm session with one audio stream using the PCMU codec. Once the session is established, the PCMU count in the `show sipd codecs` output is 2.



If the session originator and terminator in the previous diagram exist in two different realms, you must execute the `show sipd codecs` command twice, once for each realm. A single PCMU count will be reflected in each respective query because only one m= line emanates from each realm.

### Multiple audio stream example

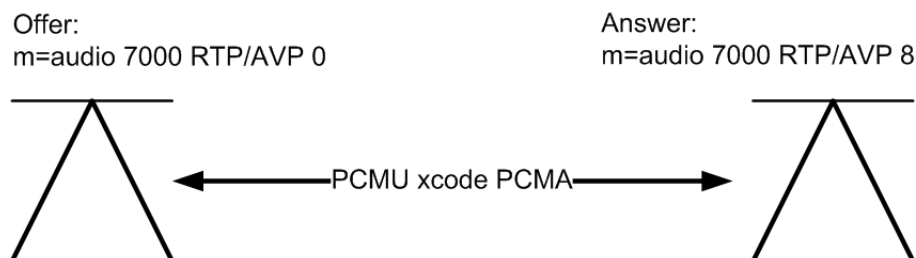
The following diagram shows an intra-realm session with two audio streams. Each stream uses a different codec. Once the session is established, the PCMU count in the `show sipd codecs` output is 2, and the PCMA count is 2.



If the session originator and terminator in the previous diagram exist in two different realms, you must execute the `show sipd codecs` command twice, once for each realm. A single PCMU count and a single PCMA count will be reflected in each respective query because two m= lines emanate from each realm.

### Transcoded audio stream example

The following diagram shows an intra-realm transcoding scenario where the originator and terminator are using different audio codecs. The Oracle Communications Session Border Controller transcodes the media, which is invisible to the endpoints. Once the session is established, the PCMU count in the `show sipd codecs` output is 1, and the PCMA count is 1.



If the session originator and terminator in the previous diagram exist in two different realms, you must execute the `show sipd codecs` command twice, once for each realm. A single PCMU count appears in one query and a single PCMA count appears in the other query because only one m= line emanate from each realm.

An example `show sipd codecs <realm ID>` command follows:

```

ACMEPACKET# show sipd codecs net172
09:20:04-30 Realm net172
Codec Statistics

```

	---- Recent ----			----- Lifetime -----		
	Active	High	Total	Total	PerMax	High
Transcoded	0	0	0	0	0	0
Transrated	0	0	0	0	0	0
Transparent	0	0	0	0	0	0
PCMU Count	0	0	0	0	0	0
PCMA Count	0	0	0	0	0	0
G722 Count	0	0	0	0	0	0
G723 Count	0	0	0	0	0	0
G726-16 Count	0	0	0	0	0	0
G726-24 Count	0	0	0	0	0	0
G726-32 Count	0	0	0	0	0	0
G726-40 Count	0	0	0	0	0	0
G728 Count	0	0	0	0	0	0
G729 Count	0	0	0	0	0	0
GSM Count	0	0	0	0	0	0
iLBC Count	0	0	0	0	0	0
H261 Count	0	0	0	0	0	0
H263 Count	0	0	0	0	0	0
T38 Count	0	0	0	0	0	0
AMR Count	0	0	0	0	0	0
AMR-WB Count	0	0	0	0	0	0
EVRC Count	0	0	0	0	0	0
EVRC0 Count	0	0	0	0	0	0
EVRC1 Count	0	0	0	0	0	0
EVRCB Count	0	0	0	0	0	0
EVRCB0 Count	0	0	0	0	0	0
EVRCB1 Count	0	0	0	0	0	0
Other Count	0	0	0	0	0	0

### show xcode load

The **show xcode load** command shows the current transcoding module (TCM) load both in number of sessions and percent loading. The load percentage depends on the precise mix of codecs, ptimes, and features enabled on the active sessions. The maximum lifetime load is also displayed. Uninstalled TCMs are marked with dashes.

```

ACMEPACKET#show xcode load
17:19:11
Total Sessions:          0
Licensed AMR Sessions:   0
Licensed AMR-WB Sessions: 0
Licensed EVRC Sessions:  0
Licensed EVRCB Sessions: 0

```

TCU	TCM	#Sess	----- Load -----	
			Current	Maximum
0	00	-	-	-
0	01	-	-	-
0	02	-	-	-
0	03	0	0.00%	99.81%
0	04	0	0.00%	99.81%
0	05	-	-	-
0	06	-	-	-
0	07	-	-	-

The TCU column is populated with a 0 for a TCM in the middle slot and a 1 for a TCM in the top slot.

### show xcode session-all

The **show xcode session-all** command displays all of the currently active sessions by their unique session id.

```
ACMEPACKET#show xcode session-all
15:22:51
Requesting xclient sessions table
    Total Active Sessions: 200
    Displaying sessions 1 to 100:
    Session Id: 0x10007
    Session Id: 0x10008
    Session Id: 0x10009
    Session Id: 0x1000a
    Session Id: 0x1000b
```

 **Note:** When there are more than 100 active sessions , the command now displays only active sessions 1 to 100 as opposed to all the active session:

### show xcode session-byid

The session-byid command gives more detailed information about the session. The session-byid command displays the configuration of each channel as well as a number of packet statistics for each channel. This same information can be looked up by IP address and port by using the session-byip command. If only the configuration portion is required, use the session-config command with the session id as the argument. This command is entered as:

```
show xcode session-byid <session_id>
```

For example:

```
ACMEPACKET#show xcode session-byid 0xf006e
##### SESSION 0xf006e #####
Channel 0:
    DSP device           = 14
    Source MAC           = 00:08:25:a0:9a:f3
    Destination MAC      = 00:0e:0c:b7:32:e2
    VLAN ID              = 0
    Egress Interface     = 0
    Src IP:Port          = 172.16.0.235:24448
    Dst IP:Port          = 172.16.0.87:16000
    Src RTPC IP:Port     = 172.16.0.235:24449
    Dst RTPC IP:Port     = 172.16.0.87:16001
    Codec                = G711_ULAW_PCM
    Payload Type         = 0
    Pkt Interval         = 20 msec
    2833 Payload Type    = DISABLED
    Xtone Mode           = XTONE_XTHRU
    Status               = DISABLED
DSP Counters:
    RxInPktCnt          = 474
    RxInByteCnt         = 75840
    RxOutPktCnt         = 749
    RxInSidPktCnt       = 0
    RxNoPktCnt          = 275
    RxBadPktTypeCnt     = 0
    RxBadRtpPayloadTypeCnt = 0
    RxBadPktHdrFormatCnt = 0
    RxBadPktLengthCnt   = 0
    RxMisorderedPktCnt  = 0
    RxBadPktChecksumCnt = 0
    RxUnderrunSlipCnt   = 0
    RxOverrunSlipCnt    = 0
    RxLastVocoderType   = 0
    RxVocoderChangeCnt  = 0
    RxMaxDetectedPdv    = 168
    RxDecdrRate         = 15
    RxJitter:
        CurrentDelay     = 160
        EstimatedDelay    = 0
        ClkDriftingDelta = 0
```

```

    ClkDriftingCorrectionCnt 0
    InitializationCnt 1
    RxCircularBufferWriteErrCnt 0
    RxApiEventCnt 0
    TxCurrentVocoderType 0
    TxInPktCnt 749
    TxOutPktCnt 750
    TxOutByteCnt 120000
    TxInBadPktPayloadCnt 0
    TxTimestampGapCnt 0
    TxTdmWriteErrCnt 0
    RxToneDetectedCnt 0
    RxToneRelayEventPktCnt 0
    RxToneRelayUnsupportedCnt 0
    TxToneRelayEventPktCnt 0
    TxApiEventCnt 0
    TxNoRtpEntryPktDropCnt 0
    ConnectionWaitAckFlag 1
    RxMipsProtectionDropCnt 0
    TxMipsProtectionDropCnt 0
Channel 1:
    DSP device = 14
    Source MAC = 00:08:25:a0:9a:f4
    Destination MAC = 00:1b:21:7a:29:b1
    VLAN ID = 0
    Egress Interface = 2
    Src IP:Port = 192.168.0.235:24448
    Dst IP:Port = 192.168.0.87:32000
    Src RTCP IP:Port = 192.168.0.235:24449
    Dst RTCP IP:Port = 192.168.0.87:32001
    Codec = G729_A
    Payload Type = 18
    Pkt Interval = 20 msec
    2833 Payload Type = DISABLED
    Xtone Mode = XTONE_XTHRU
    Status = DISABLED
DSP Counters:
    RxInPktCnt 748
    RxInByteCnt 14960
    RxOutPktCnt 751
    RxInSidPktCnt 0
    RxNoPktCnt 3
    RxBadPktTypeCnt 0
    RxBadRtpPayloadTypeCnt 0
    RxBadPktHdrFormatCnt 0
    RxBadPktLengthCnt 0
    RxMisorderedPktCnt 0
    RxBadPktChecksumCnt 0
    RxUnderrunSlipCnt 0
    RxOverrunSlipCnt 0
    RxLastVocoderType 6
    RxVocoderChangeCnt 0
    RxMaxDetectedPdv 171
    RxDecdrRate 15
    RxJitter:
        CurrentDelay 160
        EstimatedDelay 0
        ClkDriftingDelta 0
        ClkDriftingCorrectionCnt 0
        InitializationCnt 1
    RxCircularBufferWriteErrCnt 0
    RxApiEventCnt 0
    TxCurrentVocoderType 6
    TxInPktCnt 748

```

```
TxOutPktCnt          748
TxOutByteCnt         14960
TxInBadPktPayloadCnt 0
TxTimestampGapCnt    0
TxTdmWriteErrCnt     0
RxToneDetectedCnt    0
RxToneRelayEventPktCnt 0
RxToneRelayUnsupportedCnt 0
TxToneRelayEventPktCnt 0
TxApiEventCnt        0
TxNoRtpEntryPktDropCnt 0
ConnectionWaitAckFlag 0
RxMipsProtectionDropCnt 0
TxMipsProtectionDropCnt 0
```

### show xcode session-byattr

The show xcode session-byattr command lists all sessions matching the specified attribute name. The only supported attribute is "fax", which will display session information only for FAX-transcoded sessions; all other attributes will return an error. For example:

```
ACMEPACKET#show xcode session-byattr fax
17:52:17
1.  [Chan A_SRC] Ip Address: 192.168.16.1 Port: 5220
    [Chan B_SRC] Ip Address: 172.16.0.40 Port: 10230
    Session Id:0x002021d0"
2.  [Chan A_SRC] Ip Address: 192.168.16.1 Port: 3010
    [Chan B_SRC] Ip Address: 172.16.0.40 Port: 10226
    Session Id:0x00301042"
Oct 26 20:53:22.968 Total Matches:2 Total Active Sessions:2
-----
```

### show xcode session-byipp

The show xcode session-byipp command requires an IP address and port. It lists detailed information about the sessions identified by the specified IP address and port number. Information will be provided for all transcoded call legs matching the IP address, including in both the ingress and egress directions. The show xcode session-byipp command is entered as:

```
show xcode session-byipp <ip_addr> <port_num>
```

This command displays the same information as the **session-byid** command. If a wildcard \* is provided for the port number, the command will display sessions with the matching IP address only, regardless of port number.

### show xcode xlist

The show xcode xlist command displays the TCU (0 = middle, 1 = top), TCM number, number of DSPs on each module, the number of active sessions, and the load percentage. It also displays the state such as Active or Boot Failure. Uninstalled TCMs are indicated by a dash.

```
ACMEPACKET#show xcode xlist
18:22:32
TCU  TCM  DSPs  #Sess  Load  State
===  ===  =====  =====  =====  =====
0    00    2      -      -      -
0    01    2      -      -      -
0    02    2      -      -      -
0    03    2      1      0%    2 Active
0    04    2      1      0%    2 Active
0    05    2      -      -      -
0    06    2      -      -      -
0    07    2      -      -      -
[... ]
1    00    2      1      0%    2 Active
1    01    2      0      0%    2 Active
```



1	02	2	0	0%	2	Active
1	03	2	0	0%	2	Active
1	04	2	0	0%	2	Active
1	05	2	0	0%	2	Active
1	06	2	0	0%	2	Active
1	07	2	0	0%	2	Active
1	08	2	0	0%	2	Active

## Logs

A log file named `log.xserv` can be used for debugging the transcoding feature. This log records the API between the host software and the DSPs and any errors that are encountered.

## Alarms

The transcoding feature employs several hardware and software alarms to alert the user when the system is not functioning properly or overload conditions are reached.

Name/ID	Severity/ Health Degredation	Cause(s)	Traps Generated
No DSPs Present with Transcoding Feature Card (DSP_NONE_PRESENT)	Minor/0	A transcoding feature card is installed but no DSP modules are discovered.	apSysMgmtHardwareErrorTrap
DSP Boot Failure (DSP_BOOT_FAILURE)	Critical/0	A DSP device fails to boot properly at system initialization. This alarm is not health affecting for a single DSP boot failure. DSPs that fail to boot will remain uninitialized and will be avoided for transcoding.	apSysMgmtHardwareErrorTrap
DSP Communications Timeout (DSP_COMMS_TIMEOUT)	Critical/100	A DSP fails to respond after 2 seconds with 3 retry messages. This alarm is critical and is health affecting.	apSysMgmtHardwareErrorTrap
DSP Alerts (DSP_CORE_HALT)	Critical/100	A problem with the health of the DSP such as a halted DSP core. The software will attempt to reset the DSP and gather diagnostic information about the crash. This information will be saved in the /code directory to be retrieved by the user.	apSysMgmtHardwareErrorTrap
DSP Temperature (DSP_TEMPERATURE_HIGH)	Clear 85°C Warning 86°C / 5	A DSP device exceeds the temperature threshold. If the temperature exceeds 90°C, a minor alarm will be set. If it exceeds 95°C, a major alarm	apSysMgmtHardwareErrorTrap

## Transcoding

Name/ID	Severity/ Health Degredation	Cause(s)	Traps Generated
	Minor 90°C / 25 Major 95°C/ 50 Critical 100°C/ 100	will be set. If it exceeds 100°C, a critical alarm will be set. The alarm is cleared if the temperature falls below 85°C. The alarm is health affecting.	
Transcoding Capacity Threshold Alarm (XCODE_UTIL_OVE R_THRESHOLD) / 131329	Clear 80% Warning 95%	A warning alarm will be raised when the transcoding capacity exceeds a high threshold of 95%. The alarm will be cleared after the capacity falls below a low threshold of 80%. This alarm warns the user that transcoding resources are nearly depleted. This alarm is not health affecting.	apSysMgmtGroupTrap
Licensed AMR Transcoding Capacity Threshold Alarm/ 131330	Clear 80% Warning 95%	A warning alarm is triggered if the AMR transcoding capacity exceeds a high threshold of 95% of licensed session in use. The alarm clears after the capacity falls below a low threshold of 80%. This alarm is not health affecting.	apSysMgmtGroupTrap
Licensed AMR-WB Transcoding Capacity Threshold Alarm/ 131331	Clear 80% Warning 95%	A warning alarm is triggered if the AMR-WB transcoding capacity exceeds a high threshold of 95% of licensed session in use. The alarm clears after the capacity falls below a low threshold of 80%. This alarm is not health affecting.	apSysMgmtGroupTrap
Licensed EVRC Transcoding Capacity Threshold Alarm/ 131332	Clear 80% Warning 95%	A warning alarm is triggered if the EVRC transcoding capacity exceeds a high threshold of 95% of licensed session in use. The alarm clears after the capacity falls below a low threshold of 80%. This alarm is not health affecting.	apSysMgmtGroupTrap
Licensed EVRCB Transcoding Capacity	Clear 80%	A warning alarm is triggered if the EVRCB transcoding	apSysMgmtGroupTrap

Name/ID	Severity/ Health Degredation	Cause(s)	Traps Generated
Threshold Alarm/ 131333	Warning 95%	capacity exceeds a high threshold of 95% of licensed session in use. The alarm clears after the capacity falls below a low threshold of 80%. This alarm is not health affecting.	

## Transcoding Capacity Traps

The Oracle Communications Session Border Controller sends the apSysMgmtGroupTrap as transcoding capacity nears its limit. This trap is sent and cleared for three conditions:

- Total DSP usage exceeds 95%
- Total AMR sessions exceed 95% of licensed capacity
- Total AMR-WB sessions exceed 95% of licensed capacity
- Total EVRC sessions exceed 95% of licensed capacity
- Total EVRCB sessions exceed 95% of licensed capacity

The apSysMgmtGroupTrap contains the condition observed (apSysMgmtTrapType) and the corresponding value reached (apSysMgmtTrapValue).

```
apSysMgmtGroupTrap          NOTIFICATION-TYPE
OBJECTS                      { apSysMgmtTrapType, apSysMgmtTrapValue }
STATUS                        current
DESCRIPTION
    " The trap will generated if value of the monitoring object
    exceeds a certain threshold. "
 ::= { apSystemManagementNotifications 1 }
```

When the resource usage retreats below a defined threshold, the Oracle Communications Session Border Controller sends an apSysMgmtGroupClearTrap.

```
apSysMgmtGroupClearTrap     NOTIFICATION-TYPE
OBJECTS                      { apSysMgmtTrapType }
STATUS                        current
DESCRIPTION
    " The trap will generated if value of the monitoring object
    returns to within a certain threshold. This signifies that
    an alarm caused by that monitoring object has been cleared. "
 ::= { apSystemManagementNotifications 2 }
```

The following table summarizes trigger and clear conditions for transcoding capacity alerts as sent in the the apSysMgmtGroupTrap:

Monitored Transcoding Resource	SNMP Object & OID in apSysMgmtTrapType	Trap Sent	Clear Trap Sent
Total DSP Usage	apSysXCodeCapacity 1.3.6.1.4.1.9148.3.2.1.1.34	95%	80%
AMR License Capacity Usage	apSysXCodeAMRCapacity 1.3.6.1.4.1.9148.3.2.1.1.35	95%	80%

## Transcoding

Monitored Transcoding Resource	SNMP Object & OID in apSysMgmtTrapType	Trap Sent	Clear Trap Sent
AMR-WB License Capacity Usage	apSysXCodeAMRWBCapacity 1.3.6.1.4.1.9148.3.2.1.1.36	95%	80%
EVRC License Capacity Usage	apSysXCodeEVRCCapacity 1.3.6.1.4.1.9148.3.2.1.1.39	95%	80%
EVRCB License Capacity Usage	apSysXCodeEVRCBCapacity 1.3.6.1.4.1.9148.3.2.1.1.40	95%	80%

The following SNMP Objects are inserted into the apSysMgmtTrapType when sending and clearing a transcoding capacity trap. You may query them individually with an SNMP GET.

- apSysXCodeCapacity (1.3.6.1.4.1.9148.3.2.1.1.34)
- apSysXCodeAMRCapacity (1.3.6.1.4.1.9148.3.2.1.1.35)
- apSysXCodeAMRWBCapacity (1.3.6.1.4.1.9148.3.2.1.1.36)
- apSysXCodeEVRCCapacity (1.3.6.1.4.1.9148.3.2.1.1.39)
- apSysXCodeEVRCBCapacity(1.3.6.1.4.1.9148.3.2.1.1.40)

## SNMP

The following sections have been removed from this document. Their content has been reformatted and appears in the MIB Reference guide in the *Codec and Transcoding MIB (ap-codec.mib)* section:

- SNMP Retrieval of Transcoding Statistics
- Acme Packet Codec and Transcoding MIB (ap-codec.mib)
- Acme Packet System Management MIB (ap-smgmt.mib)

## Generating RTCP

The Oracle Communications Session Border Controller is capable of creating and sending RTCP reports using Transcoding resources. This produces RFC 3550 compliant RTCP report information on media traffic for active sessions. The system calculates these statistics using measurements on traffic between a target end station and itself. With respect to a given media session, the system does not produce end-to-end reports. In addition, the system can drop RTCP reports generated upstream so target stations don't receive RTCP reports that are redundant to its own.

RTCP reporting generated at the Oracle Communications Session Border Controller provides the following benefits:

- Provide RTCP reporting to core applications that may otherwise not be receiving this data.
- Provide RTCP reporting to access elements, including mobile client applications, that may otherwise not be receiving this data.
- Generate valid RTCP data for transcoded calls.
- Generate RTCP data from the Oracle Communications Session Border Controller's perspective.
- Applicable RTCP reports continue to be sent when a call is muted or placed on-hold.

The user enables and specifies the type of calls on which the system generates RTCP reports via configuration. Applicable configuration includes creating **rtcp-policy** objects and applying them to realms. These objects specify whether to issue reports for transcoded or all calls traversing the realm. There is also a configuration to disable the policy.

## How it Works

The Oracle Communications Session Border Controller generates RTCP by sending RTCP sender report information to the media termination point within 5 seconds after an RTP session starts, per RFC 3550. All RTCP messaging includes Sender Reports and, if the end station is receiving media, statistics corresponding to the received media. This messaging persists, within 5-second windows for the duration of the session. The system sends a goodbye message no more than 20 ms after the call is complete. The network administrator should note this timing and ensure that network NAT configuration does not block these final messages. Depending on call type and configuration, the system listens for and drops RTCP reports generated upstream. RTCP reports generated prior to transcoding provide unreliable information.

All reports include the source description with a CNAME string. The CNAME string is the IP address and port number used for the corresponding media stream's RTCP in the format:

```
<UDP Local Port Num for RTCP>@<IP address of the applicable steering pool>
```

The system sources this data from the applicable steering pool configuration.


## Functional Matrix - Configuration vs. Call Type

RTCP generation and any corresponding RTCP blocking is a function of the type of call (transcoded or non-transcoded), and configuration. Applicable **rtcp-policy** configuration includes specifying whether the system should generate RTCP for all calls, or for transcoded calls only.

The behavior is as follows:

- For Non-Transcoded Calls—Based on the RTCP-generation configuration:
  - **xcoded-calls-only**—No RTCP generated; existing RTCP passes
  - **all-calls**—RTCP generated; existing RTCP blocked
- For Transcoded Calls—RTCP generated; existing RTCP blocked for both configurations
- The RTCP generation setting includes a disabled setting (**none**) that is set by default. This specifies that the policy never generate RTCP. Realms configured with these disabled policy settings cause the system to pass existing RTCP for non-transcoded calls and block existing RTCP for transcoded calls.

When RTCP generation is enabled for non-transcoded calls using the **all-calls** setting, these calls must negotiate a codec that the Oracle Communications Session Border Controller can transcode. This ensures that the call utilizes the system's transcoding resources, which can then generate the RTCP.

 **Note:** A realm's **rtcp-policy** takes precedence over its **block-rtcp** setting. Specifically, if **block-rtcp** is disabled and the **rtcp-policy** is set to block, the system blocks existing RTCP.

## RTCP Generation Platform Support

RTCP generation is supported on AP4500 and AP6300 platforms when they have a transcoding module populated with DSPs.

When generating RTCP for a non-transcoded call using DSPs, the system processes the call as if it is a transcoded call, but uses a "null" transcoding methodology that sets the input codec equal to the output codec.

There are important alarms that help the user verify the status of this hardware. Refer to the Transcoding Troubleshooting and Maintenance section in this guide for more information about transcoding hardware status. Refer to the *Maintenance and Troubleshooting Guide* for information about interface hardware status.

There is no specific license required to generate RTCP reports.

Users can determine that they have a hardware configuration that supports RTCP generation as follows using the **show xcode xlist** command to verify the presence of transcoding hardware with DSP modules.

## Configuring RTCP Generation

The procedure below provides the steps needed to configure RTCP generation on the Oracle Communications Session Border Controller.

## Transcoding

Before proceeding, make sure you know whether your platform supports transcoding. As described in the section on RTCP generation, this can affect your configuration and operational results.

To have your Oracle Communications Session Border Controller generate RTCP traffic statistics reporting:

1. In Superuser mode, use the following command sequence to access the **rtcp-policy** element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# rtcp-policy
```

From this point, you can configure a new RTCP policy or select an existing policy for editing.

2. **name**—Name your policy for use when applying it to a **realm-config**.

```
ACMEPACKET(rtcp-policy)# name report-allcalls
```

3. **rtcp-generate**—Enter the desired setting to generate RTCP. The effect of these settings is dependent on the platform and the presence of specific hardware on that platform. See the section on Generating RTCP in the ACLI Configuration Guide for these details.

- **none**—Disables this policy.
- **all-xcoded-calls**—The system generates RTCP report information only for the transcoded calls that pass through the realm.
- **all-calls**—The system generates RTCP report information for all calls that pass through the realm.

```
ACMEPACKET(rtcp-policy)# rtcp-generate all-calls
```

4. Type **done** and **exit** to retain your configuration.
5. Access the realm to which you want to apply this **rtcp-policy**.

```
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)# select
```

6. Apply the **rtcp-policy** to your realm.

```
ACMEPACKET(realm-config)# rtcp-policy report-allcalls
```

7. Type **done** and exit configuration mode. Save and activate your configuration.

## Obtaining System Information about RTCP Generation

The Oracle Communications Session Border Controller provides information about RTCP report generation within log files.

The **log.mbcd** file, when configured to capture at the DEBUG level, includes system messages related to invoking and firing **rtcp-policy** rules.

An example of applicable log information is provided below.

```
[XC] NatFlow::check_transcoding()
[XC] SessionSetRtcpPorts for A: src=10001 dest=20001
[XC] NatFlow::check_transcoding() SessionSetRtcpOptions(A):
[XC] RtcpPolicy "my_rtcp_policy"; rtcp-generate set to:
RTCP_GEN_MODE_ALL_CALLS
[XC] Will generate RTCP with CNAME=192.160.1.100:10001
[XC] NatFlow::check_transcoding()
[XC] SessionSetRtcpPorts for B: src=10001 dest=20001
[XC] NatFlow::check_transcoding() SessionSetRtcpOptions(B):
[XC] RtcpPolicy "my_rtcp_policy"; rtcp-generate set to:
RTCP_GEN_MODE_ALL_CALLS
[XC] Will generate RTCP with CNAME=172.16.1.100:10101
```

Error and non-error system messages related to RTCP generation can be found in **log.sipd**, **log.mbcd**, and **log.xserv**.

## AP6300 NIU Hotswap Guidelines

The AP6300 Session Border Controller provides a 3-slot chassis. Each chassis is monitored by a dedicated Hot-Swap sensor that maintains state information for the resident Network Interface Units.

The Oracle Communication 6300 Session Border Controller (SBC) is the first multi-slot SBC that runs C series software. A hotswap sensor, located on each of the 6300's three slots, provides the ability to track the state of individual network interface units (NIUs) in the 6300 chassis.

There are 8 possible associated states as shown below.

PRESENT/NOT PRESENT	Checks the physical presence of a NIU inserted in a 6300 slot
LATCH OPEN/LATCH CLOSED	Checks the physical latch in the back of a NIU
POWER GOOD/ POWER BAD	Checks the power conditions of an NIU
READY/NOT READY	Checks all of the three previous state

A fully functional NIU is always in the following state: present, latch closed, power good, and ready.

The status of each NIU can be check by issuing the command **show platform hotswap** from the ACLI.

### Network Interface Unit Removal/Replacement -- Standalone Node

A user or field engineer can remove a Network Interface Unit (NIU) from the 6300 chassis. When an NIU is removed, all services associated with the NIU (for example, media transmission, signaling, transcoding, and so on) will be interrupted and the system will be out-of-service (OOS).

Use the following procedure to remove an NIU.

1. Working from the back of the chassis, carefully open the NIU's latch. The hotswap LED placed directly above the NIU will transition from green (present/latch close/ready) to blinking blue. After 10 seconds it will stop blinking and remain blue (present/latch open/not ready).
2. You can now safely remove your NIU.

Removing the NIU generates a critical alarm notifying the user that the NIU has been removed. Additionally, the Vacuum Fluorescent Display (VFD) displays a hardware alarm and starts blinking. From the ACLI, issue a "reboot force" command.

To see the alarm use: "display-alarms" from the ACLI.

To replace an NIU, insert an NIU of a different or the same type as the one used before.

The alarm manager clears the critical alarm, and issues new minor alarm to alert the user that an NIU was inserted after the boot sequence. At this point the system is still OOS.

After rebooting the system, service will be restored, and the new NIU will be functional

### NIU Removal/Replacement -- High Availability Deployment

You cannot remove an NIU from the chassis of a 6300 SBC functioning in the active mode. If an NIU is accidentally removed, (1) all services will be interrupted; (2) a critical alarm will be triggered to notify the user that the NIU has been removed; (3) the system will relinquish its active state; and (4) proceed to reboot itself.

Under this scenario, service will not be interrupted because the standby node will assume the active role

In a 6300 functioning in standby mode, the user or field engineer can remove an NIU from the chassis. All services associated with the NIU (for example, media transmission, signaling, transcoding, and so on) will be interrupted and the standby node will be OOS. Removal of an NIU from the standby, however, will not affect the active node which remains in service.

Use the following procedure to remove an NIU from the standby node.

## Transcoding

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1. Working from the back of the chassis, carefully open the NIU's latch. The hotswap LED placed directly above the NIU will transition from green (present/latch close/ready) to blinking blue. After 10 seconds it will stop blinking and remain blue (present/latch open/not ready)
2. You can now safely remove the NIU. To replace an NIU, insert an NIU of the same type as the one used before.

Removing the NIU generates a critical alarm notifying the user that the NIU has been removed. Additionally, the VFD displays a hardware alarm and starts blinking

To see the alarm use: "display-alarms" from the ACLI.

The alarm manager clears the critical alarm, and issues a new minor alarm to alert the user that an NIU was inserted after the boot sequence. At this point the system is still OOS.

From the ACLI, issue a "reboot force" command.

After rebooting the standby, it will revert to standby mode.



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## DTMF Transfer and Support

### DTMF Interworking

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Multimedia devices and applications can exchange user-input DTMF information end-to-end over IP networks. The Oracle Communications Session Border Controller provides the capabilities required to interconnect networks and devices that use different DTMF indication signaling protocols.

### DTMF Indication

There are three ways to convey DTMF information for packet-based communications:

- DTMF audio tones: DTMF digit waveforms are encoded inline with voice packets. This method only works with uncompressed audio codecs like G.711. Compressed audio codecs like G.729 and G.723 are incompatible with DTMF audio. DTMF audio is also referred to as in-band tones.
- Out-of-band signaling events:
  - SIP INFO messages with Content-Type: application/dtmf-relay define out-of-band signaling events for transmitting DTMF information. SIP INFO messages separate DTMF digits from the voice stream and send them in their own signaling message.
- RTP named telephony events (NTE): RFC 2833 telephone-events are a standard that describes how to transport DTMF tones in RTP packets according to section 3 of RFC 2833. Of the three RTP payload formats available, the Oracle Communications Session Border Controller supports RTP NTE.

### RFC 2833 telephone-event

RFC 2833 specifies a way of encoding DTMF-indications in RTP media streams. It does not encode the audio of the tone itself, instead data represents the sent tone. RFC 2833 can be used with SIP.

RFC 2833 defines the format of NTE RTP packets used to transport DTMF digits and other telephony events between two peer endpoints. With the NTE method, the endpoints perform per-call negotiation of the DTMF transfer method. They also negotiate to determine the payload type value for the NTE RTP packets.

The NTE payload takes the place of codec data in a standard RTP packet. The payload type number field of the RTP packet header identifies the contents as 2833 NTE. The payload type number is negotiated per call. The local device sends the payload type number to use for RFC 2833 packets using SDP, which tells the other side what payload type number to use when sending the named event packets to the local device. Most devices use payload type number 101 for RFC 2833 packets, although no default is specified in the standard.

The RFC 2833 packet's RTP header also makes use of the timestamp field. Because events often last longer than the RFC 2833 packets sending interval, the timestamp of the first 2833 packet for an event represents the beginning

## DTMF Transfer and Support

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reference time for subsequent RFC 2833 packets for that same event. For events that span multiple RTP packets, the RTP timestamp identifies the beginning of the event. As a result, several RTP packets might carry the same timestamp.

### SIP INFO Messages

SIP INFO messages can send indications of DTMF audio tones between peers as part of the signaling path of the call. Upon receipt of a SIP INFO message with DTMF content, the gateway generates the specified DTMF tone on the receiving end of the call.

## DTMF Transfer Processing Overview

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Enabling 2 UAs to communicate different DTMF indications with each other is facilitated in two steps. First, the Oracle Communications Session Border Controller takes an active role in the SDP negotiation between two UAs, this is the capability negotiation step. You can configure your system to suggest the DTMF indication methods that each endpoint can and can not use.

The second step is translation evaluation. After the SDP negotiation is complete, based on system configuration and the media support on each call leg, the Oracle Communications Session Border Controller performs DTMF indication conversion or passive forwarding of DTMF indication messages between UAs.

## Capability Negotiation

---


SDP capability negotiation is the first phase of enabling DTMF transfer. The completion of the SDP offer/answer exchange yields a set of supported codecs between each UA and the Oracle Communications Session Border Controller .

For DTMF transfer consideration, SDP manipulation is directed by parameters set in one of three configuration elements:

- codec policy
- signaling interface's RFC 2833 mode
- session agent's RFC 2833 mode

The Oracle Communications Session Border Controller performs SDP manipulation (addition, removal, or modification of supported codecs) toward the called party first by any applicable codec policies. If one or more of the actions which define telephone-event Modification by Codec Policy occurs, then SDP manipulation is only performed by the codec policy configuration, not by RFC 2833 mode parameters in the signaling interface or session agent.

If a codec policy attached to either the ingress or egress realm triggers SDP manipulation, then the other realm uses codec policy for any telephone-event SDP manipulation; none of the RFC 2833 Mode configurations are used for SDP manipulation.

 **Note:** If the call does not trigger the evaluation of any codec policies, all DTMF transfer processing is only subject to RFC 2833 Mode rules.

If none of the telephone-event Modification by Codec occur, then the Oracle Communications Session Border Controller performs SDP manipulation according to the RFC 2833 Mode parameters in the signaling interface or session agent.

## SDP Manipulated by Codec Policy

When a call is received, any applicable codec policies are applied and evaluated. If telephone-event SDP is modified by codec policies, then SDP manipulation by RFC 2833 Mode is not performed for either side of the call.

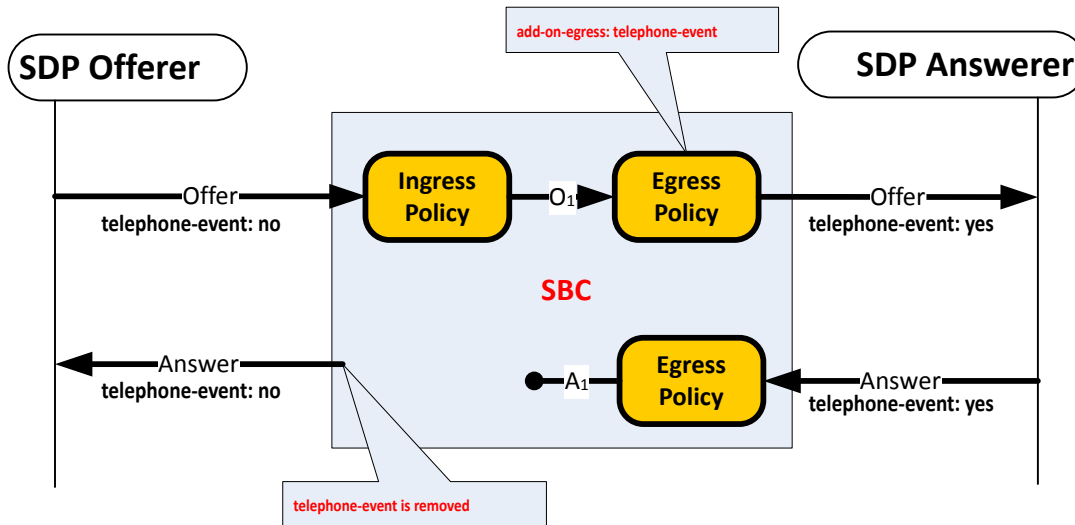
### telephone-event Modification by Codec Policy

For qualifying if telephone-event modification in SDP was performed by codec policy, one of the following events had to have happened:

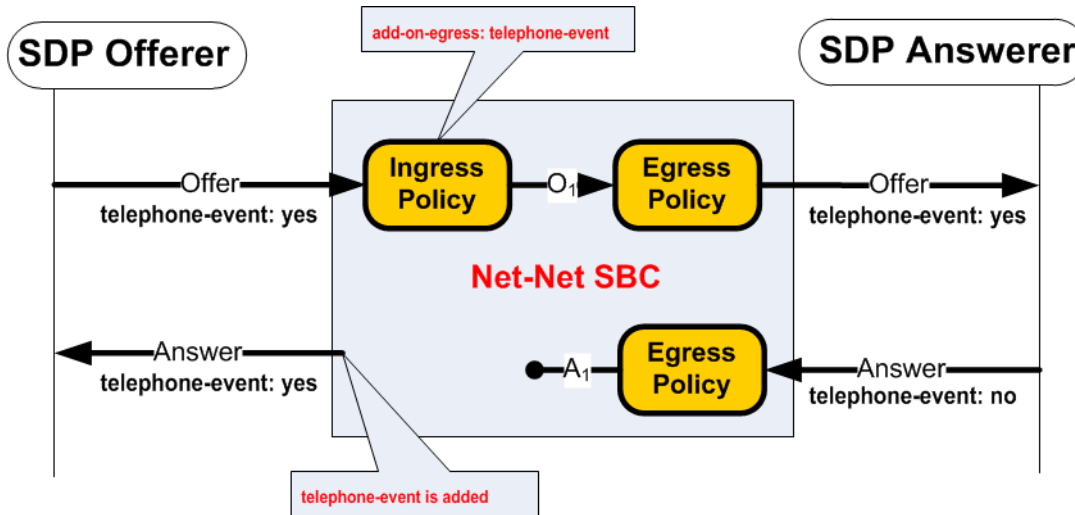
- Codec policy explicitly deleted telephone-event by configuring `allow telephone-event: no`
- Codec policy explicitly added telephone-event by configuring `add-on-egress telephone-event`
- Codec policy implicitly denied telephone-event by allowing one or more codecs but not adding telephone-event to the allow list
- Codec policy has `audio: no` configured in the allow list

The following three cases highlight how codec policies can manipulate telephone-event SDP. Once any of these cases occurs, SDP manipulation by RFC 2833 Mode parameter will not be performed.

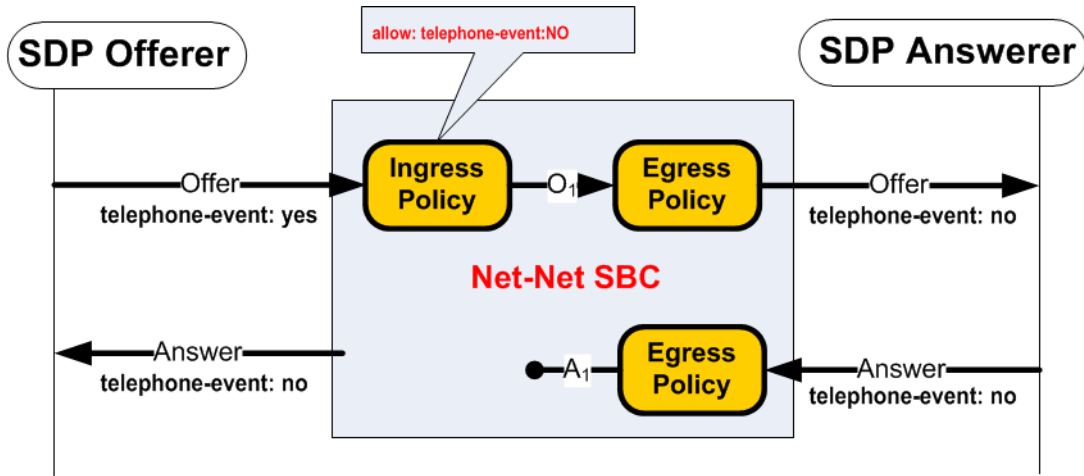
1. telephone-event added to SDP: The codec policy adds telephone-event to the SDP sent to the egress realm. The UA supports telephone-event.



2. telephone-event maintained in SDP: The codec policy maintains the offered telephone-event in the SDP sent into the egress realm. Although telephone-event was not answered in the egress realm, telephone-event is added back to the offerer-side SDP because of the add-on-egress setting in the ingress realm.



3. telephone-event removed from SDP: codec policy removes telephone-event from the initial SDP offer. telephone-event is not forwarded to the Answerer, and subsequently not returned from the answerer, or forwarded again to the offerer.



### SDP Manipulated by RFC 2833 Mode

If none of the telephone-event Modification by Codec Policy events occur, as previously explained, SDP may still be modified by RFC 2833 Mode parameter if preferred or dual mode is configured.

The RFC 2833 mode parameter functions similarly to the add-on-egress parameter; it suggests telephone-event support, by adding it in SDP if not already there. This parameter must be set to preferred or dual to add telephone-event to SDP.

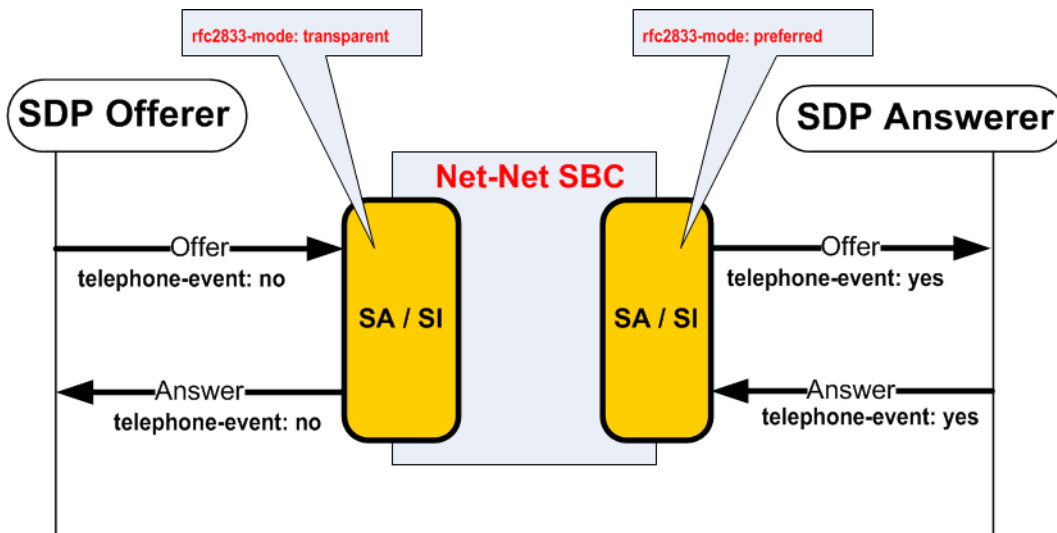
### Transparent RFC 2833 Support

Setting a signaling interface or session agent's RFC 2833 mode to transparent disables the addition of RFC 2833 telephone-event to SDP upon egress. The Oracle Communications Session Border Controller passes the offered SDP capabilities to the next-hop signaling element.

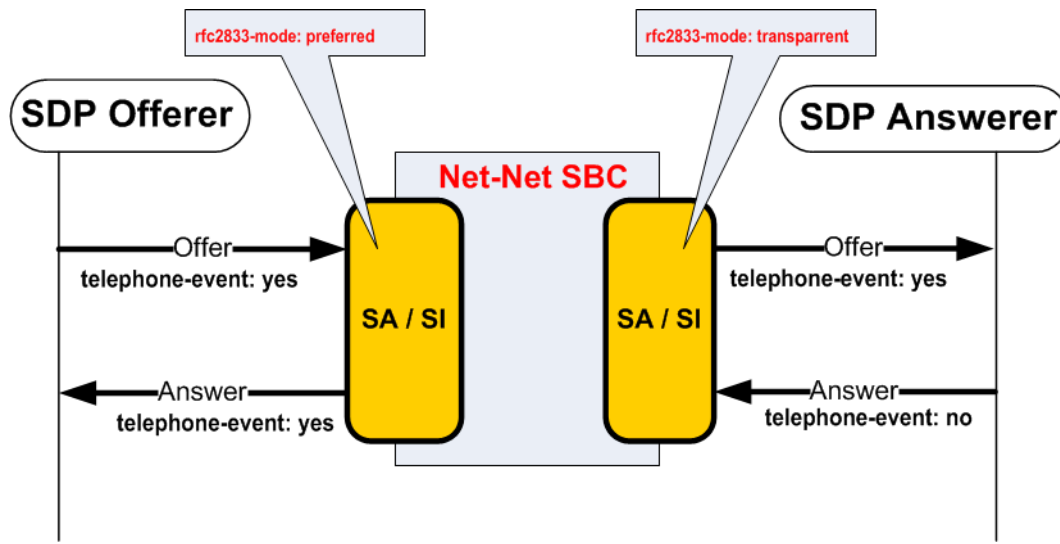
### Preferred RFC 2883 Support

Setting a signaling interface or session agent's RFC 2833 mode to preferred indicates that the RFC 2833 telephone-event DTMF transfer method is the preferred method for sending a DTMF indication. In the capability negotiation phase a telephone-event media type will be inserted in the outgoing SDP offer, if it was not present in the original offer.

1. In the following example RFC 2833 mode is set to preferred on the egress side of the call. Because there is no telephone-event in the SDP, and RFC 2833 mode is set to preferred, the Oracle Communications Session Border Controller adds telephone-event to the SDP offer.



- In the following example, RFC 2833 is set to preferred mode on the SDP offer side of the call. The Oracle Communications Session Border Controller maintains the telephone-event support even though telephone-event is not supported on the SDP answerer's side of the call.



## RFC 2833 Payload Type Mapping

The Oracle Communications Session Border Controller does not require that call legs use the same media type for telephone-event. If each call leg uses a different media type value, the Oracle Communications Session Border Controller facilitates payload type mapping to ensure the telephone-event media stream be reliably transported across the call.

- On the SDP offer side, when the Oracle Communications Session Border Controller returns its SDP answer, it uses the same media type that the SDP offerer offered.
- The Oracle Communications Session Border Controller forwards the originally offered telephone-event media type to the SDP answerer. If telephone-event was added by RFC 2833 mode, the Oracle Communications Session Border Controller adds telephone-event with the media type value configured in the RFC 2833 payload parameter. If telephone-event was added by a codec policy, the Oracle Communications Session Border Controller adds telephone-event with the media type value configured in the media profile.
- If the SDP answerer returns a new value for telephone-event, the Oracle Communications Session Border Controller still supports RFC 2833 on that side of the call and uses the media type that the answerer sent.

## Translation Evaluation

After SDP has been negotiated, the Oracle Communications Session Border Controller determines what types of DTMF translation takes place for the call. The Oracle Communications Session Border Controller sequentially evaluates the following rules for each call leg to determine what DTMF indication type it will forward to an endpoint.

- RFC 2833—When the SDP offer/answer exchange resolves to both the Oracle Communications Session Border Controller and the endpoint supporting RFC 2833 on one side of the call, the Oracle Communications Session Border Controller will send DTMF indications in RFC 2833 format.
- DTMF audio tones—Three conditions must be met for the Oracle Communications Session Border Controller to support DTMF audio tones, as transcoded from another DTMF indication form:
  - The applicable codec policy's dtmf in audio parameter is set to preferred
  - The endpoint and Oracle Communications Session Border Controller have negotiated to a DTMFable codec (G711)
  - Transcoding resources are available

## DTMF Transfer and Support



**Note:** Because of rule number one, rule number two can not happen if RFC 2833 is supported in SDP —Only one media-based DTMF transfer method, RFC 2833 or DTMF audio tones may be used on a call leg.

3. If neither RFC 2833 nor DTMF Audio tones are supported on a call leg, as a result of SDP negotiation, then the Oracle Communications Session Border Controller forwards DTMF indication messages to that side in signaling message format (SIP INFO).

In the following images that illustrate DTMF transfer scenarios, a gears icon appears when relevant. This icon indicates that the Oracle Communications Session Border Controller performs DTMF indication processing, creating DTMF audio tones or RFC 2833 telephone-event messages from another form of DTMF indication.

### RFC 2833 Sent by Offerer

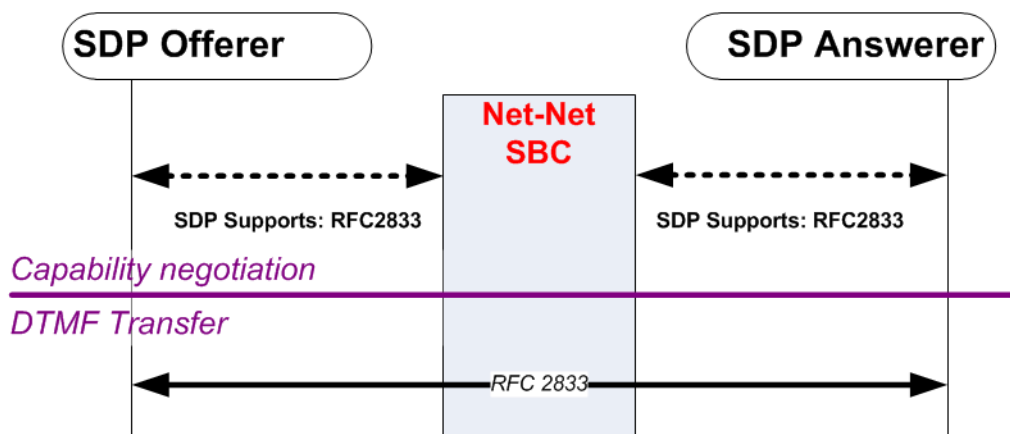
In the following three examples, the SDP offerer sends DTMF indication messages in RFC 2833 format. The SDP answerer can receive DTMF indications in the format identified in each example.

#### RFC 2833 to RFC 2833

When the SDP offer and answer sides of a call both support RFC 2833, the Oracle Communications Session Border Controller forwards RFC 2833 messages between both sides of the call. No processing is used to transform these DTMF-indication messages to another format.



**Note:** When the audio stream is transcoded, DTMF audio is completely removed from the audio stream.



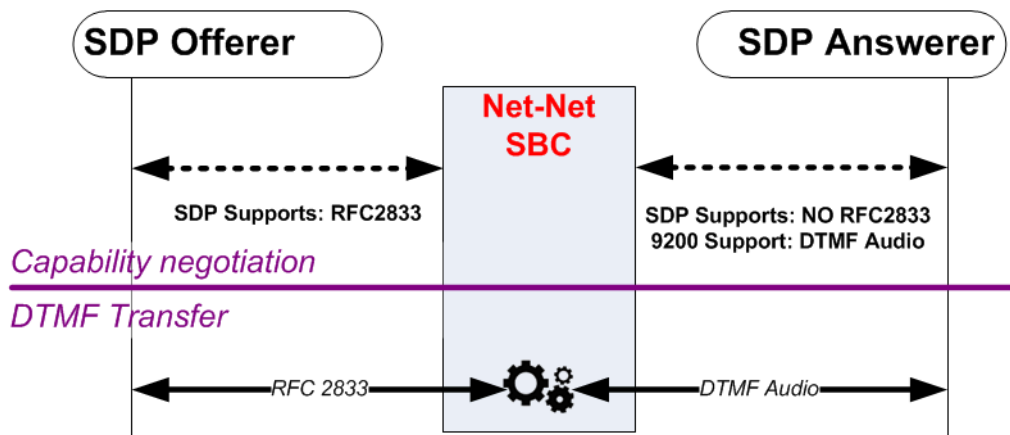
A SIP INFO message received from either the offerer or answerer is forwarded unconverted to the other side of the call.

If there is no audio transcoding enabled for this call, and the egress side is set to dual, a received SIP INFO message will not be converted to both RFC 2833 and SIP INFO messages for sending to the other side of the call.

If DTMF audio tones are received from either the offerer or answer, they are forwarded unconverted to the other side (when the audio portion of the call is not transcoded).

#### RFC 2833 to DTMF Audio Tones

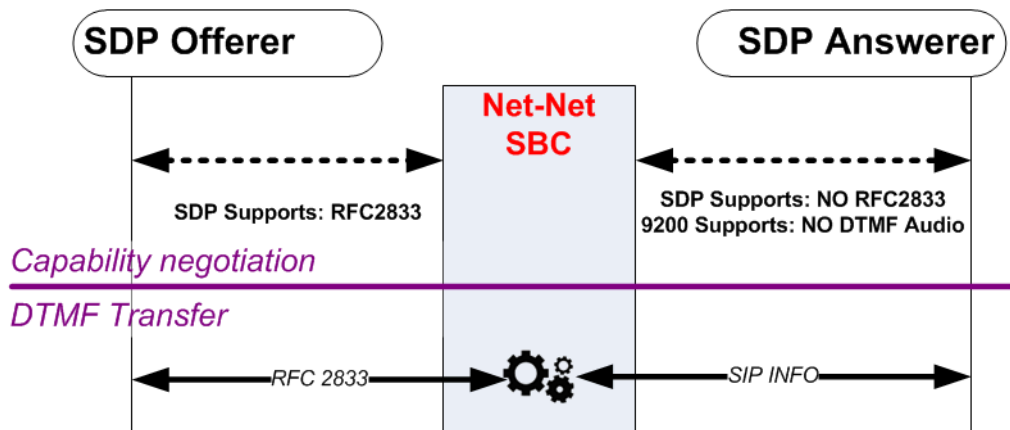
When the SDP offer side supports RFC 2833, and the SDP answer side supports the three DTMF Audio Tone conditions and does not support RFC 2833, the Oracle Communications Session Border Controller converts from RFC 2833 to DTMF audio tones for the call.



A SIP INFO message received by the Oracle Communications Session Border Controller from either the offerer or answerer is converted into the DTMF transfer method that the previous diagram shows for the egress side of the message. In this case, transcoding resources are used.

### RFC 2833 to SIP INFO

When the SDP offer side supports RFC 2833 and the SDP answer side does not support the DTMF conditions nor RFC 2833, the Oracle Communications Session Border Controller converts from RFC 2833 to SIP INFO.



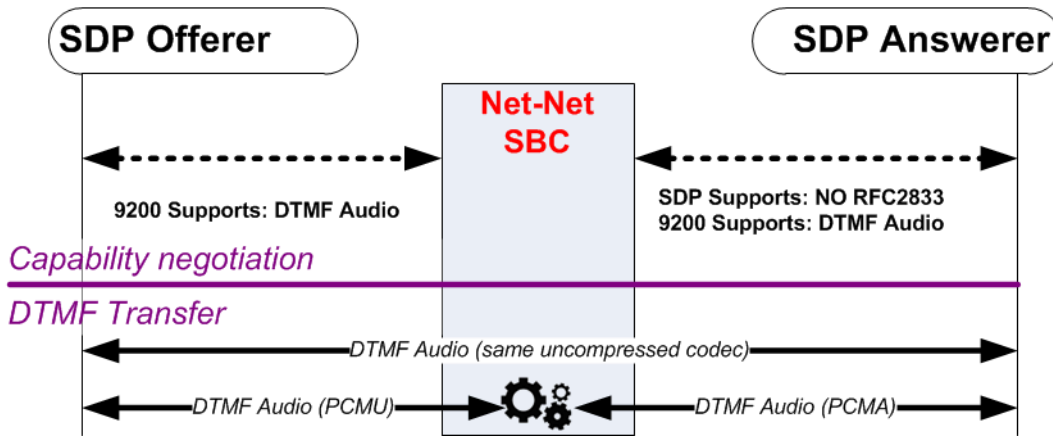
If a SIP message is received from the offerer, it is forwarded unconverted to the answerer.

### DTMF Audio Tones Sent by Offerer

In the following three examples, the SDP offerer sends DTMF indication messages in DTMF audio tones format. The SDP answerer can receive DTMF indications in the format identified in each example.

#### DTMF Audio to DTMF Audio

If the SDP offer and answer sides both support the same type of G711 codec, the audio stream is forwarded between the two sides without processing.

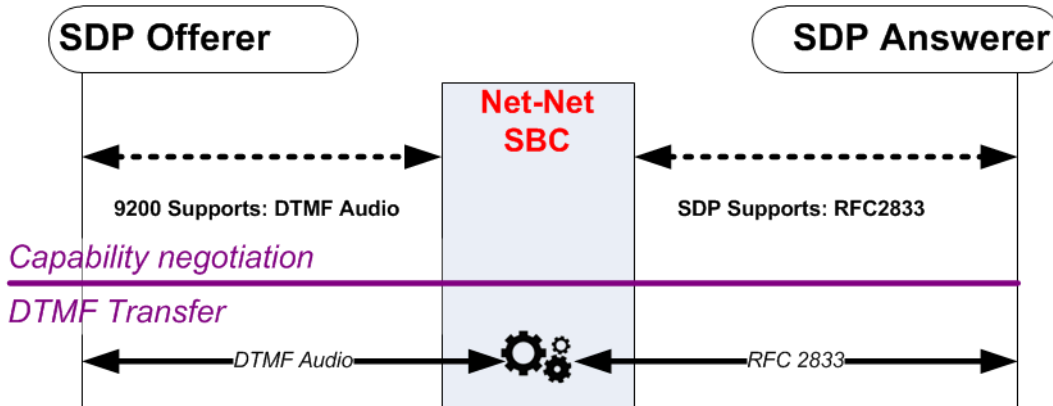


If the two sides of the call support DTMF audio tones, but use different audio codecs, and the SDP answer side supports the three DTMF Audio Tone conditions and does not support RFC 2833 then the Oracle Communications Session Border Controller will preserve DTMF audio tone indication across the call.

Transcoding resources are used only if different audio codecs are used or the Override Preferred DTMF Audio feature is enabled.

**DTMF Audio to RFC 2833**

When the SDP offer side supports DTMF audio tones, and the SDP answer side supports RFC 2833, and transcoding resources are available, and does NOT support either or both of the first two DTMF Audio tone conditions, then the Oracle Communications Session Border Controller will convert incoming DTMF audio tones to outgoing RFC 2833 packets.

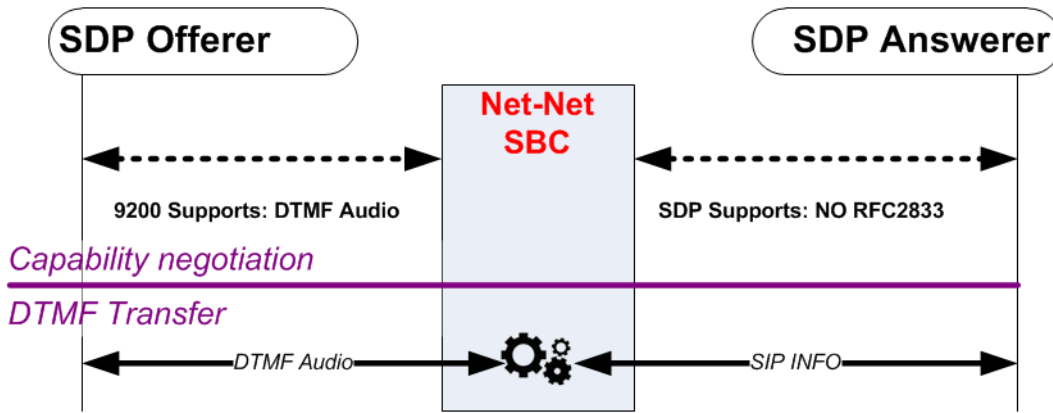


Transcoding resources are always required in this scenario.

**DTMF Audio to SIP**

When the SDP offer side supports DTMF audio tones, and the SDP answer side does not support RFC 2833, and does not support the three DTMF Audio Tone conditions, then the Oracle Communications Session Border Controller converts incoming DTMF audio tones to SIP INFOmessages.





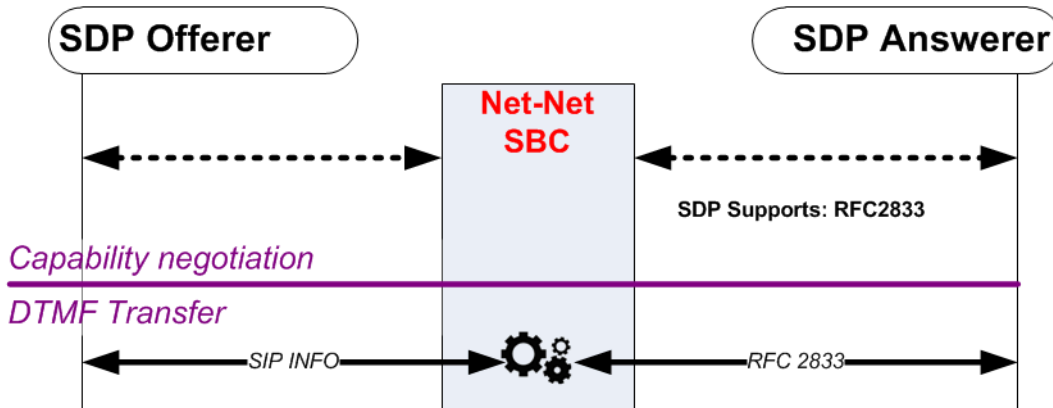
Transcoding resources are always required in this scenario.

### SIP INFO Sent By Offerer

In the following three examples, the SDP offerer sends DTMF indication messages in SIP INFO message format. The SDP answerer can receive DTMF indications in the format identified in each example.

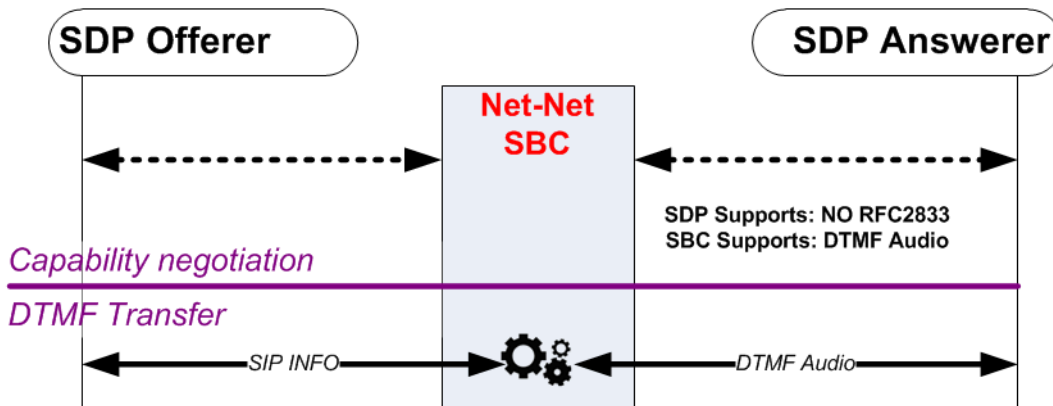
#### SIP INFO to RFC 2833

When the SDP offer side sends a SIP INFO message, and the SDP answer side supports RFC 2833, then the Oracle Communications Session Border Controller will convert incoming SIP INFO messages to outgoing RFC 2833 packets.



#### SIP INFO to DTMF Audio

SIP INFO will only be converted to DTMF audio tones only if RFC 2833 is not supported, dtmf-in-audio is enabled, the answer side supports a G711 codec, and transcoding resources are available.

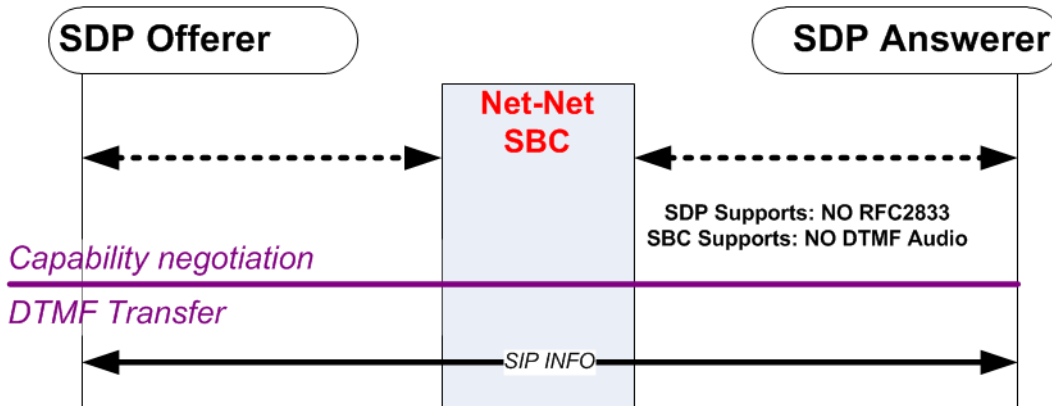


## DTMF Transfer and Support

Transcoding resources are always required in this scenario.

### SIP INFO to SIP INFO

When the SDP offer side sends a SIP INFO message and the SDP answer side does not support RFC 2833 and does not support the three DTMF audio tone conditions, the SIP INFO message will always be forwarded as the like SIP INFO message.




## Dual Mode

Dual mode is used to send both RFC 2833 and the protocol-specific DTMF indication to a UA when possible: SIP INFO from a SIP interface.

To consider dual mode scenarios, the Oracle Communications Session Border Controller sets up the call SDP. At the conclusion of the capability negotiation, the Oracle Communications Session Border Controller is configured to support DTMF Audio tones or RFC 2833 independently for each side of the call.

When the call leg supports RFC 2833 as the means of DTMF transfer, the Oracle Communications Session Border Controller checks if the SIP interface's (or session agent's) RFC 2833-mode parameter is configured to dual. If it is, the Oracle Communications Session Border Controller sends both RFC 2833 and SIP INFO messages to the UA on this side of the call.

 **Note:** Whether RFC 2833 support was initiated between the Oracle Communications Session Border Controller and the UA by a codec policy or by the `rfc2833-mode` parameter, the Oracle Communications Session Border Controller looks to the `rfc-2833` parameter to consider if dual mode is supported.

When the call leg supports DTMF audio tones as the means of DTMF transfer, the Oracle Communications Session Border Controller checks if the codec policy's `dtmf-in-audio` parameter is configured to dual. If it is, the Oracle Communications Session Border Controller sends both DTMF audio tones and SIP INFO messages to the UA on this side of the call.

## P-Dual-Info Header

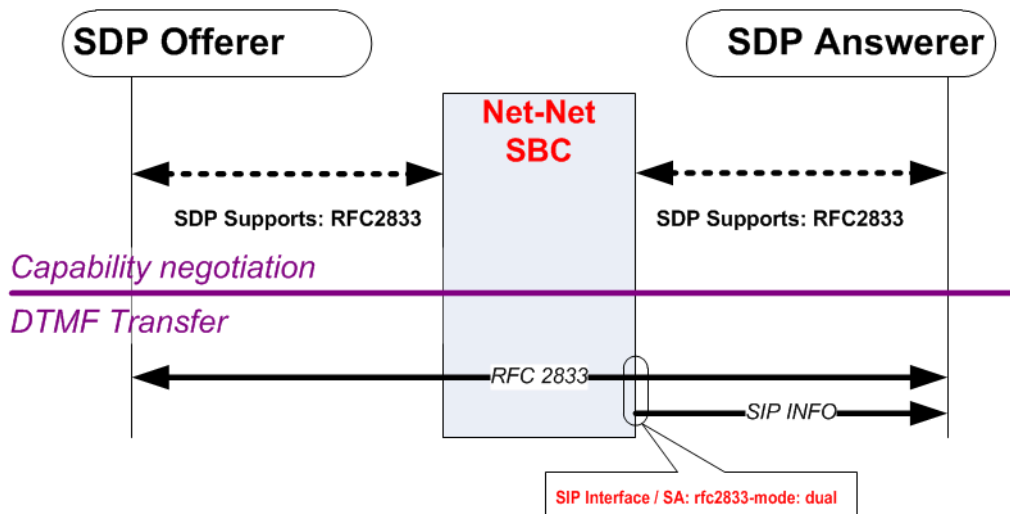
When the Oracle Communications Session Border Controller forward both media DTMF indication and signaling based DTMF indication for the same received DTMF indication, a P-Dual-Info header is added to the forwarded signaling message. You can configure the appearance of the header with the `dual-info` option. The default header appearance is:

```
P-dual-info: true
```

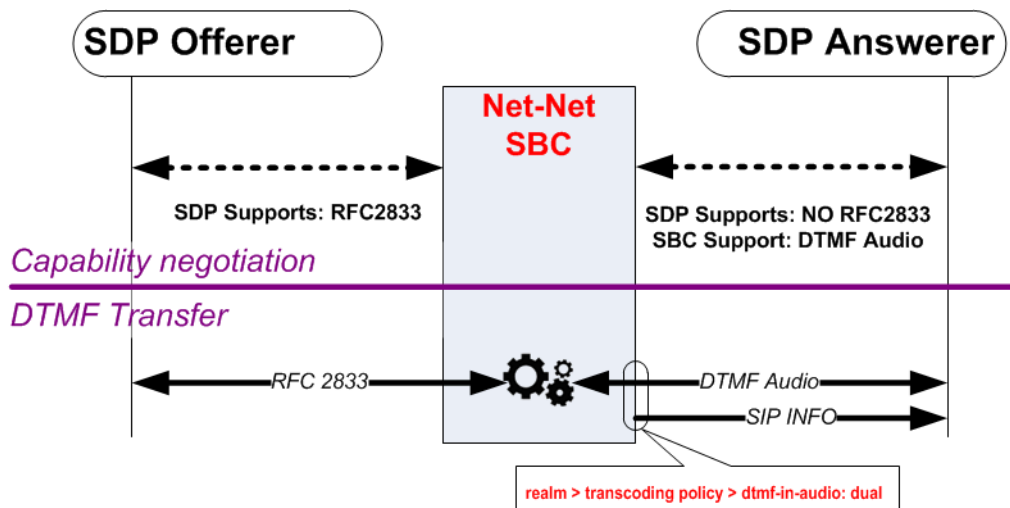
P-Dual-Info headers are only inserted into SIP INFO messages.

**Example 1**


In this example, RFC 2833 is supported on the egress side of the call. The egress SIP interface or session agent's `rfc2833-mode` is set to dual mode. When the Oracle Communications Session Border Controller forwards RFC 2833 to the SDP answerer, it also creates and forwards a corresponding SIP INFO message toward the target.

**Example 2**

In this example RFC 2833 telephone-event is not supported on the egress side of the call, but DTMF audio tones are. If the Oracle Communications Session Border Controller receives an RFC 2833 message, it is converted to DTMF audio tones. When the Oracle Communications Session Border Controller forwards DTMF audio tones to the SDP answerer, it also creates and forwards a corresponding SIP INFO message toward the target.

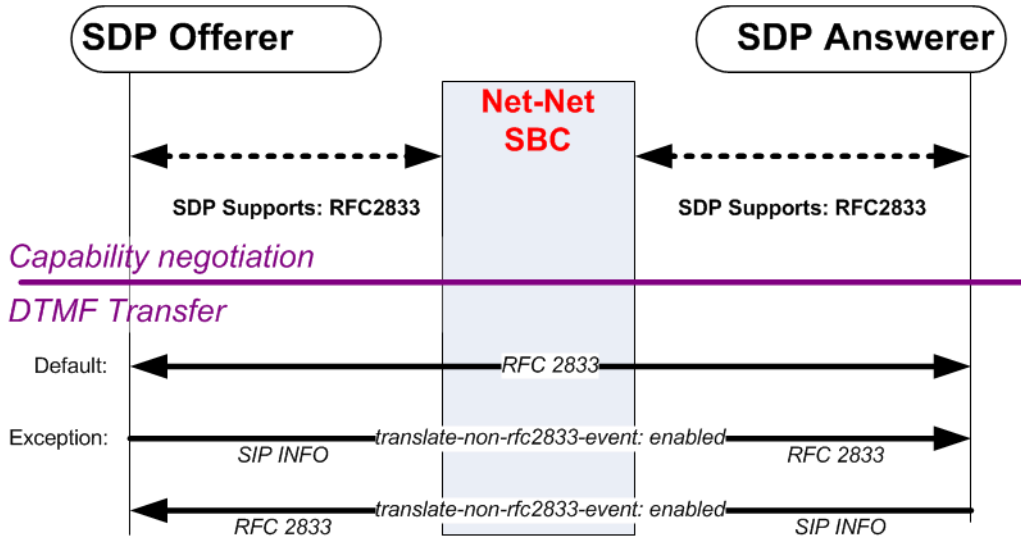
**Identical Inband with Signaling DTMF Transfer Exception**

Endpoints may send signaling-based DTMF indication (SIP INFO) to the Oracle Communications Session Border Controller at any time. Most of the time they are passed through the Oracle Communications Session Border Controller unchanged. You can enable an exception to this behavior forcing signaling messages to either RFC 2833 or DTMF audio tones depending on the call's DTMF transfer mode. The two parameters to enable their respective exceptions are located in the media manager configuration.

 **Note:** These behavior exceptions are only applicable when both sides of the call support and prefer the same DTMF transfer mode.

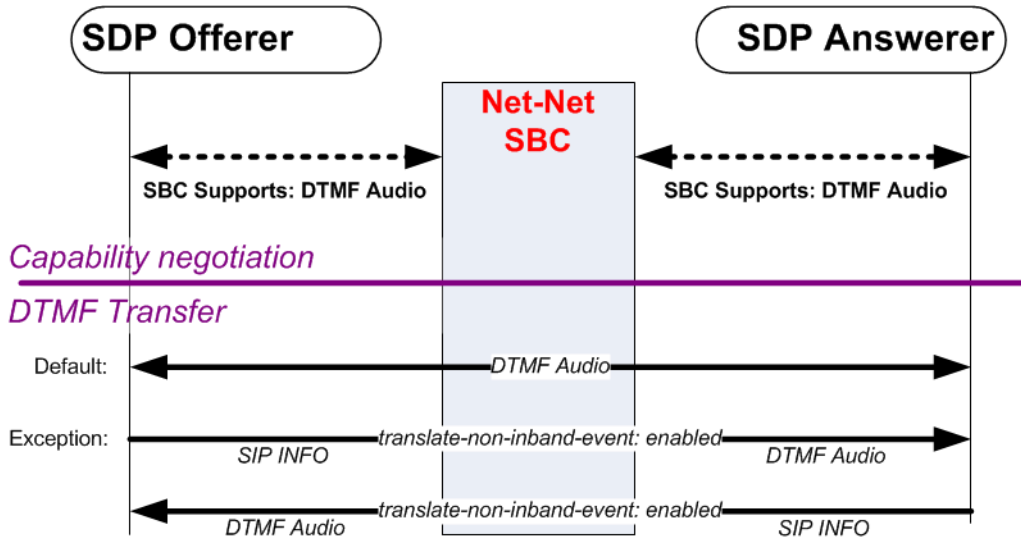
### Override Preferred RFC 2833

When RFC 2833 is supported on both sides of the call, it is the preferred method of DTMF indication transport. To override this behavior, enable the translate non rfc 2833 parameter.



### Override Preferred DTMF Audio

When DTMF audio tones are supported on both sides of the call, it is the preferred method of DTMF indication transport. To override this behavior, enable the translate non inband event parameter.



**Note:** Enabling one or both of these exceptions can cost DTMF translation resources whether they occur in the network processors or on transcoding modules. The translate-non-inband-event exception is especially costing because it reserves transcoding for all calls that resolve to support for DTMFable to DTMFable codecs.

### DTMF Transfer for Spiral Calls

A spiral call occurs when a call's signaling messages loop back through the Oracle Communications Session Border Controller. Most commonly the signaling path is from one UA, through the Oracle Communications Session Border Controller, to a call server, back through the Oracle Communications Session Border Controller, to another UA. The media path is from one UA, through the Oracle Communications Session Border Controller, to the other UA. For

DTMF indication processing, only the call legs between the endpoints and Oracle Communications Session Border Controller are considered.

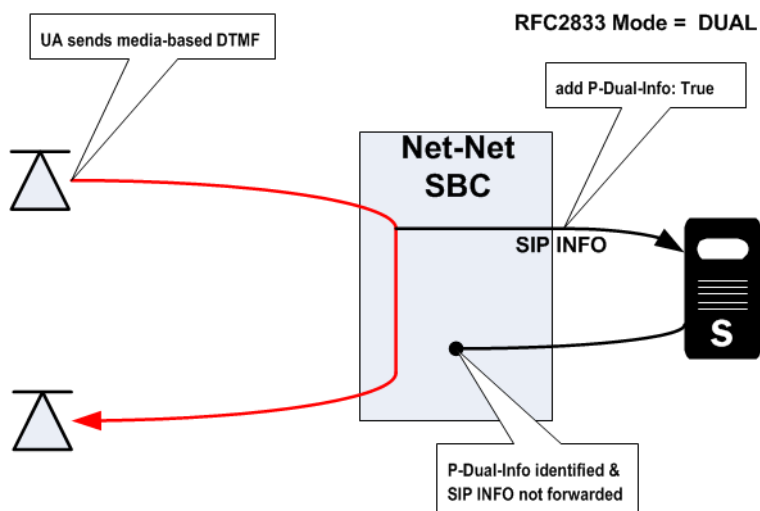
The Oracle Communications Session Border Controller evaluates that the signaling path to and from the call server terminates on the same IP address and port on the Oracle Communications Session Border Controller. This means that it's a spiral call. In addition, the media IP addresses and ports in the SDP indicate that the Oracle Communications Session Border Controller does not need to send the media into the call server's realm.

An issue occurs when DTMF indication is relayed either by RFC 2833 or DTMF audio tones for a spiral call. Since the DTMF indication is in the media path, the call server remains unaware of the signaling; no media-based DTMF indication digits will ever reach the call server. In order to include the call server in the DTMF-indication signaling, you should set the `dtmf-in-audio` and/or `rfc-2833` mode used for the realm or signaling interface where the call server is to dual.

## P-Dual-Info Header

The P-Dual-Info header is used in a spiral call scenario, when the call leg to (and from) the call server is set to dual. While the media portion of a spiral call goes from endpoint to endpoint through the Oracle Communications Session Border Controller, the signaling portion of the call loops through a call server.

The Oracle Communications Session Border Controller inserts a P-Dual-Info header into a SIP info message sent to the call server, which in turn forwards the SIP INFO message back to the Oracle Communications Session Border Controller. Seeing the P-Dual-Info header, the Oracle Communications Session Border Controller knows not to forward this SIP INFO message to the target endpoint because it would be a duplication of DTMF indication already sent to the endpoint in media format.



## DTMF Transfer Hardware Processing

DTMF transfer processing, the conversion between two DTMF transfer types, occurs in either the transcoding NIU's Digital Signal Processors (DSPs) or the Network Processors (NPs). Understanding where the processing takes place is important to determine which subsystem uses extra processing load per conversion.

There are a few rules you can use to determine which subsystem performs the DTMF transfer processing:

- If audio transcoding is enabled for the call, DTMF transfer processing occurs in the transcoding modules.
- If DTMF audio tones are generated from RFC 2833 or from signaling messages (SIP INFO), DTMF transfer processing occurs in the transcoding modules.
- If the global `translate non inband event` parameter is enabled, DTMF transfer processing occurs in the transcoding modules.
- If signaling to RFC 2833 processing occurs in either direction of the call, and the previous 3 conditions are not valid, DTMF transfer processing occurs in the NPs.

# DTMF Transfer Configuration

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## RFC 2833 Session Agent Configuration

Session agents, used as a way to classify and act on a subset of a signaling interface's traffic, also have **rfc2833-mode** and **rfc2833-payload** parameters. The configurations of these parameters overrides the configuration of the same-named parameters on the signaling interface where the session agent resides. You can set the **rfc2833-mode** parameter to **none** for a session agent to revert to the parent signaling interface's two RFC 2833 settings.

## ACLI Configuration and Instructions

This section explains how to configure the RFC 2833 mode on a signaling interface and on a session agent configured for that signaling interface. The session agent's configuration takes precedence over the signaling interface, unless the session agent's **rfc2833-mode** is set to **none**. In that case, the signaling interface's configuration is used for applicable traffic.

### SIP Interface

To configure the RFC 2833 mode on a SIP interface:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-interface** and press Enter.

```
ACMEPACKET (session-router) # sip-interface
```

4. If you are adding support for this feature to a pre-existing SIP interface, then you must select the specific configuration instance, using the ACLI **select** command.
5. **rfc2833-mode**—Set this parameter to either **transparent**, **preferred**, or **dual** based upon the behavior you want for this SIP interface.
  - **transparent**—does not add RFC 2833 telephone-event into SDP if not present, and does not prefer.
  - **preferred**—adds RFC 2833 telephone-event media type into SDP and prefers to use this method for DTMF indication.
  - **dual**—adds RFC 2833 telephone-event media type into SDP and sends both SDP and signaling-based DTMF indications if possible.
6. **rfc2833-payload**—Set this parameter to the media-type value you wish to use when inserting RFC 2833 telephone-events into an SDP offer. 101 is the generally accepted media type for RFC 2833 telephone-events.
7. Save and activate your configuration.

### Session Agent

Session agent RFC 2833 mode configurations override those on the signaling interface where they exist. The **none** parameter is used to defer to the signaling interface.

To configure the RFC 2833 mode on a session agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **session-agent** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET(session-router) # session-agent
ACMEPACKET(session-agent) #
```

4. Select the session agent where you want this feature.

```
ACMEPACKET(session-agent) # select 1
```

5. **rfc2833-mode**—Set this parameter to either **none**, **transparent**, **preferred**, or **dual** based upon the behavior you want for this SIP interface.
  - **none**—defaults to the behavior of the SIP interface for traffic that matches this session agent.
  - **transparent**—does not add RFC 2833 telephone-event into SDP if not present, and does not prefer.
  - **preferred**—adds RFC 2833 telephone-event media type into SDP and prefers to use this method for DTMF indication.
  - **dual**—adds RFC 2833 telephone-event media type into SDP and sends both SDP and signaling-based DTMF indications if possible.
6. **rfc2833-payload**—Set this parameter to the media-type value you wish to use when inserting RFC 2833 telephone-events into an SDP offer. 101 is the generally accepted media type for RFC 2833 telephone-events.
7. Save and activate your configuration.

### Codec Policy

To configure a codec policy to support DTMF audio tones, as transcoded:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter.

```
configure# media-manager
```

3. Type **codec-policy** and press Enter.

```
ACMEPACKET(media-manager) # codec-policy
ACMEPACKET(codec-policy) #
```

4. If you are adding support for this feature to a pre-existing configuration, then you must select the specific configuration instance, using the ACLI **select** command.

```
ACMEPACKET(codec-policy) # select
<name>:
1: private
2: public
selection:1
ACMEPACKET(codec-policy) #
```

5. **dtmf-in-audio**—Set this parameter to **disabled**, **preferred**, or **dual** based upon how the Oracle Communications Session Border Controller should support the conversion of signaling messages or RFC 2833 to DTMF Audio tones in the realm where this codec policy is active.
  - **disabled**—does not support DTMF audio tones as transcoded in this realm.
  - **preferred**—supports DTMF audio tones as transcoded in this realm.
  - **dual**—supports both transcoded DTMF audio tones and signaling-based DTMF indications if possible.
6. Save and activate your configuration.

### Translate Non2833 Event Behavior

To configure the exceptional behavior:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-level configuration elements.

```
ACMEPACKET(configure) # media-manager
ACMEPACKET(media-manager) #
```

## DTMF Transfer and Support

---

3. Type **media-manager** and press Enter to begin configuring this feature.

```
ACMEPACKET (media-manager) # media-manager
ACMEPACKET (media-manager-config) #
```

4. **translate-non-rfc2833-event**—Set this parameter to enabled to use non-default behavior described in Override Preferred RFC 2833.
5. **translate-non-inband-event**—Set this parameter to enabled to use non-default behavior described in Override Preferred DTMF Audio.
6. Save and activate your configuration.

### P-dual-info Header Appearance

Customizing the P-Dual-Info header is performed globally from the sip config.

To configure how the P-dual-info header appears:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the system-level configuration elements.

```
ACMEPACKET (configure) # session-router
```

3. Type **sip-config** and press Enter. The system prompt changes to let you know that you can begin configuring individual parameters.

```
ACMEPACKET (session-router) # sip-config
ACMEPACKET (sip-config) #
```

4. Type options followed by a Space.

5. After the Space, type the P-Dual-Info header information in the following format:

```
+dual-info="<header-name>"
```

For example:

```
ACMEPACKET (sip-config) # options dual-info=P-Dual-Info
```

6. Save your work using the ACLI **done** command.

## RFC 2833 Customization

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### RTP Timestamp

As a media flow with injected RFC 2833 telephone-event packets exits the Oracle Communications Session Border Controller, the newly generated RTP packets are timestamped in one of two ways. If RFC 2833 generation is preformed in the NPs, the default method of creating the timestamp for a generated RFC 2833 packet is to use the previous RTP packet's timestamp and add 1.

Alternatively the Oracle Communications Session Border Controller, can estimate the actual time that the injected RFC 2833 telephone-event packet will leave the system and use that. This method tags the injected DTMF indication packet more accurately than the than previous packet + 1 method. As an additional bonus, the packet's checksum is regenerated. This alternate timestamp creation behavior is configured by setting the **rfc2833-timestamp** parameter to enabled.

When RFC 2833 telephone-event generation is preformed by the transcoding modules, the packets' timestamps are the set to the time that the packets leave the Oracle Communications Session Border Controller,.

### RFC 2833 telephone-event duration intervals

If an incoming SIP INFO message's DTMF indication duration is unspecified, the Oracle Communications Session Border Controller, uses a default 250 ms duration for the generated RFC 2833 telephone-event. Otherwise, the SIP INFO's specified event duration is used. RFC 2833 telephone-event packets are still generated at 50 ms intervals upon



egress. At the conclusion of the DTMF indication, the three end-event packets are sent. The packet arrangement when the user presses the digit 5 for 160ms, with the default 50ms interval follows:



When either no DTMF event duration is specified, or the event duration is less than the 50ms default minimum, you can set the default RFC 2833 telephone-event duration using **default-2833-duration** parameter in the media manager configuration. This is the value that the Oracle Communications Session Border Controller, uses for the duration of a telephone event when none is specified in the incoming message. The **default-2833-duration**'s valid range is 50-5000ms. The Oracle Communications Session Border Controller, also uses this configured value when it receives a SIP INFO message with a duration less than the minimum signal duration.

You can configure the minimum duration at which RFC 2833 telephone-events are generated by the Oracle Communications Session Border Controller, using the **min-signal-duration** option in the media manager configuration, thus changing the lower threshold of the **default-2833-duration** parameter from 50 ms to your own value. If the duration the Oracle Communications Session Border Controller, receives is less than the threshold, it uses the value configured in the **default-2833-duration** parameter.



**Note:** Timestamp changes and duration changes only take effect when the 2833 timestamp (rfc-2833-timestamp) is enabled in the media manager configuration. If you enable the rfc-2833-timestamp parameter, but do not configure the default-2833-duration parameter, the default-2833-duration parameter defaults to 100 ms.

## RFC 2833 End Packets

When the Oracle Communications Session Border Controller, generates RFC 2833 telephone-event packets, they are forwarded from the egress interface every 50 ms by default. Each packet includes the digit and the running total of time the digit is held. Thus DTMF digits and events are sent incrementally to avoid having the receiver wait for the completion of the event.

At the conclusion of the signaled event, three end packets stating the total event time are sent. This redundancy compensates for RTP being an unreliable transport protocol.

You can configure your Oracle Communications Session Border Controller, to generate either the entire start-interim-end RFC 2833 packet sequence or only the last three end 2833 packets for non-signaled digit events using the **rfc2833-end-pkts-only-for-non-sig** parameter. If the parameter were enabled, the RFC 2833 telephone-event packets for the same event would appear as represented by the following graphic.



## ACLI Instructions and Examples

To configure RFC 2833 customization:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media-level configuration elements.

```
ACMEPACKET (configure)# media-manager
ACMEPACKET (media-manager)#
```

3. Type **media-manager** and press Enter to begin configuring this feature.

```
ACMEPACKET (media-manager)# media-manager
ACMEPACKET (media-manager-config)#
```

4. **rfc2833-timestamp**—Set this parameter to enabled to use the estimated real departure timestamp on an injected RFC 2833 telephone-event packet.

## DTMF Transfer and Support

---

5. **default-2833-duration**—Set this parameter to the default value you wish to use when the Oracle Communications Session Border Controller, creates or generates DTMF-indication messages.
6. **rfc2833-end-pkts-only-for-non-sig**—Set this parameter to enabled for the Oracle Communications Session Border Controller, to only send the three RFC 2833 telephone-event end-packets to indicate a total event duration, rather than the running total from time=0.
7. **options**—Set the options parameter by typing **options**, a Space, the option name **min-signal-duration=x** (where **x** is the value in milliseconds you want to use for the threshold) with a plus sign in front of it. Then press Enter.
8. Save and activate your configuration.

---

## RCS Services

### Message Session Relay Protocol

---

The Oracle Session Director supports Message Relay Protocol (MSRP) sessions initiated by Session Description Protocol (SDP) messages exchanged via the Session Initiation Protocol (SIP) offer/answer model. MSRP usage with SDP and SIP is described in Section 8 of RFC 4975, The Message Relay Protocol. The Session Director functions as a Back-to-Back User Agent (B2BUA) for MSRP sessions, terminating incoming MSRP, proxying for the MSRP session originator, initiating outgoing MSRP to the endpoint peer, and providing Network Address Translation (NAT) services.

#### MSRP Platform Requirements

MSRP is supported on the following platforms:

- AP4500
- AP6100
- AP6300

The AP3820 does not support MSRP.

Users with AP4500 systems must have an ETCv1 or ETCv2 NIU card installed in their systems. The AP6100 and AP6300 have the requisite hardware in their systems by default.

The Oracle Communications Session Border Controller now supports MSRP over IPv6 and IPv4/IPv6 interworking for MSRP.

#### MSRP IP Address Family Support

The Oracle Communications Session Border Controller supports MSRP over IPv4 and IPv6. The Oracle Communications Session Border Controller also can perform IPv4-to-IPv6 and IPv6-to-IPv4 interworking. This support is available automatically and does not require any configuration.

#### MSRP Operational Description

A sample RFC 4975-compliant Offer/Answer SDP exchange for an MSRP session is shown below.

Alice	Bob
(1) (SIP) INVITE	The first three messages
----->	use a SIP offer/answer

```

| (2) (SIP) 200 OK | model with accompanying
|<-----| SDP to negotiate an MSRP
| (3) (SIP) ACK | session
|----->|
| (4) (MSRP) SEND | Message 4 starts the MSRP
|----->| connection
| (5) (MSRP) 200 OK |
|<-----|
| (6) (SIP) BYE |
|----->| Message 7 terminates the
| (7) (SIP) 200 OK | SIP and MSRP connection
|<-----|

```

1. Alice sends an INVITE request with accompanying SDP to Bob.

The SDP media (M) line is defined in RFC 4975, and adheres to the format

m=<media> <port> <protocol> <format-list>

MSRP operations require the following values:

media	=	message
protocol	=	TCP/MSRP (for an unencrypted connection)
format-list	=	*
port	=	the TCP port (7777 in the SDP example, although any valid port number can be specified) monitored by the message originator for a response to the SDP offer

The required SDP attributes, accept-types and path, are also defined in RFC 4975.

**accept-types** contains a list of media types that the message originator is willing to receive. It may contain zero or more registered media-types, or an \* wildcard character in a space-delimited string.

**path** contains the MSRP URI of the message originator. An MSRP URI is constructed as shown below.

scheme	=	msrp (for an unencrypted connection), or
	=	msrps (for an encrypted connection)
//		
address	=	IP address of the message originator, or
	=	FQDN of the message originator
;		
port	=	the port (7777 in the SDP example, although any valid port number can be specified) monitored by the message originator for MSRP responses
session-id	=	a random local value generated by the message originator used to produce an ephemeral MSRP URI lasting only for the duration of the current MSRP session
;		
protocol	=	tcp

Alice->Bob (SIP):

```

INVITE sip:bob@example.com
SDP:
v=0
o=alice 2890844557 2890844559 IN IP4 alicepc.example.com
s=

```

```
c=IN IP4 alicepc.example.com
m=message 7777 TCP/MSRP *
a=accept-types:text/plain
a=path:msrp://alicepc.example.com:7777/iau39soe2843z;tcp
```

- Bob accepts the SDP offer, generates a local session-id (contained in his MSRP URI specified by the path attribute), and issues a 200 OK response to Alice.

The port parameter in the Media line indicates that Bob listens for MSRP messages on TCP port 8888

```
Bob->Alice (SIP):
```

```
SIP/2.0 200 OK
SDP:
v=0
o=bob 2890844612 2890844616 IN IP4 bob.example.com
s=
c=IN IP4 bob.example.com
m=message 8888 TCP/MSRP *
a=accept-types:text/plain
a=path:msrp://bob.example.com:8888/9di4eae923wzd;tcp
```

- Alice ACKs Bob's answer, establishing a SIP session between the two.

```
Alice->Bob (SIP):
```

```
ACK sip:bob@example.com
```

- Alice initiates an MSRP session with an MSRP SEND request to Bob.

All MSRP requests begin with the MSRP start line, which contains three elements.

protocol-id	=	MSRP
transaction-id	=	an ephemeral transaction identifier (d93kswow in the following MSRP example) used to correlate MSRP requests and responses, and to frame the contents of the MSRP message
method	=	SEND (MSRP method that supports data transfer)

The MSRP start line is followed by the To-Path and From-Path headers, which contain destination and source addresses — the MSRP URIs exchanged during the MSRP negotiation.

The Message-ID header contains a random string generated by the message originator. This ephemeral value whose lifetime is measured from message start to message end, is used to correlate MSRP status reports with a specific message, and to re-assemble MSRP message fragments (chunks in MSRP terminology).

The Byte-Range header contains the message length, in bytes, and the specific byte range carried by this message. Contents of this header are generally of interest only if the message has been fragmented.

The Content-Type header describes the message type, and must conform to the results of the MSRP negotiation.

The actual message follows the Content-Type header.

Finally, the SEND request is closed with an end-line of seven hyphens, the transaction-id, and a

\$ to indicate that this request contains the end of a complete message, or

+ to indicate that this request does not contain the message end

```
Alice->Bob (MSRP):
```

```
MSRP d93kswow SEND
To-Path: msrp://bob.example.com:8888/9di4eae923wzd;tcp
From-Path: msrp://alicepc.example.com:7777/iau39soe2843z;tcp
Message-ID: 12339sdqwer
Byte-Range: 1-16/16
Content-Type: text/plain
```

```
Hi, I'm Alice!
-----d93kswow$
```

- Bob acknowledges receipt with an MSRP 200 OK response to Alice.

Note that the response includes the initiator-originated transaction-id, d93kswow.

```
Bob->Alice (MSRP):

MSRP d93kswow 200 OK
To-Path: msrp://alicepc.example.com:7777/iau39soe2843z;tcp
From-Path: msrp://bob.example.com:8888/9di4eae923wzd;tcp
-----d93kswow$
```

- Alice sends a BYE request to Bob.

Alice sends a BYE request to terminate the SIP session and MSRP sessions. Alice can, of course, send more SEND requests to Bob before sending the BYE.

```
Alice->Bob (SIP):

BYE sip:bob@example.com
```

- Bob sends a 200 OK response to Alice.

```
Bob->Alice (SIP):

SIP/2.0 200 OK
```

The SIP session and the MSRP session are terminated.

## Secure MSRP Session Negotiation

An Offer/Answer SDP exchange for a TLS (secure) MSRP session is specified in RFC 4572, Connection-Oriented Media Transport over the Transport Layer Security (TLS) Protocol in the Session Description Protocol (SDP). RFC 4572 defines the syntax and semantics for an SDP fingerprint attribute that identifies the certificate (most likely a self-signed certificate) that will be presented during the TLS negotiation. A sample SDP exchange is shown below.

The protocol field (TCP/TLS/MSRP) of the media (M) line designates a TLS encrypted connection.

The fingerprint attribute is constructed as follows:

protocol	identifies the hashing method used to produce the certificate fingerprint, SHA-1 in the following sample SDP
hash value	a sequence of uppercase hexadecimal bytes separated by colons with the sequence length determined by the hash function

Offer SDP: (Alice to Bob)

```
v=0
o=usera 2890844526 2890844527 IN IP4 alice.example.com
s=
c=IN IP4 1.1.1.1
m=message 7394 TCP/TLS/MSRP *
a=accept-types:text/plain
a=path:msrps://alice.example.com:7394/2s93i9ek2a;tcp
a=fingerprint:SHA-1
4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:AB
```

Answer SDP: (Bob to Alice)

```
v=0
o=userb 2890844530 2890844532 IN IP4 bob.example.com
s= -
c=IN IP4 2.2.2.2
t=0 0
m=message 8493 TCP/TLS/MSRP *
```

```
a=accept-types:text/plain
a=path:msrps://bob.example.com:8493/si438dsaodes;tcp
a=fingerprint:SHA-1
DA:39:A3:EE:5E:6B:4B:0D:32:55:BF:EF:95:60:18:90:AF:D8:07:09
```

## MSRP Session Setup

The B2BUA processes MSRP media descriptions in offer/answer SDPs to negotiate and establish MSRP sessions and then constructs internal data flows for the actual MSRP sessions. After establishing an MSRP session, the B2BUA forwards MSRP requests and responses from and to the session participants.

### Initiating MSRP Sessions

After accepting an offer SDP with MSRP session initiation, the B2BUA constructs an egress offer SDP as follows.

1. The B2BUA sets the transport protocol of the m= line to the transport protocol of the selected egress profile.
2. If the value of listen-port of the selected egress profile is not zero, the B2BUA sets the port of the m= line to the value of listen-port. If the value of listen-port is zero, the port in the m= line is chosen from a steering port of the egress realm.
3. The B2BUA adds an a=setup attribute to the SDP. The value of this attribute is determined by the value of the **preferred-setup-role** CLI command. For example, if the value of **preferred-setup-role** is passive (the default value) the B2BUA adds the attribute a=setup:passive.
4. The B2BUA performs NAT on the MSRP Universal Resource Identifier (URI) in the a=path attribute.

The B2BUA does not include an a=fingerprint in the offer SDP if the selected egress profile has TCP/TLS/MSRP transport protocol. However, if the egress profile specifies both TCP/MSRP and TCP/TLS/MSRP the B2BUA selects the TCP/TLS/MSRP transport protocol, resulting in an egress offer containing an m= line with TCP/TLS/MSRP transport protocol. The B2BUA offers only a single media line, TCP/MSRP or TCP/TLS/MSRP. The B2BUA does not perform recursion (that is, first initiation attempt with TCP/TLS/MSRP and re-attempt with TCP/MSRP if first attempt is rejected).

### Connection Negotiation

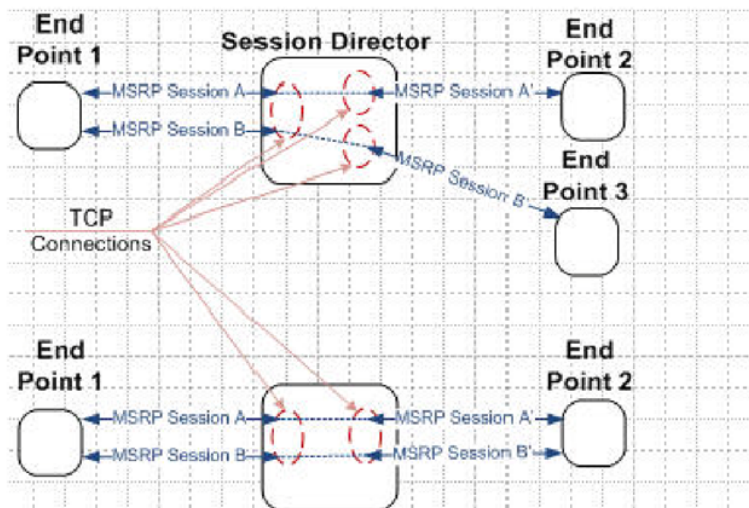
When making the offer MSRP session, a user agent client can follow RFC 4145, TCP-Based Media Transport in the Session Description Protocol (SDP), and include an a=setup attribute in a MSRP media line, or it may assume the default connection direction as specified in section 8.5 of RFC 4975, The Message Session Relay Protocol, which specifies that the endpoint that sent the original offer is responsible for connecting to its remote peer. The B2BUA, in contrast, always includes an a=setup attribute in its offer SDP, the attribute value set by the **preferred-setup-role** CLI command.

If the B2BUA receives an answer SDP that does not include an a=setup attribute, it assumes that the user agent server does not support connection negotiation per RFC 4145. In that case, the B2BUA assumes the active role as specified in RFC 4975, and makes an outgoing connection.

If the B2BUA receives an offer or answer SDP that does include an a=setup attribute, it follows the connection negotiation as specified by RFC 4145. When the B2BUA receives an offer SDP with the attribute a=setup:actpass attribute, it will set the a=setup attribute of the answer SDP to the value set by the **preferred-setup-role** CLI command.

### Multiple MSRP Connections

The MSRP B2BUA supports sharing of a single TCP connection by multiple MSRP sessions. In such a topology, each TCP connection maintains a list active MSRP sessions.



With regard to the above figure, the upper Session Director's TCP connection between Endpoint 1 and the MSRS B2BUA is shared by 2 MSRP sessions. At bottom, each MSRP session uses a separate TCP connection. When the B2BUA assumes the active role, it always initiate a separate connection to the MSRP endpoint peers, endpoints 2 and 3 as shown above.

When the list of active MSRP sessions for a shared TCP connection becomes empty as a result of SIP session terminations or disconnections of peer TCP connections, B2BUA disconnects the shared TCP connection. If the shared connection is disconnected by the peer, the B2BUA disconnects all the separate TCP connections it initiated for the MSRP sessions that use the shared connection.

### Accepting Connections

When the B2BUA is in passive mode, it listens for incoming connection from the active party, monitoring the port specified by the `listen-port` CLI command.

### Making Connections

When the B2BUA is in active mode, it makes the connection to the passive party, selecting a port allocated from the steering-pool of the applicable realm.

## MSRP Session Termination

An MSRP session is terminated by sending or receiving a BYE request in the parent SIP session, that is the session that set up the MSRP exchange, followed by the disconnect of the TCP connection that supports MSRP message exchange.

Some SIP endpoints close their MSRP TCP connections upon receiving a BYE possibly before its peer finishes sending all the MSRP messages and closes the connection. To facilitate graceful session completion, the B2BUA offers a configurable time delay between the receipt of a BYE request from an MSRP endpoint and the transmission of the BYE to the recipient endpoint peer.

With the configurable BYE delay enabled, the MSRP B2BUA upon receiving a BYE request acknowledges the request with a 200 OK response. The B2BUA, however, does not immediately forward the BYE to the other MSRP endpoint.

Rather, the B2BUA triggers two user-configurable timers that monitor the specific MSRP session initiated by the current SIP exchange. The first timer measures inactivity intervals on media flows (calling-to-called and called-to-calling) associated with the MSRP session to be closed. The second timer sets an unconditional outer limit at which point the delayed BYE is transmitted to the MSRP endpoint and the MSRP is terminated.

Expiration of either timer generates an internal stop event which generates transmission of the delayed BYE to the MSRP endpoint and termination of the underlying SIP connection.



Note that the MSRP-specific timers, the session inactivity timer and the MSRP delayed-BYE timer, are roughly analogous to two existing TCP timers, the TCP subsequent guard timer and the TCP flow time limit timer. The following chart summarizes timer operations.

#### MSRP interval timer

##### Purpose:

Measures inactivity periods within MSRP data sessions. The timer is triggered in the absence of MSRP data. If new MSRP is not detected prior to timer expiration, a stop event is generated resulting in delayed BYE transmission and MSRP connection termination. If new MSRP traffic is detected prior to timer expiration, the timer is reset.

##### Associated ACLI Command:

**session-inactivity-timer**

Allowable command values: 0 | 5 to 10 (seconds)

Default value: 5

When set to 0, session monitoring is disabled. No MSRP session monitoring is done when the corresponding SIP session receives a BYE request.

##### Timer Expiration:

Initiates forwarding of the delayed BYE request to the recipient MSRP endpoint and tear down of the SIP connection.

#### TCP subsequent guard timer

##### Purpose:

Measures the maximum interval allowed between media-over-TCP packets.

##### Associated ACLI Command:

**tcp-subsq-guard-timer**

Allowable command values: 0 to 999999999 (seconds)

Default value: 300

##### Timer Expiration:

Initiates tear down of the TCP connection. In the possible, but unlikely event that the value assigned to this timer is less than the value assigned to the MSRP interval timer, expiration initiates forwarding of the delayed BYE request to the recipient MSRP endpoint and tear down of the TCP connection.

#### MSRP delayed -BYE timer

##### Purpose:

Measures inactivity periods within MSRP traffic sessions. The timer is triggered in the absence of MSRP data. If new MSRP is not detected prior to timer expiration, a stop event is generated resulting in delayed BYE transmission and MSRP connection termination. If new MSRP traffic is detected prior to timer expiration, the timer is reset.

##### Associated ACLI Command:

**msrp-delayed-bye-timer**

Allowable command values: 0 to 60 (seconds)

Default value: 15

##### Timer Expiration:

Initiates forwarding of the delayed BYE request to the recipient MSRP endpoint and tear down of the SIP connection.

#### TCP flow time limit timer

##### Purpose:

Measures the maximum allowed lifetime of a media-over-TCP connection.

Associated CLI Command:

**tcp-flow-limit-timer**

Allowable command values: 0 to 999999999 (seconds)

Default value: 86400 (1 day)

Timer Expiration:

Initiates tear down of the TCP connection. In the possible, but unlikely event that the value assigned to this timer is less than the value assigned to the MSRP delayed-BYE timer, expiration initiates forwarding of the delayed BYE request to the recipient MSRP endpoint and tear down of the TCP connection.

## Network Address Translation

If the value of the **uri-translation** CLI command is **enabled**, the B2BUA performs network address translation on the MSRP URI in the a=path attribute and in the From-Path and To-Path of MSRP requests and response. The host part of the URI will be the IP address of the steering pool of the realm. The port will be chosen as follows:

If the B2BUA role is in passive mode, and the listen-port CLI command is non-zero, the B2BUA monitors that specified port.

If the B2BUA role is in passive mode, and the listen-port CLI command is zero, the B2BUA selects a port from the steering pool of the applicable realm and monitors that chosen port.

If the B2BUA role is in active mode, the port is chosen from the steering pool of the applicable realm. Since B2BUA is active, that port is used only to support the outgoing connection.

## Certificate Fingerprint

If the value of the field **require-fingerprint** in the ingress and/or egress tcp-media-profile is enabled, and the transport protocol is TCP/TLS/MSRP the B2BUA requires the offer and/or the answer SDPs, respectively, to have an a=fingerprint attribute as specified in RFC 4572, Connection-Oriented Media Transport over the Transport Layer Security (TLS) Protocol in the Session Description Protocol (SDP).

If the B2BUA receives an offer SDP without a=fingerprint attribute, it rejects the offer SDP. If the rejected SDP is in an INVITE request, the B2BUA issues a 488 Not Acceptable Here response. If the rejected SDP is in a 200 OK response, the B2BUA ACKs that 200 OK, sends a BYE to the server user agent, and a 503 Service Unavailable response to the client user agent.

If the B2BUA receives an answer SDP without a=fingerprint attribute, it terminates the related SIP session. If the rejected SDP is in a 200 OK response, the B2BUA ACKs that 200 OK, sends a BYE to the server user agent, and a 503 Service Unavailable response to the client user agent. If the rejected SDP is in an ACK, the B2BUA simply sends a BYE to both the server and client user agent.

If the value of the field **require-fingerprint** in the profile is disabled, the B2BUA accepts offer and answer SDP that do not include an a=fingerprint attribute.

Because the B2BUA certificate is expected to be signed by a trusted Certification Authority, an a=fingerprint attribute is not included in offer and answer SDPs sent by the B2BUA.

The B2BUA maintains a fingerprint mismatch counter for each MSRP session. This counter is incremented when the B2BUA cannot match the certificate it receives during the TLS handshake with the fingerprint it receives in the SDP.

## MSRP Configuration

---

MSRP configuration consists of the following steps.

1. Configure the msrp-config configuration object that governs MSRP global behavior.
2. Configure one or more tcp-media-profile configuration objects that define MSRP operations within a realm.

3. Assign a tcp-media-profile to a target realm.
4. If MSRP sessions are secured with TLS, create and assign tls-profile configuration objects to the tcp-media-profile of the target realm.
5. Create and assign steering-pools configuration objects to target realms.

## msrp-config Configuration

Use the following procedure to perform MSRP global configuration.

1. From superuser mode, use the following command sequence to access msrp-config configuration mode. While in msrp-config mode, you configure global MSRP behavior.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# msrp-config
ACMEPACKET(msrp-config)# ?
state                               state
uri-translation                      perform translation of MSRP URI
session-inactivity-timer            timer value (seconds) for session inactivity
                                     monitoring period
select                               select msrp config to edit
no                                   delete msrp config
show                                 show msrp config
done                                 write msrp config information
exit                                 return to previous menu
ACMEPACKET(msrp-config)#
```

2. Use the **state** parameter to enable MSRP operations.

Retain the default value, **enabled**, to enable MSRP operations.

If necessary, you can use **disabled** to temporarily suspend all MSRP operations.

```
ACMEPACKET(msrp-config)# state enabled
ACMEPACKET(msrp-config)#
```

3. Use the **uri-translation** parameter to enable or disable NAT of URIs found in the From-Path and To-Path headers of MSRP requests and responses, and in a=path attributes found in SDP offers.

NAT is enabled by default.

Retain the default value (**enabled**) to enable NAT; use **disabled** to disable NAT.

```
ACMEPACKET(msrp-config)# uri-translation enabled
ACMEPACKET(msrp-config)#
```

4. Use the **session-inactivity-timer** parameter in connection with the **msrp-delayed-bye-timer** parameter to implement the delayed transmission of SIP BYE requests, thus establishing a configurable transition interval allowing for the completion of active MSRP sessions.

The **session-inactivity-timer** parameter specifies the maximum inactivity interval (defined as the absence of transmitted data) tolerated before the MSRP connection is terminated.

Retain the default value (**5**), or specify another inactivity interval within the range 5 to 10 seconds.

```
ACMEPACKET(msrp-config)# session-inactivity-timer 7
ACMEPACKET(msrp-config)#
```

5. Use **done**, **exit**, and **verify-config** to complete MSRP global configuration.
6. If you wish to implement the delayed transmission of SIP BYE requests, use the following command sequence to access sip-config configuration model

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)#
```

- Use the **msrp-delayed-bye-timer** parameter to enable the delayed transmission of SIP BYE requests, thus establishing a configurable transition interval allowing for the completion of active MSRP sessions.

The **msrp-delayed-bye-timer** parameter specifies the maximum delay period allowed before transmitting the delayed BYE request.

Retain the default value (15), or specify another delay period within the range 1 to 60 seconds.

Delayed transmission of BYE requests is enabled by default. Use the special value of 0 to disable delay, and transmit BYE requests immediately upon receipt.

```
ACMEPACKET(sip-config)# msrp-delayed-bye-timer 20
ACMEPACKET(sip-config)#
```

## tcp-media-profile Configuration

Use the following procedure to construct a TCP Media Profile that defines MSRP operations within a realm.

- From superuser mode, use the following command sequence to access tcp-media-profile configuration mode. While in tcp-media-profile mode, you begin construction of a TCP Media Profile.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# tcp-media-profile
ACMEPACKET(tcp-media-profile)# ?
name                name
profile-list        list of TCP media profiles
select              select profile to edit
no                  delete profile
show                show profile
done                write profile information
exit                return to previous menu
ACMEPACKET(tcp-media-profile)#
```

- Use the **name** parameter to provide a unique identifier for this TCP Media Profile instance.

```
ACMEPACKET(tcp-media-profile)# name tlsMSRP
ACMEPACKET(tcp-media-profile)#
```

- Use the **profile-list** command to move to tcp-media-profile-entry configuration mode. While in this mode, you complete configuration of the named tls-media-profile.

```
ACMEPACKET(tcp-media-profile)# profile-list
ACMEPACKET(tcp-media-profile-entry)# ?
media-type           media type
transport-protocol  transport protocol
listen-port          listening port
preferred-setup-role preferred setup role
tls-profile           tls profile name
require-fingerprint always require TLS certificate fingerprint
select              select a profile entry to edit
no                  delete selected profile entry
show                 show profile entry information
done                write profile entry information
exit                return to previous menu
ACMEPACKET(tcp-media-profile-entry)#
```

- Use the **listen-port** parameter to identify the TCP port monitored by the B2BUA for incoming MSRP connections.

Supported values are integer values within the range 0 (the default value) through 65535.

The 0 default value indicates that the listening port will be chosen by the B2BUA from the steering pool of the realm (which the tcp-media-profile belongs to).

```
ACMEPACKET(tcp-media-profile-entry)# listen-port 43000
ACMEPACKET(tcp-media-profile-entry)#
```

- Use the **media-type** parameter in conjunction with the **transport-protocol** parameter to identify the media-types and transport protocols (found in the SDP media description, **m=**, field as described in RFC 4566, SDP: Session Description Protocol) subject to this TCP Media Profile.

**media-type** identifies the media subject to this TCP Media Profile. Retain the default value, **message**, for MSRP operations.

**transport-protocol** identifies the transport layer protocols subject to this TCP Media Profile. Use either **TCP/MSRP** to specify unsecured TCP traffic or **TCP/TLS/MSRP** to specify secured/encrypted TLS traffic.

```
ACMEPACKET(tcp-media-profile-entry) # transport-protocol TCP/TLS/MSRP
ACMEPACKET(tcp-media-profile-entry) #
```

- If **transport-protocol** is **TCP/TLS/MSRP**, use the **tls-profile** parameter to identify the TLS profile that specifies the cryptographic resources available to support TLS operations.

This parameter can be safely ignored if **transport-protocol** is **TCP/MSRP**.

```
ACMEPACKET(tcp-media-profile-entry) # tls-profile msrp1
ACMEPACKET(tcp-media-profile-entry) #
```

- If **transport-protocol** is **TCP/TLS/MSRP**, use the **require-fingerprint** parameter to enable or disable endpoint authentication using the certificate fingerprint methodology defined in RFC 4572, Connection-Oriented Media Transport over the Transport Layer Security (TLS) Protocol in the Session Description Protocol (SDP).

By default, mutual authentication is **disabled**.

This parameter can be safely ignored if **transport-protocol** is **TCP/MSRP**.

```
ACMEPACKET(tcp-media-profile-entry) # require-fingerprint enabled
ACMEPACKET(tcp-media-profile-entry) #
```

- Use the **preferred-setup-role** parameter to specify the value the B2BUA uses for the **a=setup** attribute when negotiating the setup up role, regardless of the role (offerer or answerer) assumed by the B2BUA in the SDP offer/answer exchange.

The value of **preferred-setup-role** is used for the value of the **a=setup** attribute when the B2BUA makes an offer SDP and when the B2BUA replies to an offer SDP that has **a=setup:actpass**. It is not used when the B2BUA is forced into a role by the offerer, that is, if the offerer sends **a=setup:active**, the B2BUA must answer with **a=setup:passive** (and vice versa).

Allowable values are **passive** (the default) and **active**.

**active** indicates that the B2BUA creates an outgoing connection.

**passive** indicates that the B2BUA accepts an incoming connection.

Acme Packet strongly recommends that users retain the default value, **passive**.

```
ACMEPACKET(tcp-media-profile-entry) # preferred-setup-role passive
ACMEPACKET(tcp-media-profile-entry) #
```

- Use **done**, **exit**, and **verify-config** to complete **tcp-media-profile** configuration.

- Repeat Steps 1 through 9 to configure additional **tcp-media-profiles** as required.

## realm Configuration

Use the following procedure to assign a single, specific **tcp-media-profile** to a target realm.

- From superuser mode, use the following command sequence to access **realm-config** configuration mode. While in **realm-config** mode, you assign a **tcp-media-profile** to a realm.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# media-manager
ACMEPACKET(media-manager)# realm-config
ACMEPACKET(realm-config)#
```


- Use the **select** command to identify the target realm.
- Use the **tcp-media-profile** parameter to assign a specific, named tcp-media-profile to the target realm.

```
ACMEPACKET (realm-config) # tcp-media-profile tlsMutualAuth
ACMEPACKET (realm-config) #
```

- Use **done**, **exit**, and **verify-config** to complete tcp-media-profile assignment.

## tls-profile Configuration

Use the following procedure to create a tls-profile configuration object, which specifies cryptographic resources available in support of TLS operations.

 **Note:** The option allow-self-signed-cert is only available for MSRP connections.

- Access the **tls-profile** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# tls-profile
ACMEPACKET(tls-profile)#
```

- Use the **name** parameter to provide a unique identifier for this TLS Profile instance.

```
ACMEPACKET(tls-profile)# name tlsMutualAuth
ACMEPACKET(tls-profile)#
```

- If the require-fingerprint attribute of the tcp-media-profile is set to **enabled**, use the **mutual-authenticate** parameter to enable mutual authentication.

```
ACMEPACKET(tls-profile)# mutual-authenticate enabled
ACMEPACKET(tls-profile)#
```

- Retain default values for other parameters.
- Type **done** to save your configuration.
- Repeat Steps 1 through 5 to configure additional tls-profiles as required.

## MSRP Management Monitoring

The **show mbcd** statistics command now displays MSRP flow counts for the current statistics window.

```
ACMEPACKET# show mbcd statistics
16:38:49-145
MBCD Status          -- Period --  ----- Lifetime -----
                    Active   High   Total      Total  PerMax   High
Client Sessions      1       1       0           1       1       1
Client Trans         0       0       0           3       3       1
Contexts             2       2       0           3       2       3
Flows                5       5       0           7       4       7
Flow-Port            0       0       0           0       0       0
Flow-NAT             5       5       0          11       6       7
Flow-RTCP            0       0       0           0       0       0
Flow-Hairpin         0       0       0           0       0       0
Flow-Released        0       0       0           0       0       0
MSM-Release          0       0       0           0       0       0
Rel-Port             0       0       0           0       0       0
Rel-Hairpin          0       0       0           0       0       0
NAT Entries          5       5       0          11       6       7
Free Ports           12      12       0          14      12      12
Used Ports            0       0       0           2       2       2
Port Sorts           -       -       0           0       0       0
Queued Notify        0       0       0           0       0       0
MBC Trans            0       0       0           3       3       3
```

MBC Ignored	-	-	0	0	0	0
ARP Trans	0	0	0	0	0	0
Relatch NAT	0	0	0	0	0	0
Relatch RTCP	0	0	0	0	0	0
Flow-MSRP	2	2	0	2	2	2
S RTP Only Flows	0	0	0	0	0	0
S RTP Only Flow	0	0	0	0	0	0
S RTP Collapsed	0	0	0	0	0	0
S RTP Sessions	0	0	0	0	0	0
Flow Rate = 0.0						
Load Rate = 0.0						
ACMEPACKET#						

A new command (**show msrp stats**) displays the following cumulative counts:

```
ACMEPACKET# show msrp statistics
PPM_ID_MSRP:
  msrpStatsActiveSessions           : 0
  msrpStatsMaxActiveSessions        : 2
  msrpStatsEstablishedSessions      : 2
  msrpStatsProvisionedSessions      : 2
  msrpStatsFinishedSessions         : 2
  msrpStatsReleasedSessions         : 2
  msrpStatsFailedSessionsCannotRoute : 0
  msrpStatsFailedSessionsCannotConnect : 0
  msrpStatsFailedSessionsFingerprintMismatch : 0
  msrpStatsFailedMessagesCannotBeSent : 0
  msrpStatsFailedMessagesMalformed  : 0
  msrpStatsSendQFullEvents          : 0
  msrpStatsSendQCongestedEvents     : 0
  msrpStatsSendQCongestionRelievedEvents : 0
  msrpStatsStreamRequestsReceived   : 4
  msrpStatsStreamRequestSent        : 4
  msrpStatsStreamResponsesReceived  : 4
  msrpStatsStreamResponseSent       : 4
  msrpStatsStreamErrorNoTransId     : 0
  msrpStatsStreamErrorNoMsgType     : 0
  msrpStatsStreamErrorNoByteLength  : 0
ACMEPACKET#
```

## RCS<sub>e</sub> TLS/TCP Re-Use Connections

In an RCS<sub>e</sub> environment the sip-interface reuse-connections option is used to make the Oracle Communications Session Border Controller retain the TCP/TLS connection established by the endpoint during the registration for all subsequent messages to that endpoint, essentially providing for a persistent connection between the Oracle Communications Session Border Controller and the user equipment (UE).

Field experience uncovered an implementation deficiency associated with these persistent connections particularly within RCS<sub>e</sub> deployments. The basic scenario is as follows:

1. The UE registers in a TLS realm on SBC1. SBC1 stores the IP:Port from VIA (and Contact) as alias of the currently established connection.
2. The UE transits to another realm/sip-port (same or different Oracle Communications Session Border Controller) without previously unregistering or closing the TCP connection with the TLS sip-port on SBC1.
3. UE goes back to the TLS realm in SBC1 and establishes a new connection — same source IP as in Step 1, but a different port as in Step 1.

The problem arises at Step 3. If the Oracle Communications Session Border Controller has not detected that the TLS connection established in Step 1 has been effectively terminated, it will not update the alias connection to that established in Step 3, but instead continue to attempt to use the Step 1 connection.

This means that the next message from the core side to the UE will fail, since the Oracle Communications Session Border Controller will attempt to send the message of the dead TLS connection — that is using the IP address:port pair passed in Step 1.

All communications to this UE will fail until it sends the next message to the Oracle Communications Session Border Controller, when the alias connection will be update to the TLS connection in Step 3.

To resolve this issue, the Oracle Communications Session Border Controller needs to always update the alias table when it receives a new inbound connection on the configured sip-interface.

### Option Configuration Guidelines

The following table lists the full range of options that pertain to TLS/TCP Connection reuse with an emphasis on use in RCSe environments

Option	Connection Behavior
reuse-connections	Use/retain first inbound connection until explicitly closed
reuse-connections=latest	Use the last inbound connection, update the alias for each new connection
reuse-connections=no	Establish new connection at each UE access
NOT CONFIGURED	Equivalent to reuse-connections=no

### RCSE TLS/TCP Re-Use Connections Configuration

To configure the reuse-connections option:

1. From Superuser mode, use the following command sequence to navigate to the sip-interface configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)# sip-config
ACMEPACKET(sip-config)# sip-interface
ACMEPACKET(sip-interface)# option reuse-connections=latest
ACMEPACKET(sip-interface)#
```

2. Use **done**, **exit**, and **verify-config** to complete configuration.



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## Call Traffic Monitoring



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### SelectiveCall Recording SIPREC

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The SIPREC protocol is the protocol used to interact between a Session Recording Client (SRC) (the role performed by Oracle Communications Session Border Controller) and a Session Recording Server (SRS) (a 3rd party call recorder or Oracle Communications Interactive Session Recorder's Record and Store Server (RSS)). It controls the recording of media transmitted in the context of a communications session (CS) between multiple user agents.

SIPREC provides a selective-based call recording solution that increases media and signaling performance on 3rd party call recording servers, more robust failovers, and the ability to selectively record.

-  **Note:** SIPREC isolates the 3rd party recorders from the communication session. The 3rd party recorders can determine whether or not recording is desired.
-  **Note:** The SRC starts a recording session for every call within a configured realm. All call filtering, if desired, must be accomplished by the SRS. The SRS performs the filtering and selection of which sessions it should record.

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### License Hardware Requirements

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SIPREC requires the purchase of a Session Recording license. Contact your Oracle Representative for more information.

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### SIPREC Feature

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The SIPREC feature supports active recording, where the Oracle Communications Session Border Controller acting as the SRC, purposefully streams media to the Oracle Communications Interactive Session Recorder's RSS (or 3rd party call recorder) acting as the SRS. The SRC and SRS act as SIP User Agents (UAs). The SRC provides additional information to the SRS to describe the communication sessions, participants and media streams for the recording session to facilitate archival and retrieval of the recorded information.

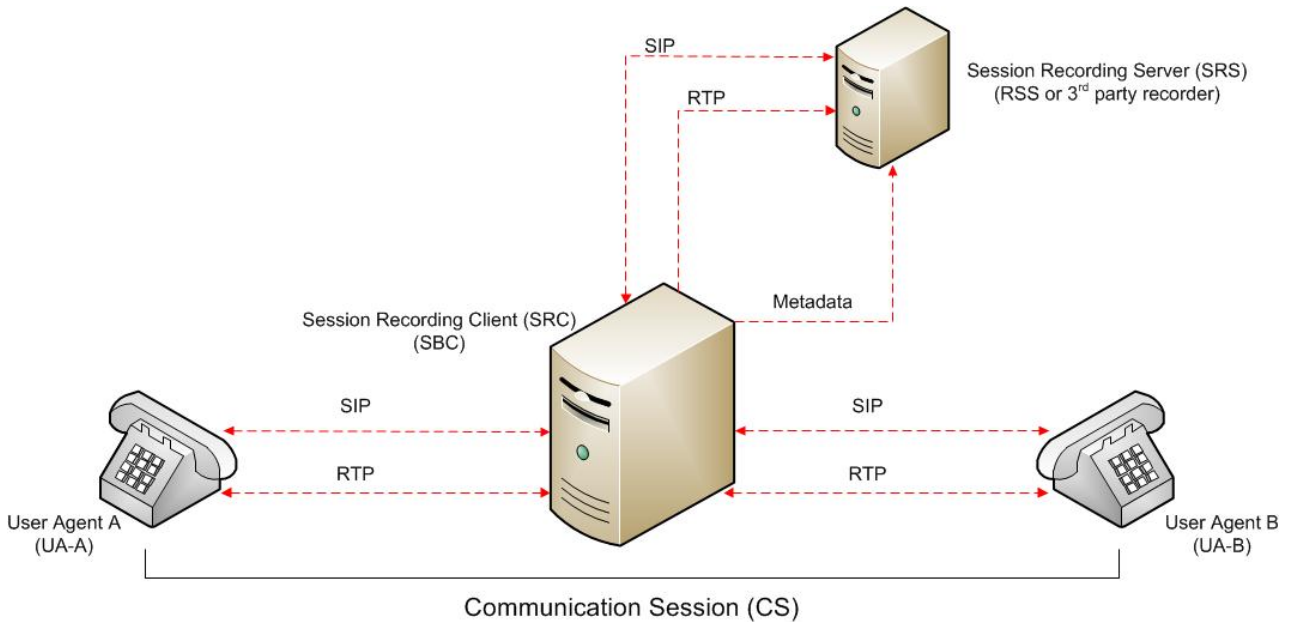
The Oracle Communications Session Border Controller acting as the SRC, is the source for the recorded media. The Oracle Communications Session Border Controller consumes configuration information describing the ecosystem within which it operates. The interface, realm and session agent configuration objects specify the SIPREC configuration. A SIP UA can elect to allow or disallow any network element from recording its media.

## Call Traffic Monitoring

During the establishment of a SIP Session, the Oracle Communications Session Border Controller determines if SIPREC is configured for recording the call. If so, it then duplicates the media prior to initiating the session with the SRS. (Media replication is set up prior to the recording session). The SRS may choose to record, not record, or cancel the recording session, and then communicates via SIP signaling to the Oracle Communications Session Border Controller. If the call is not to be recorded, the SRS signals termination of the recording session.

The Oracle Communications Session Border Controller maintains SIPREC metadata information associated with recording sessions. The recording session metadata describes the current state of the recording session and its communication session(s). It is updated when a change of state in the communication session(s) is observed by the Oracle Communications Session Border Controller. The SRS is responsible for maintaining call history, etc. The Oracle Communications Session Border Controller creates and logs call detail records (CDRs) in the current manner, the 3rd party SRS vendor may collate this information if desired. (For more information about the contents of metadata, see [Metadata Contents](#)).

The following illustration shows two endpoints, User Agent A (UA-A) and User Agent B (UA-B). Their session is being recorded by an SRC (the Oracle Communications Session Border Controller) and an SRS.



## Configuring SIPREC

This section defines the information required to configure SIPREC on the Oracle Communications Session Border Controller. It also provides a sample procedure for configuring SIPREC using the Acme Packet Command Line Interface (ACLI).

### Session Recording Server (SRS)

The Oracle Communications Interactive Session Recorder's RSS acts as the SRS in the network. A **session-recording-server** attribute under the **session-router** object in the Oracle Communications Session Border Controller ACLI allows you to enable/disable the SRS. This object is the session recording server that receives replicated media and records signaling. Additional parameters for SRS are configured under the **session-agent**, **realm-config**, and **sip-interface** objects. The rules of precedence for which the Oracle Communications Session Border Controller uses these parameters are:

**session-agent** takes precedence over the **realm-config**, and **realm-config** takes precedence over **sip-interface**.

Each SRS is associated with a **realm-config**. The realm specifies the source interface from which replicated traffic originates. The destination is an IP Port parameter (IP address or hostname with an optional port) that defines the SIP address (request URI) of the actual SRS.


For an additional level of security, Oracle recommends the SRS be configured in its own realm so as to apply a set of access control lists (ACLs) and security for the replicated communication.

Although the Oracle Communications Session Border Controller supports large UDP packets, Oracle recommends the **sip-interface** associated with the SRS realm, be provisioned with a TCP port.

## Session Recording Group

The Oracle Communications Session Border Controller uses the **session-recording-group** attribute under the **session-router** object in the ACLI to set high availability (HA) for 3rd party call recorders. Using this object, you can define a collection of one or more SRSs. The Oracle Communications Session Border Controller utilizes SIP's transport mechanism and keeps track of statistics on each SRS to manage the distribution of traffic and load balancing. (For more information on Oracle Communications Session Border Controller load balancing in session recording groups, see [Load Balancing](#)). When multiple SRSs are in a session recording group, the Oracle Communications Session Border Controller uses heuristics to intelligently route the recording dialog to one or more SRSs utilizing the selection strategy.


The **simultaneous-recording-servers** configuration attribute controls the number of simultaneous SIP dialogs that the Oracle Communications Session Border Controller establishes to the SRSs in the session recording group per communication session. For instance, if a session recording group contains 3 SRSs, and **simultaneous-recording-servers** is set to **2**, the recording agent initiates a SIP INVITE to the next two SRSs based on the session recording group strategy. In this way, duplicative recording sessions are instantiated, allowing for recording redundancy in multiple SRSs or within a session recording group.

 **Note:** The Oracle Communications Session Border Controller streams media to all SRSs. Each SRS chooses whether or not to ignore the media by returning a `recvonly(receive only)` media line. This permits an SRS to select specific media to record in the recording session, as well as determine whether or not to record the media.

The number of simultaneous recording servers does not dictate the number of recording devices required to be active for a communication session. If two SRSs exist in a session recording group and **simultaneous-recording-servers** is set to **2**, if at least one recording device to any of the servers completes, the recording server is treated as being established.

## Load Balancing

The Oracle Communications Session Border Controller supports recording server load balancing across members of a session recording group using the following strategies:

 **Note:** SRS groups support “round-robin” and “hunt” strategies only.

**[Round-robin]:** The Oracle Communications Session Border Controller remembers the last SRS that was used. Each new recording session selects the next SRS in the session recording group. When **simultaneous-recording-servers** is greater than 1, the next *n* recording servers are selected from the session recording group.

**[hunt]:** The Oracle Communications Session Border Controller successively attempts to contact SRSs in the session recording group until a successful recording dialog is established with the SRS, starting from the first SRS in the session recording group. The Oracle Communications Session Border Controller attempts to contact each SRS in the session reporting group once. When contact is exhausted, the recording device is considered failed. A SIP failure (response greater than 399, timeout or TCP setup failure) causes the Oracle Communications Session Border Controller to attempt the next possible SRS. When **simultaneous-recording-servers** is greater than 1, the Oracle Communications Session Border Controller attempts to establish *n* recording devices in a hunting fashion.

### Session Recording Group within Logical Remote Entities

Each logical remote entity (session-agent, realm-config and sip-interface) has a **session-recording-server attribute**. This attribute is a reference to a specific SRS configuration and can be used to specify a session recording group instead. If a session recording group is specified instead of an SRS, the session recording group name must be prefixed with "**SRG:**" followed by the session recording group name. This distinguishes between an SRS being referenced and a session recording group being referenced.

With SIPREC, if an SRS or session recording group is configured on both the ingress and egress logical remote entities, both the ingress and egress SRS/session recording groups are used. This means that the Oracle Communications Session Border Controller records the media between participants twice (or more) - once for the ingress recorders and once for the egress recorders.

If both the ingress and egress SRS/session recording group are the same, the Oracle Communications Session Border Controller makes an optimization and only records the media once. Even if the ingress session recording group is the same exact set of SRSs as the egress session recording group (but with a different name), the Oracle Communications Session Border Controller replicates media to both destinations. However, if the same set of SRSs has the exact same identifier, the

Oracle Communications Session Border Controller sends media to one and not both SRSs.

### Selective Recording

SIPREC defines a number of use cases for which the Oracle Communications Session Border Controller can record communication sessions. These use cases include the use of selective based recording. A **selective recording** is one in which a unique recording server is created per communication session.

 **Note:** The Oracle Communications Session Border Controller does not support persistent recording.

For SRSs using selective recording, recording servers are unique per session recording group. For each selective SRS in a session recording group, during the setup of a new communication session, the recording metadata is the same for each recording device. The SRC initiates a new SIP INVITE to the SRS carrying the metadata for that new recording server. The recording agent terminates the SIP dialog at the time that the recording session ends.

The lifetime of a recording session extends beyond the lifetime of the recorded communication. The SRC (Oracle Communications Session Border Controller) re-uses the recording session ID in the metadata instead of creating a new ID for each recording.

### High Availability (HA) Support

An Oracle Communications Session Border Controller using SIPREC supports HA in the network. The Oracle Communications Session Border Controller replicates all metadata states between the active and standby Oracle Communications Session Border Controllers. Any recording dialogs in progress do not survive the failover, but all calls in progress are preserved. Additionally, the recording dialogs are replicated as well to the failed over Oracle Communications Session Border Controller so that in-dialog SIP requests continue to function.

Each recorded communication session replicated to a single SRS counts as two calls instead of one. The Oracle Communications Session Border Controller creates two flows between the two participants and two additional flows to the SRS for each of the parent flows.

### SIPREC Configuration Procedure

The following configuration example assumes the Oracle Communications Session Border Controller has the session recording license enabled on the Oracle Communications Session Border Controller. Changes to the call session recording configuration for SIPREC are dynamic. Active calls in progress remain unaffected by the configuration changes. New calls, however, utilize the changes after a **Save** and **Activate** of the configuration.

The following attributes must be configured:

- session-recording-server

- **session-recording-group** (for RSS or 3rd party SRS high availability (HA) only)

and at least one of the following attributes:

- realm-config
- session-agent
- sip-interface

### Session-recording-server Attribute

To configure the session-recording-server attribute:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related objects.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-recording-server** and press Enter to access the session recording server-related attributes.

```
ACMEPACKET(session-router)# session-recording-server
ACMEPACKET(session-recording-server)#
```

4. **name** — Enter a unique name for the session recording server. This name can be referenced when configuring realm-config, session-agent, and sip-interface. Valid values are alpha-numeric characters. Default is no value specified.

```
ACMEPACKET(session-recording-server)# name SRS1
```

5. **(optional) description** — Enter a description for the session recording server. Valid values are alpha-numeric characters. Default is no value specified.

```
ACMEPACKET(session-recording-server)# description <recording server name>
```

6. **realm** — Enter the realm for which the session recording server belongs. Valid values are alpha-numeric characters. Default is no value specified.

```
ACMEPACKET(session-recording-server)# realm <realm name>
```



**Note:** Oracle recommends that the session recording server be configured in its own realm.

7. **mode** — Enter the recording mode for the session recording server. Valid values are:

- **selective** (default) - Unique recording server created per communication session
- **persistent** - Not supported.

```
ACMEPACKET(session-recording-server)# recording-mode selective
```

8. **destination** — Enter the destination IP address with IP port (port specification is optional) that defines the SIP address (request URI) of the session recording server. Enter values in the format 0.0.0.0:<port number>. Default is no value specified.

```
ACMEPACKET(session-recording-server)# destination 172.34.2.3:5060
```

9. **port** — Enter the port number to contact the session recording server. Valid values are 1024 to 65535. Default is 5060.

10. **transport-method** — Enter the protocol that the session recording server uses to accept incoming packets from the session reporting client on the network. Default is DynamicTCP. Valid values are:

- "" - No transport method used. Same as leaving this parameter value blank.
- UDP - User Datagram Protocol (UDP) is used for transport method.
- UDP+TCP - UDP and Transmission Control Protocol (TCP) are used for transport method.
- DynamicTCP - One TCP connection for EACH session is used for the transport method.
- StaticTCP - Only one TCP connection for ALL sessions is used for the transport method. This option saves resource allocation (such as ports) during session initiation.

## Call Traffic Monitoring

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- DynamicTLS - One Transport Layer Security (TLS) connection for EACH session is used for the transport method.
- StaticTLS - Only one TLS connection for ALL sessions is used for the transport method. This option saves resource allocation (such as ports) during session initiation.
- DTLS - Datagram TLS is used for the transport method.
- TLS+DTLS - TLS and DTLS are used for the transport method.
- StaticSCTP - Only one Stream Control Transmission Protocol (SCTP) connection for ALL sessions is used for the transport method. This option saves resource allocation (such as ports) during session initiation.

```
ACMEPACKET(session-recording-server)# protocol UDP
```

11. Enter **done** to save the session recording configuration.

```
ACMEPACKET(session-recording-server)# done
```

12. Enter **exit** to exit the session-recording-server configuration.

```
ACMEPACKET(session-recording-server)# exit
```

13. Enter **exit** to exit the session-router configuration.

```
ACMEPACKET(session-router)# exit
```

14. Enter **exit** to exit the configure mode.

```
ACMEPACKET(configure)# exit
```

15. Enter **save-config** to save the session recording configuration.

```
ACMEPACKET# save-config
```

16. Enter **activate-config** to activate the session recording configuration.

```
ACMEPACKET# activate-config
```

### Session-recording-group Attribute (for HA only)

For environments that required high availability (HA) requirements, configure the **session-recording-group** attribute.

To configure the session-recording-group attribute and enable HA:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related objects.

```
ACMEPACKET(configure)# session-router  
ACMEPACKET(session-router)#
```

3. Type **session-recording-group** and press Enter to access the session recording group-related attributes.

```
ACMEPACKET(session-router)# session-recording-group  
ACMEPACKET(session-recording-group)#
```

4. **name** — Enter a unique name for the session recording group that is a collection of one or more session recording servers. This name can be referenced when configuring realm-config, session-agent, and sip-interface. Valid values are alpha-numeric characters. Default is no value specified.

```
ACMEPACKET(session-recording-group)# name <SRG Group Name>
```

 **Note:** The name of the session recording group must be prefixed with SRG.


5. **(optional) description** — Enter a description for the session recording group. Valid values are alpha-numeric characters. Default is no value specified.

```
ACMEPACKET(session-recording-group)# description <Recording Group Name>
```

6. **session-recording-servers** — Enter the names of the session recording servers that belong to this session recording group. Valid values are alpha-numeric characters. Default is no value specified.

```
ACMEPACKET(session-recording-group)# session-recording-servers SRS1,SRS2
```



 **Note:** You must enter multiple servers as values for the session-recording-servers attribute.

7. **strategy** — Enter the load balancing strategy that the session reporting client (Oracle Communications Session Border Controller) uses when sending recordings to the session reporting server. Valid values are:

- **Round-robin** (default) - The Oracle Communications Session Border Controller remembers the last SRS that was used. Each new recording session selects the next SRS in the session recording group. When simultaneous-recording-servers is greater than 1, the next n recording servers are selected from the session recording group.
- **hunt** - The Oracle Communications Session Border Controller successively attempts to contact SRSs in the session recording group until a successful recording dialog is established with the SRS, starting from the first SRS in the session recording group. The Oracle Communications Session Border Controller attempts to contact each SRS in the session reporting group once. When contact is exhausted, the recording device is considered failed. A SIP failure (response greater than 399, timeout or TCP setup failure) causes the Oracle Communications Session Border Controller to attempt the next possible SRS. When simultaneous-recording-servers is greater than 1, the Oracle Communications Session Border Controller attempts to establish n recording devices in a hunting fashion.
- **least busy** - For some 3rd party recording devices, the number of concurrent recording servers proves to be the most taxing for system resources. The Oracle Communications Session Border Controller tracks the number of recording servers active to a given SRS at any given time. It uses this information to determine which SRS would be the best candidate for the next RS. The SRS with the fewest number of active recording servers receives the next RS. If two or more SRSs in a session recording group currently have the same number of active recording servers, the SRS configured first in the session recording group takes precedence.
- **lowest sustained rate** (fewest-setups-per-minute) - For some 3rd party recording servers, processing large amounts of sessions in a short amount of time proves to be the most taxing on their system's resources. The Oracle Communications Session Border Controller tracks the number of recording server setups over a sliding window of five minutes. The SRS within the session recording group with the fewest setups per the window of time is selected as the next candidate for receiving the recorded session. If two or more SRSs in a session recording group currently have the same value for setups in the given window of time, then the SRS configured first in the session recording group takes precedence.

```
ACMEPACKET(session-recording-group) # strategy round-robin
```

8. **simultaneous-recording-servers** — Enter the number of simultaneous SIP dialogs that the session reporting client (Oracle Communications Session Border Controller) establishes to the session reporting servers in the session reporting group per communication session. Valid values are 1 to 3. Default is 0.

```
ACMEPACKET(session-recording-group) # simultaneous-recording-servers 2
```

9. Enter **done** to save the session recording group configuration.

```
ACMEPACKET(session-recording-group) # done
```

10. Enter **exit** to exit the session recording group configuration.

```
ACMEPACKET(session-recording-group) # exit
```

11. Enter **exit** to exit the session-router configuration.

```
ACMEPACKET(session-router) # exit
```

12. Enter **exit** to exit the configure mode.

```
ACMEPACKET(configure) # exit
```

13. Enter **save-config** to save the session recording group configuration.

```
ACMEPACKET# save-config
```

14. Enter **activate-config** to activate the session recording group configuration.

```
ACMEPACKET# activate-config
```

### Realm-config Attribute

To configure the realm-config attribute and enable session recording:

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1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media manager-related objects.

```
ACMEPACKET (configure) # media-manager  
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter to access the realm-config-related attributes.


```
ACMEPACKET (media-manager) # realm-config  
ACMEPACKET (realm-config) #
```

4. **session-recording-server** — Enter the name of the session-recording server or the session-recording-group in the realm associated with the session reporting client (Oracle Communications Session Border Controller). Valid values are alpha-numeric characters. Default is no value specified.

```
ACMEPACKET (realm-config) # session-recording-server <srs-name>
```

or


```
ACMEPACKET (realm-config) # session-recording-server SRG:<group-name>
```

 **Note:** The value for this attribute is the name you specified in Step 4 of the *Session-recording-server Attribute* or Step 4 of the *Session-recording-group Attribute (for HA only)*. If specifying a session-recording-group, you must precede the group name with SRG:.

5. **session-recording-required** — Enter whether or not you want a call to be accepted by the Oracle Communications Session Border Controller if recording is not available. Valid values are:

- **Enabled** - Restricts call sessions from being initiated when a recording server is not available.
- **Disabled** (default)- Allows call sessions to initiate even if the recording server is not available.

```
ACMEPACKET (realm-config) # session-recording-required disabled
```

 **Note:** Oracle recommends that the session-recording-required parameter remain disabled.

6. Enter **done** to save the realm configuration.

```
ACMEPACKET (realm-config) # done
```

7. Enter **exit** to exit the realm configuration.

```
ACMEPACKET (realm-config) # exit
```

8. Enter **exit** to exit the media manager configuration.

```
ACMEPACKET (media-manager) # exit
```

9. Enter **exit** to exit the configure mode.

```
ACMEPACKET (configure) # exit
```

10. Enter **save-config** to save the realm configuration.

```
ACMEPACKET# save-config
```

11. Enter **activate-config** to activate the realm configuration.

```
ACMEPACKET# activate-config
```

### Session-agent Attribute

To configure the session-agent attribute and enable session recording:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related objects.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```



3. Type **session-agent** and press Enter to access the session agent-related attributes.


```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

4. **session-recording-server** — Enter the name of the session-recording server or the session-recording-group to apply to the session recording client (Oracle Communications Session Border Controller). Valid values are alpha-numeric characters. Default is no value specified.

```
ACMEPACKET(session-agent)# session-recording-server <srs-name>
```

or


```
ACMEPACKET(session-agent)# session-recording-server SRG:<group-name>
```

 **Note:** The value for this attribute is the name you specified in Step 4 of the [Session-recording-server Attribute](#) or Step 4 of the [Session-recording-group Attribute \(for HA only\)](#). If specifying a session-recording-group, you must precede the group name with **SRG**:

5. **session-recording-required** — Enter whether or not you want a call to be accepted by the Oracle Communications Session Border Controller if recording is not available. Valid values are:

- **Enabled** - Restricts call sessions from being initiated when a recording server is not available.
- **Disabled** (default)- Allows call sessions to initiate even if the recording server is not available.

```
ACMEPACKET(session-agent)# session-recording-required disabled
```

 **Note:** Oracle recommends that the session-recording-required parameter remain disabled.

6. Enter **exit** to exit the session agent configuration.

```
ACMEPACKET(session-agent)# exit
```

7. Enter **exit** to exit the session router configuration.

```
ACMEPACKET(session-router)# exit
```

8. Enter **exit** to exit the configure mode.

```
ACMEPACKET(configure)# exit
```

9. Enter **save-config** to save the session agent configuration.

```
ACMEPACKET# save-config
```

10. Enter **activate-config** to activate the session agent configuration.

```
ACMEPACKET# activate-config
```

## Sip-interface Attribute

To configure the sip-interface attribute and enable session recording:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related objects.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-interface** and press Enter to access the SIP interface-related attributes.


```
ACMEPACKET(session-router)# sip-interface
ACMEPACKET(sip-interface)#
```

4. **session-recording-server** — Enter the name of the session-recording server or the session-recording-group to apply to the SIP interface on the session recording client (Oracle Communications Session Border Controller). Valid values are alpha-numeric characters. Default is no value specified.

```
ACMEPACKET(sip-interface)# session-recording-server SRG:<session recording
server name or session-recording group name>
```

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---

 **Note:** The value for this attribute is the name you specified in Step 4 of the *Session-recording-server Attribute* or Step 4 of the *Session-recording-group Attribute (for HA only)*.

5. **session-recording-required** — Enter whether or not you want a call to be accepted by the Oracle Communications Session Border Controller if recording is not available. Valid values are:

- **Enabled** - Restricts call sessions from being initiated when a recording server is not available.
- **Disabled** (default)- Allows call sessions to initiate even if the recording server is not available.

```
ACMEPACKET(sip-interface) # session-recording-required disabled
```

 **Note:** Oracle recommends that the session-recording-required parameter remain disabled.

6. Enter **exit** to exit the SIP interface configuration.

```
ACMEPACKET(sip-interface) # exit
```

7. Enter **exit** to exit the session router configuration.

```
ACMEPACKET(session-router) # exit
```

8. Enter **exit** to exit the configure mode.

```
ACMEPACKET(configure) # exit
```

9. Enter **save-config** to save the SIP interface configuration.

```
ACMEPACKET# save-config
```

10. Enter **activate-config** to activate the SIP interface configuration.

```
ACMEPACKET# activate-config
```

## Metadata Contents

The recording metadata contains a set of related elements which define the recording session. A recording session may contain zero or more communication sessions and/or communication session groups. A communication session represents a call instance; a communication session group represents a related group of communication sessions. A recording session is composed of a sequence of complex element types. Not all element types are required to describe a recording session initiated from the Oracle Communications Session Border Controller. The recording session XML schema defines the following element types:

- **dataMode** - partial or complete metadata description (required)
- **group** - a collection of related communication sessions
- **session** - a single communication session of two or more participants (required)
- **participant** - a SIP endpoint representation (required)
- **stream** - a media stream
- **extensiondata** - application specific data outside of the SIPREC scope.

The recording agent generates dataMode, session, participant, and stream elements. Extension data is attached to other elements within the metadata through the use of the parent attribute. The recording metadata is defined as a sequence of element types; therefore all associations between elements are represented as references to element identifiers.

The state of the metadata within a recording session reflects the state of the communication session(s) which is being recorded. SIPREC implements stop-times and reason codes when communication sessions end within a recording session. Once a communication session, participant, or media stream has been marked as 'stopped' and accepted by the SRS, the metadata item is removed from the current metadata state. In addition, media lines within the SDP or the recording session may be re-used/re-labeled for reuse if new communication sessions and media streams are created within the recording session.

The XML schema for the recording metadata is defined in the IETF draft RFC *draft-ram-siprec-metadata-format-02* [7].

The ACLI command to show recorded metadata is **show rec**. For more information on this command see the section, *Show rec*.

## Show Commands for Recording Sessions

The Oracle Communications Session Border Controller allows you to utilize the following **show** commands via the ACLI to display statistical information about recording sessions:

- show rec
- show rec redundancy

### Show rec

The **show rec** command displays the count of all metadata objects in sessions managed by the recording agent. These statistics include metadata monitored over an active **period** of time and over a lifetime period (where lifetime totals reflect from the last reboot of the Oracle Communications Session Border Controller to the present time). The following example shows the use of this command.

1. Log into the Oracle Communications Session Border Controller as a User or Superuser.

```
ACMEPACKET> enable
ACMEPACKET(enable)#
```

2. Type **show rec** and press Enter to display the recording metadata statistics. The following output is an example of the show rec command.

```
ACMEPACKET(enable)# show rec
```

#### Show rec output

```
13:49:44-81645
Recording Agent Status      -- Period -- ----- Lifetime -----
                               Active      High   Total   Total   PerMax   High
Rec Sessions                0         1       1       1       1       1
Comm Groups                 0         0       0       0       0       0
Comm Sessions              0         1       1       1       1       1
Media Streams              0         2       2       2       2       2
Participants                0         2       2       2       2       2
```

The following table describes the metadata objects in the show rec command output.

Object	Description
Rec Sessions	Number of recording sessions during an active period of time and over a lifetime period.
Comm Groups	Number of active communication session recording groups during an active period of time and over a lifetime period.
Comm Sessions	Number of active communication sessions during an active period of time and over a lifetime period.
Media Streams	Number of active media streams during an active period of time and over a lifetime period.
Participants	Total number of participants in session recordings during an active period of time and over a lifetime period.

### Show rec redundancy

The **show rec redundancy** command displays information for session recording server statistics when the Oracle Communications Session Border Controller is configured for HA. These statistics include metadata monitored over an active period of time and over a lifetime period (where lifetime totals reflect from the last reboot of the Oracle Communications Session Border Controller to the present time) on both the primary and redundant Oracle Communications Session Border Controller. The following example shows the use of this command.

1. Log into the Oracle Communications Session Border Controller as a User or Superuser.

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```
ACMEPACKET> enable
ACMEPACKET(enable)#
```

2. Type **show rec redundancy** and press Enter to display the session recording server statistics for Oracle Communications Session Border Controllers in HA mode. The following output is an example of the show rec redundancy command.

```
ACMEPACKET(enable)# show rec redundancy
```

Show rec redundancy output

Primary System

```
13:49:44-81645
Recording Agent Status      -- Period -- ----- Lifetime -----
Active                    High  Total  Total  PerMax  High
Rec Sessions              0     1     1     1     1     1
Comm Groups               0     0     0     0     0     0
Comm Sessions             0     1     1     1     1     1
Media Streams             0     2     2     2     2     2
Participants              0     2     2     2     2     2
```

Redundant System

```
13:49:44-81646
Recording Agent Status      -- Period -- ----- Lifetime -----
Active                    High  Total  Total  PerMax  High
Rec Sessions              0     1     1     1     1     1
Comm Groups               0     0     0     0     0     0
Comm Sessions             0     1     1     1     1     1
Media Streams             0     2     2     2     2     2
Participants              0     2     2     2     2     2
```

The following table describes the session recording server statistics in the **show rec redundancy** command output.

Object	Description
Rec Sessions	Number of recording sessions during an active period of time and over a lifetime period.
Comm Groups	Number of active communication session recording groups during an active period of time and over a lifetime period.
Comm Sessions	Number of active communication sessions during an active period of time and over a lifetime period.
Media Streams	Number of active media streams during an active period of time and over a lifetime period.
Participants	Total number of participants in session recordings during an active period of time and over a lifetime period.

## SIPREC Recording Session Refresh

The Oracle Communications Session Border Controller provides for periodic in-dialog OPTIONS request/response exchanges to test the availability of the recording session dialog.

As shown below, and previously explained, establishment of a SIPREC session involves the creation of three distinct dialogs: a recording session dialog between the SIPREC client and SIPREC server, a SIP dialog between the calling party (A) and the SBC (acting as a B2BUA), and another SIP dialog between the SBC (acting as a B2BUA) and the called party (B). The two SIP dialogs are viewed conceptually as a single communications session dialog.

Both the call recording and the communication session dialogs are triggered by an INVITE at the SIPREC client. Upon reception, the recording client (the SBC) buffers the original INVITE, and sends a copy if it, along with call-

related meta-data to the recording server. A 200 OK response from the server establishes the recording session dialog that will carry replicated call content; signalling information is conveyed by the metadata in the recording dialog. After establishing the recording session, the client transmits the buffered INVITE to the called party to begin establishing the communications session dialog.

In the absence of a 200 OK response, the recording client is required to reject the offered INVITE with a 503 (Service Unavailable) SIP error code and an accompanying Reason header indicating that the recording session could not be established.

After establishing the recording session dialog, the client does not monitor its connection state. Its only opportunity to verify the state occurs after the termination of the communications session dialog, which triggers a client-originated BYE to the recording server. Reception of a 200 OK/ACK confirms the persistence of the recording session, while the lack of acknowledgement indicates the failure of the recording session at some unspecified point in time.

This failure to identify a prematurely terminated recording session has caused service providers and some regulatory agencies to require timely recognition of a failed call recording dialog.

Release S-CZ7.2.0, and later releases, address this requirement by providing for the exchange of periodic in-dialog OPTIONS request/response exchanges to test the availability of the recording session dialog.

### Timer\_B

Section 17.1.1.2 of RFC 3261, SIP: Session Initiation Protocol describes a TIMER\_B, the SIP INVITE transaction timer, which specifies the maximum interval between an INVITE request and response. The RFC suggests a default value of 32 seconds. Depending on the installed release version, two CLI commands, trans-expire and initial-inv-trans-expire, provide user control over the TIMER\_B value at both the SIP global level and the SIP interface level. Use these commands to adjust TIMER\_B values if required.

### OPTIONS Request/Response

Availability of the recording session is tested by an in-dialog SIP OPTIONS request/response exchange initiated by the call recording client and completed by the call recording server. If the OPTIONS exchange is enabled, two timers are set when the recording session dialog is established: a refresh-timer that specifies the interval between OPTIONS requests sent by the call recording client, and a response-timer that specifies the maximum interval between the OPTIONS request and the OPTIONS response. The response-timer is set to the smaller of the configured TIMER\_B or refresh-timer values.

Expiration of the refresh-timer results in the transmission of an OPTIONS request to the call recording server. The server, in turn, must reply with a 200 OK OPTIONS response prior to the expiration of the response-timer. In the event of a positive response, the call recording client restarts the refresh-timer, and re-issues the OPTIONS request when the timer next expires.

In the event that the call recording server response is non-positive, that is either

1. not received prior to the expiration of the response timer, or
2. a non-2xx OPTIONS response

the SBC terminates the session recording dialog and records the termination in the event log.

Typically, a SIPREC environment contains multiple recorders configured to record a communications session dialog, consequently the communications session should be terminated only when all recording dialogs in the call recording server have terminated. At that point the communications session dialog is torn down with a 503 (Service Unavailable) error report.

### Recording Session Refresh Configuration

Use the following procedure to enable a SIP OPTIONS request/response mechanism used to detect a prematurely terminated call recording session dialog.

1. Access the **session-recording-server** configuration element.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# session-router
```

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---

```
ACMEPACKET(session-router)# session-recording-server
ACMEPACKET(session-recording-server)#
```

2. Type **select** to choose and configure an existing object

```
ACMEPACKET(session-recording-server)# select
<recordingServerName>:1: name=SRS02 desc=SRS Atlanta
2: name=name desc="SRS First Server:
3: name=srs01 desc=Miami

selection: 2
```

3. **session-recording-required**—ensure that this parameter is enabled.

This is a prerequisite for the provision of recording session refresh services.

4. **refresh-interval**—use this parameter to both enable the SIP OPTIONS request/response mechanism, and to assign a value to the refresh-timer, which measures the maximum allowed interval (in seconds) between the OPTIONS request sent by the call-recording client and the OPTIONS response returned by the call-recording server.

By default, refresh-interval is set to 0, which disables detection of a failed recording session dialog.

Assignment of any non-zero value enables detection and sets the allowable interval between OPTIONS requests and responses.

Consult local policy and regulatory requirements when specifying the refresh-interval value.

```
ACMEPACKET(session-router)# refresh-interval 12
ACMEPACKET(session-recording-server)#
```

5. Type **done** to save your configuration.

## Codec Negotiation

---

In a SIPREC environment, it is assumed that the recording ecosystem provides transcoding media servers for which media calls can be redirected to, relieving the issue of codec matching from the recording servers. However, if transcoding media servers are not provided, the responsibility for transcoding falls on the recording server or the recording client in a SIPREC environment. The Oracle Communications Session Border Controller/SRC is required to impose some policy decisions on the codec negotiation between the three, or more, end-points. Specifically, the codec negotiation between the two participants and the recording server is subject to additional policy actions.

The SDP answer from the SRS may not agree with the media flows established in the communication session between UA-A and UA-B. If UA-A and UA-B agree to use G729, yet the SRS's answer indicates no support for G729, the SRS is then unable to interpret the media streams. The SDP offer forwarded to the called party (in this case UA-B) limits the codec choices to those supported by the SRS.



**Note:** The recording agent forwards the original codec offer to the SRS prior to sending the invite to the UA-B. The SRS responds with the SDP answer, indicating the codec list most desirable to the SRS. The codec list in the answer is then forwarded to UA-B. This allows three parties in a conference call to participate in the negotiation of the codecs among the supported formats only.

## SIPREC Call Flows

---

This section provides examples of call flow scenarios that can occur in a SIPREC environment. SIP recording call flow examples include:

For Selective Recording:

- *Normal Call (recording required)*
- *Normal Call (recording not required)*
- *Early Media Call (recording not required)*
- *REFER Pass-Through Call (REFER handled by User Agent)*
- *REFER Call (REFER handled by the Oracle Communications Session Border Controller)*
- *SRS Indicates Busy in Call (recording not required)*

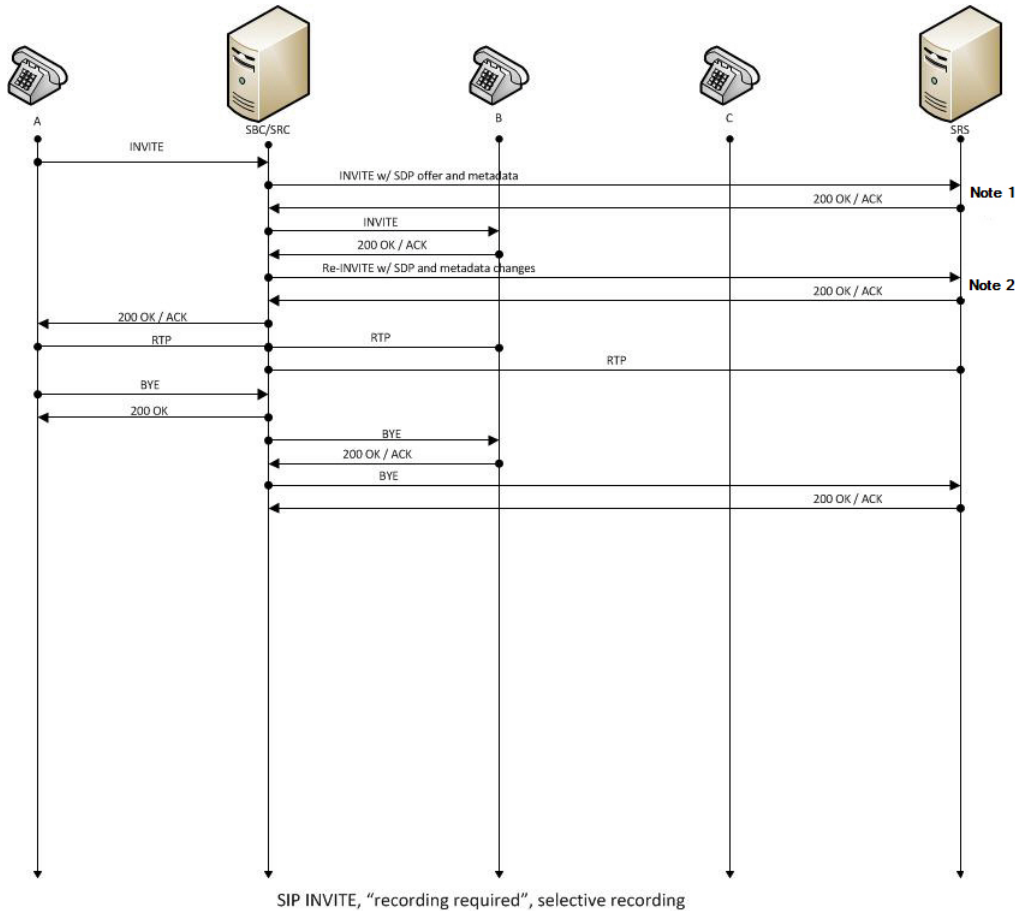


**Note:** REFER is a SIP method indicating that the recipient (identified by the Request-URI) should contact a third party using the contact information provided in the request.

## Selective Recording

### Normal Call (recording required)

The following illustration shows a normal call using selective recording with recording required. For SDP and Metadata information in Notes 1 and 2, see [Sample SDP and Metadata](#).



I

Call Flow Description	
① UA-A sends INVITE to Oracle Communications Session Border Controller.	⑩ RTP stream initiated between Oracle Communications Session Border Controller and UA-B.
② Oracle Communications Session Border Controller forwards INVITE with SDP and metadata to SRS.	⑪ RTP stream initiated between Oracle Communications Session Border Controller and SRS.
③ SRS responds with OK to Oracle Communications Session Border Controller.	⑫ UA-A sends BYE to Oracle Communications Session Border Controller.
④ Oracle Communications Session Border Controller sends INVITE to UA-B.	⑬ Oracle Communications Session Border Controller responds with OK to UA-A.
⑤ UA-B responds with OK to Oracle Communications Session Border Controller.	⑭ Oracle Communications Session Border Controller sends BYE to Oracle Communications Session Border Controller.

## Call Traffic Monitoring

Call Flow Description	
⑥ Oracle Communications Session Border Controller sends re-INVITE with SDP and metadata changes to SRS.	⑮ Oracle Communications Session Border Controller responds with OK to UA-A.
⑦ SRS responds with OK to Oracle Communications Session Border Controller.	⑯ Oracle Communications Session Border Controller sends BYE to UA-B.
⑧ Oracle Communications Session Border Controller forwards OK response to UA-A.	⑰ UA-B responds with OK to Oracle Communications Session Border Controller.
⑨ RTP stream initiated between UA-A and Oracle Communications Session Border Controller.	⑱ Oracle Communications Session Border Controller sends BYE to SRS.
	⑲ SRS responds with OK to Oracle Communications Session Border Controller.

### Sample SDP and Metadata

The following sample SDP and Metadata pertain to Notes 1 and 2 in the previous Call Flow diagram.

```
--[Note 1]-----
Content-Type: application/sdp
v=0
o=- 171 213 IN IP4 10.0.0.2
s=-
c=IN IP4 10.0.0.1
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=label:1

Content-Type: application/rs-metadata+xml
Content-Disposition: recording-session
<?xml version='1.0' encoding='UTF-8'?>
<recording xmlns='urn:ietf:params:xml:ns:recording'>
  <dataMode>complete</dataMode>
  <session id="urn:uuid:79b2fcd8-5c7f-455c-783f-db334e5d57d0">
    <start-time>2011-06-27T17:03:57</start-time>
  </session>
  <participant id="urn:uuid:10ac9063-76b7-40bb-4587-08ba290d7327"
session="urn:uuid:79b2fcd8-5c7f-455c-783f-db334e5d57d0">
    <aor>sip:sipp@168.192.24.40</aor>
    <name>sipp </name>
    <send>urn:uuid:07868c77-ef8e-4d6f-6dd5-a02ff53a1329</send>
    <start-time>2011-06-27T17:03:57</start-time>
  </participant>
  <participant id="urn:uuid:797c45f5-e765-4b12-52b0-d9be31138529"
session="urn:uuid:79b2fcd8-5c7f-455c-783f-db334e5d57d0">
    <aor>sip:service@168.192.24.60</aor>
    <name>sut </name>
  </participant>
  <stream id="urn:uuid:4a72aled-abb2-4d7c-5f4d-6d4c36e2d4ec"
session="urn:uuid:79b2fcd8-5c7f-455c-783f-db334e5d57d0">
    <mode>separate</mode>
    <start-time>2011-06-27T17:03:57</start-time>
    <label>1</label>
  </stream>
</recording>

--[Note 2]-----
Content-Type: application/sdp
```



```

v=0
o=- 171 213 IN IP4 10.0.0.2
s=-
c=IN IP4 10.0.0.1
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=label:1
m=audio 6002 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=label:2

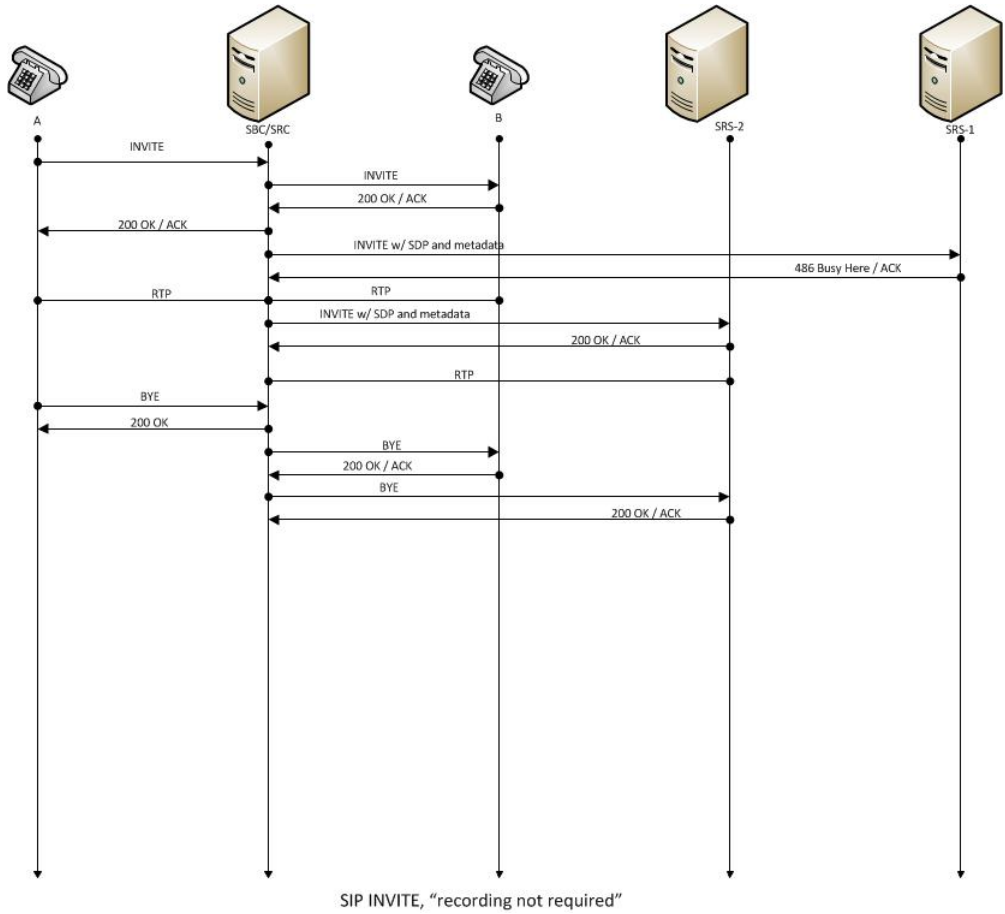
Content-Type: application/rs-metadata+xml
Content-Disposition: recording-session
<?xml version='1.0' encoding='UTF-8'?>
<recording xmlns='urn:ietf:params:xml:ns:recording'>
  <dataMode>partial</dataMode>
  <session id="urn:uuid:79b2fcd8-5c7f-455c-783f-db334e5d57d0">
    <start-time>2011-06-27T17:03:57</start-time>
  </session>
  <participant id="urn:uuid:797c45f5-e765-4b12-52b0-d9be31138529"
session="urn:uuid:79b2fcd8-5c7f-455c-783f-db334e5d57d0">
    <aor>sip:service@168.192.24.60</aor>
    <name>sut </name>
    <send>urn:uuid:4a72aled-abb2-4d7c-5f4d-6d4c36e2d4ec</send>
    <start-time>2011-06-27T17:03:58</start-time>
  </participant>
  <stream id="urn:uuid:07868c77-ef8e-4d6f-6dd5-a02ff53a1329"
session="urn:uuid:79b2fcd8-5c7f-455c-783f-db334e5d57d0">
    <mode>separate</mode>
    <start-time>2011-06-27T17:03:58</start-time>
    <label>2</label>
  </stream>
</recording>

```

### Normal Call (recording not required)

The following illustration shows a normal call using selective recording with recording optional.

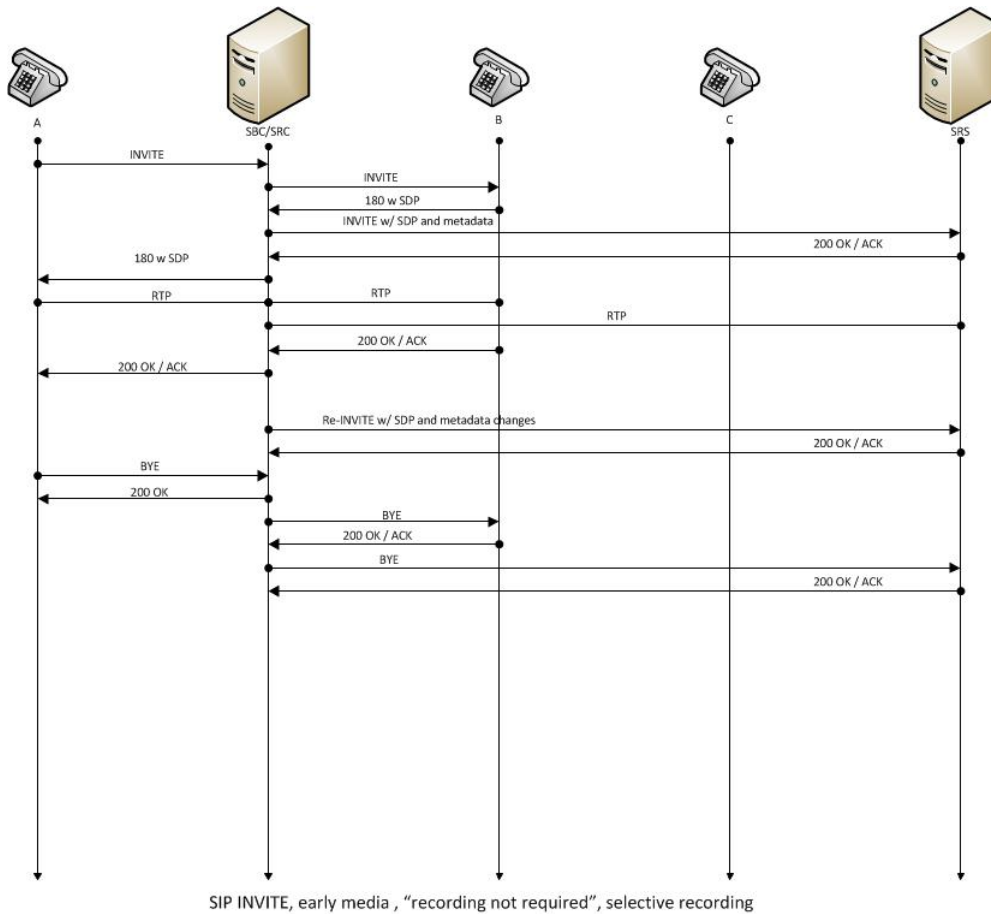
# Call Traffic Monitoring



Call Flow Description	
① UA-A sends INVITE to Oracle Communications Session Border Controller.	⑧ RTP stream initiated between Oracle Communications Session Border Controller and SRS.
② Oracle Communications Session Border Controller forwards INVITE to UA-B.	⑨ UA-A sends BYE to Oracle Communications Session Border Controller.
③ UA-B responds with OK to Oracle Communications Session Border Controller.	⑩ Oracle Communications Session Border Controller responds with OK to UA-A.
④ Oracle Communications Session Border Controller forwards OK response to UA-A.	⑪ Oracle Communications Session Border Controller sends BYE to UA-B.
⑤ Oracle Communications Session Border Controller sends INVITE with SDP and metadata to SRS.	⑫ UA-B responds with OK to Oracle Communications Session Border Controller.
⑥ SRS responds with OK to Oracle Communications Session Border Controller.	⑬ Oracle Communications Session Border Controller sends BYE to SRS.
⑦ RTP stream initiated between UA-A, Oracle Communications Session Border Controller, and UA-B.	⑭ SRS responds with OK to Oracle Communications Session Border Controller.

## Early Media Call (recording not required)

The following illustration shows an early media call using selective recording with recording optional.



Call Flow Description	
① UA-A sends INVITE to Oracle Communications Session Border Controller.	⑩ UA-B responds with OK to Oracle Communications Session Border Controller.
② Oracle Communications Session Border Controller forwards INVITE to UA-B.	⑪ Oracle Communications Session Border Controller forwards OK to UA-A.
③ UA-B sends 180 and SDP to Oracle Communications Session Border Controller.	⑫ Oracle Communications Session Border Controller sends re-INVITE with SDP and metadata changes to SRS.
④ Oracle Communications Session Border Controller sends INVITE with SDP and metadata to SRS.	⑬ SRS responds with OK to Oracle Communications Session Border Controller.
⑤ SRS responds with OK to Oracle Communications Session Border Controller.	⑭ UA-A sends BYE to Oracle Communications Session Border Controller.
⑥ Oracle Communications Session Border Controller sends 180 with SDP to UA-A.	⑮ Oracle Communications Session Border Controller responds with OK to UA-A.
⑦ RTP stream initiated between Oracle Communications Session Border Controller and UA-A.	⑯ Oracle Communications Session Border Controller sends BYE to UA-B.
⑧ RTP stream initiated between Oracle Communications Session Border Controller and UA-B.	⑰ UA-B responds with OK to Oracle Communications Session Border Controller.
⑨ RTP stream initiated between Oracle Communications Session Border Controller and SRS.	⑱ Oracle Communications Session Border Controller sends BYE to SRS.

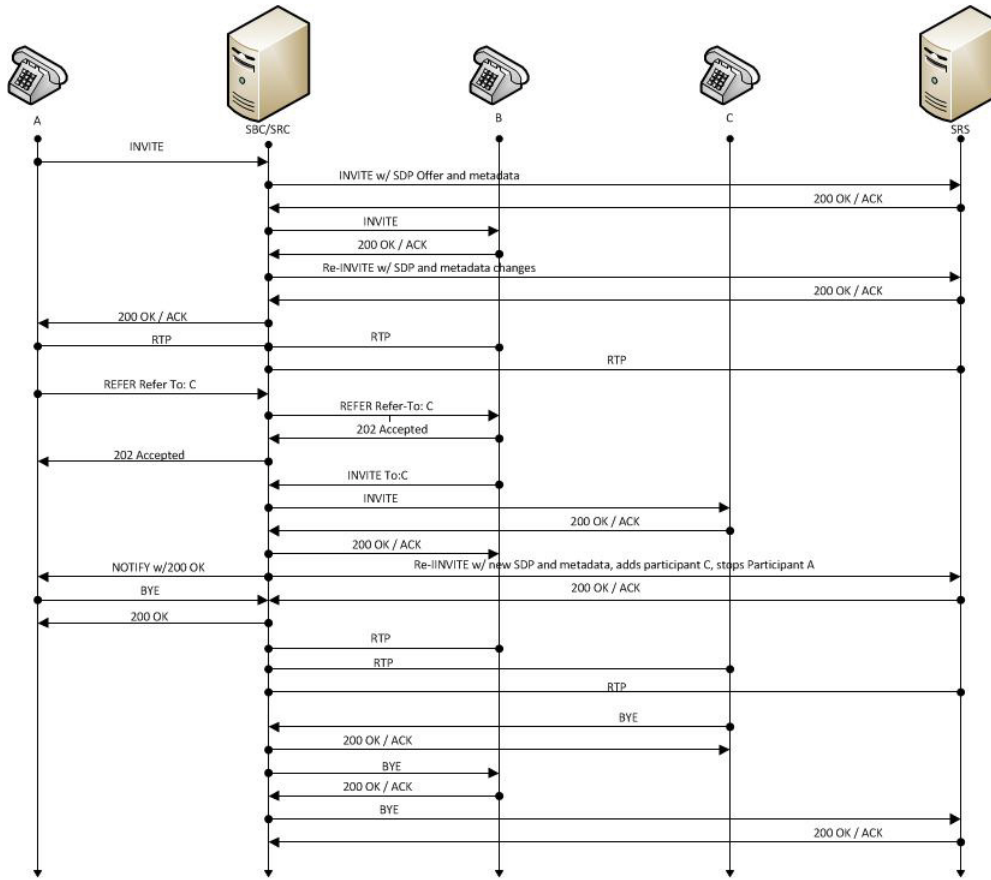
## Call Traffic Monitoring

### Call Flow Description

19 SRS responds with OK to Oracle Communications Session Border Controller.

### REFER Pass-Through Call (REFER handled by User Agent)

The following illustration shows a REFER pass-through call using selective recording and the User Agent (UA) handling the REFER on the call. Recording is required in this call flow.



SIP REFER, UA handles REFER, "recording required", selective recording

### Call Flow Description

1 - UA-A sends INVITE to Oracle Communications Session Border Controller.

18 - UA-C responds with OK to Oracle Communications Session Border Controller.

2 - Oracle Communications Session Border Controller forwards INVITE with SDP Offer and metadata to SRS.

19 - Oracle Communications Session Border Controller forwards OK response to UA-B.

3 - SRS responds with OK to Oracle Communications Session Border Controller.

20 - Oracle Communications Session Border Controller sends NOTIFY with OK response to UA-A.

4 - Oracle Communications Session Border Controller sends INVITE to UA-B.

21 - Oracle Communications Session Border Controller sends re-INVITE to SRS with new SDP and metadata, adds participant C, stops participant A .

5 - UA-B responds with OK to Oracle Communications Session Border Controller.

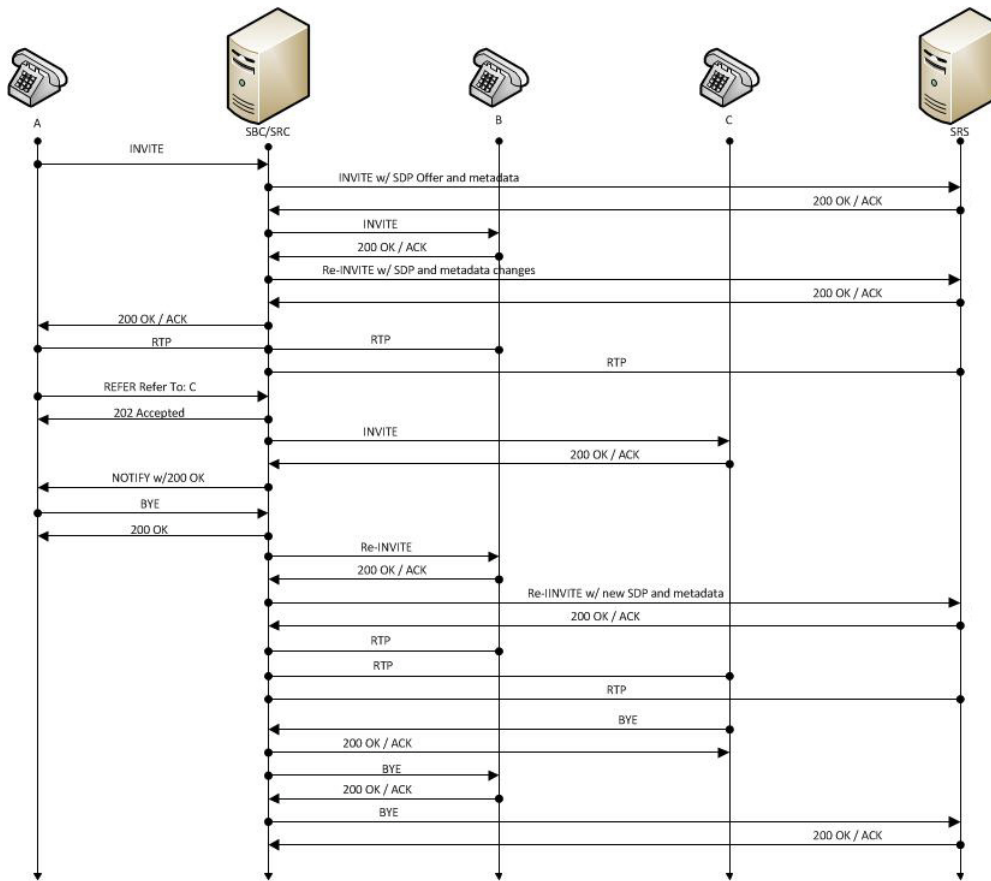
22 - SRS responds with OK to Oracle Communications Session Border Controller.

<b>Call Flow Description</b>	
6 - Oracle Communications Session Border Controller sends re-INVITE with SDP and metadata changes to SRS.	23 - UA-A sends BYE to Oracle Communications Session Border Controller.
7 - SRS responds with OK to Oracle Communications Session Border Controller.	24 - Oracle Communications Session Border Controller responds with OK to UA-A.
8 - Oracle Communications Session Border Controller forwards OK response to UA-A.	25 - Oracle Communications Session Border Controller responds with OK to UA-A.
9 - RTP stream initiated between UA-A and Oracle Communications Session Border Controller.	26 - RTP stream initiated between Oracle Communications Session Border Controller and UA-B.
10 - RTP stream initiated between Oracle Communications Session Border Controller and UA-B.	27 - RTP stream initiated between Oracle Communications Session Border Controller and UA-C.
11 - RTP stream initiated between Oracle Communications Session Border Controller and SRS.	28 - RTP stream initiated between Oracle Communications Session Border Controller and SRS.
12 - UA-A sends REFER-TO: C to Oracle Communications Session Border Controller.	29 - UA-C sends BYE to Oracle Communications Session Border Controller.
13 - Oracle Communications Session Border Controller forwards REFER-TO: C to UA-B.	30 - Oracle Communications Session Border Controller responds with OK to UA-C.
14 - UA-B responds with 202 ACCEPTED to Oracle Communications Session Border Controller.	31 - Oracle Communications Session Border Controller sends BYE to UA-B.
15 - Oracle Communications Session Border Controller forwards 202 ACCEPTED to UA-A.	32 - UA-B responds with OK to Oracle Communications Session Border Controller.
16 - UA-B sends INVITE TO: C to Oracle Communications Session Border Controller.	33 - Oracle Communications Session Border Controller sends BYE to SRS
17 - Oracle Communications Session Border Controller sends INVITE to UA-C.	34 - SRS responds with OK to Oracle Communications Session Border Controller.

**REFER Call (REFER handled by Oracle Communications Session Border Controller)**

The following illustration shows a call using selective recording and the Session Border Controller (Oracle Communications Session Border Controller) handling the REFER on the call. Recording is required in this call flow.

# Call Traffic Monitoring



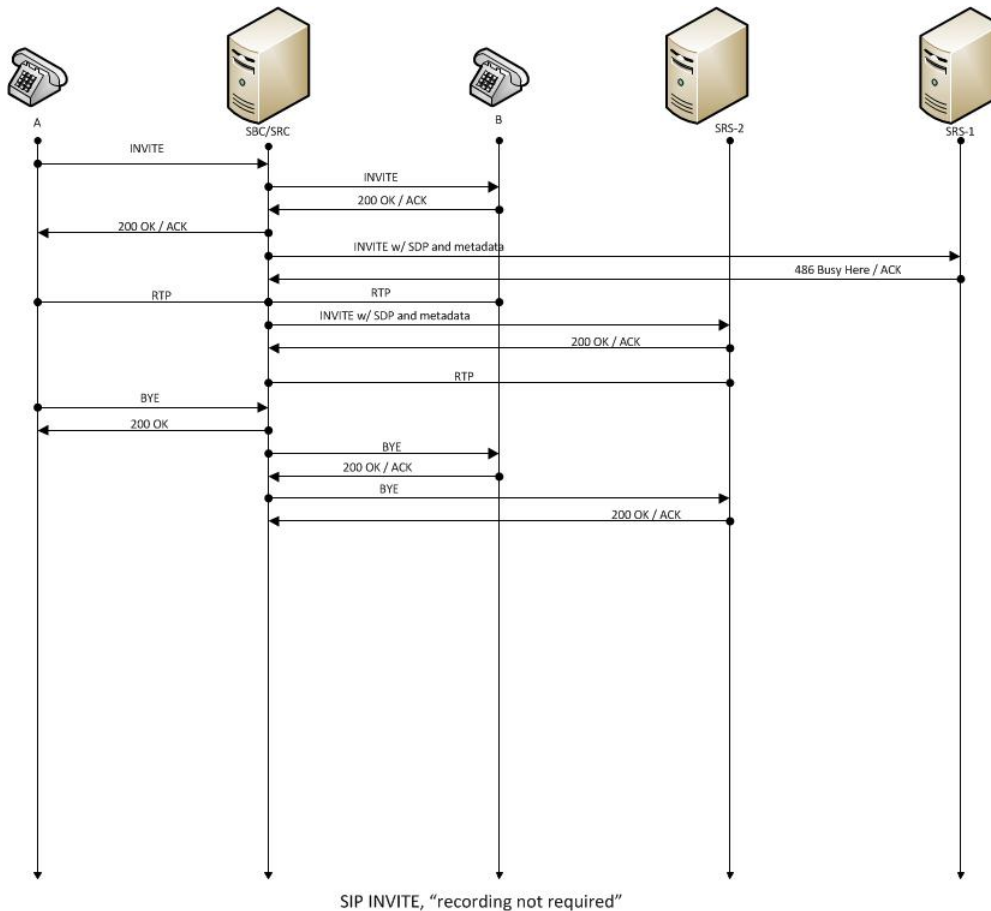
SIP REFER, SBC absorbs REFER, "recording required", selective recording

Call Flow Description	
1 - UA-A sends INVITE to Oracle Communications Session Border Controller.	16 - Oracle Communications Session Border Controller sends NOTIFY with OK response to UA-A.
2 - Oracle Communications Session Border Controller forwards INVITE with SDP Offer and metadata to SRS.	17 - UA-A sends BYE to Oracle Communications Session Border Controller.
3 - SRS responds with OK to Oracle Communications Session Border Controller.	18 - Oracle Communications Session Border Controller responds with OK to UA-A.
4 - Oracle Communications Session Border Controller sends INVITE to UA-B.	19 - Oracle Communications Session Border Controller sends re-INVITE to UA-B.
5 - UA-B responds with OK to Oracle Communications Session Border Controller.	20 - UA-B responds with OK to Oracle Communications Session Border Controller.
6 - Oracle Communications Session Border Controller sends re-INVITE with SDP and metadata changes to SRS.	21 - Oracle Communications Session Border Controller sends re-INVITE to SRS with new SDP and metadata.
7 - SRS responds with OK to Oracle Communications Session Border Controller.	22 - SRS responds with OK to Oracle Communications Session Border Controller.
8 - Oracle Communications Session Border Controller forwards OK response to UA-A.	23 - RTP stream initiated between Oracle Communications Session Border Controller and UA-B.
9 - RTP stream initiated between UA-A and Oracle Communications Session Border Controller.	24 - RTP stream initiated between Oracle Communications Session Border Controller and UA-C.

Call Flow Description	
10 - RTP stream initiated between Oracle Communications Session Border Controller and UA-B.	25 - RTP stream initiated between Oracle Communications Session Border Controller and SRS.
11 - RTP stream initiated between Oracle Communications Session Border Controller and SRS.	26 - UA-C sends BYE to Oracle Communications Session Border Controller.
12 - UA-A sends REFER-TO: C to Oracle Communications Session Border Controller.	27 - Oracle Communications Session Border Controller responds with OK to UA-C.
13 - Oracle Communications Session Border Controller Oracle Communications Session Border Controller responds with 202 ACCEPTED to UA-A.	28 - Oracle Communications Session Border Controller sends BYE to UA-B.
14 - Oracle Communications Session Border Controller sends INVITE to UA-C.	29 - UA-B responds with OK to Oracle Communications Session Border Controller.
15 - UA-C responds with OK to Oracle Communications Session Border Controller.	30 - Oracle Communications Session Border Controller sends BYE to SRS.
	31 - SRS responds with OK to Oracle Communications Session Border Controller.

**SRS Indicates Busy in Call (recording not required)**

The following illustration shows the Session Recording Server (SRS) is BUSY for a call session. Recording is not required in this call flow.



## Call Traffic Monitoring

Call Flow Description	
① UA-A sends INVITE to Oracle Communications Session Border Controller.	⑨ Oracle Communications Session Border Controller sends INVITE to SRS2 with SDP and metadata.
② Oracle Communications Session Border Controller forwards INVITE to UA-B.	⑩ SRS2 responds with OK to Oracle Communications Session Border Controller.
③ UA-B responds with OK to Oracle Communications Session Border Controller.	⑪ RTP stream initiated between Oracle Communications Session Border Controller and SRS2.
④ Oracle Communications Session Border Controller forwards OK response to UA-A.	⑫ UA-A sends BYE to Oracle Communications Session Border Controller.
⑤ Oracle Communications Session Border Controller sends INVITE to SRS1 with SDP and metadata.	⑬ Oracle Communications Session Border Controller responds with OK to UA-A.
⑥ SRS1 responds to Oracle Communications Session Border Controller with 436 BUSY HERE.	⑭ Oracle Communications Session Border Controller sends BYE to UA-B.
⑦ RTP stream initiated between UA-A and Oracle Communications Session Border Controller.	⑮ UA-B responds with OK to Oracle Communications Session Border Controller.
⑧ RTP stream initiated between Oracle Communications Session Border Controller and UA-B.	⑯ Oracle Communications Session Border Controller sends BYE to SRS2.
	⑰ SRS2 responds with OK to Oracle Communications Session Border Controller.

## Oracle Communications Session Monitor Mediation Engine

The Oracle® Communications Session Monitor Mediation Engine is a platform that collects SIP, DIAMETER, DNS and ENUM protocol message traffic received from Communications Session Monitor Probes. A Probe is software run on COTS hardware; it is deployed within a network and collects packets from span/monitor ports on Ethernet switches, or receives IP-in-IP tunneled packet-traces from Oracle's Oracle Communications Session Border Controller. A Probe takes the protocol packets, prepends a receive timestamp and other information, encapsulates the packets, and passes them to the Communications Session Monitor via a secure connection. After receiving protocol traffic from a Probe, the Communications Session Monitor stores the traffic in an internal database, and analyzes aggregated data to provide comprehensive multi-level monitoring, troubleshooting, and interoperability information.

In contrast to the Packet-Trace feature, message logging is performed by software, which sends a copy of sent/received messages over UDP, or by saving such messages in a local file. The copy includes a timestamp, port/vlan information, and IP:port information, but all in ASCII format. Message Logging is performed after all decryption, meaning that SIP/TLS traffic can be monitored. Because remote message logging sends the protocol messages over UDP, there is no guarantee or confirmation of delivery.

The Oracle Communications Session Border Controller provides support for Communications Monitoring, a user-configurable capability that enables the system to function as a Communications Session Monitor Probe. Acting as a Probe, or as an exporter, the Oracle Communications Session Border Controller can:

1. Establish an authenticated, persistent, reliable TCP connection between itself and one or more Communications Session Monitor Mediation Engines.
2. Optionally ensure message privacy by encrypting the TCP connection using TLS.
3. Use the TCP connection to send a UTC-timestamped, unencrypted copy of a protocol message to the Communications Session Monitor Engine(s).
4. Accompany the copied message with related data to include: the port/vlan on which the message was sent/received, local and remote IP:port information, and the transport layer protocol.



## IPFIX

The Oracle Communications Session Border Controller uses the IPFIX suite of standards to export protocol message traffic and related data to the Communications Session Monitor Mediation Engine.

- RFC 5101, *Specification of the IP Flow Information Export (IPFIX) Protocol for the Exchange of IP Traffic Flow Information*
- RFC 5102, *Information Model for IP Flow Information Export*
- RFC 5470, *Architecture for IP Flow Information Export*
- RFC 5655, *Specification of the IP Flow Information Export (IPFIX) File Format*
- RFC 5815, *Definitions of Managed Objects for IP Flow Information Export*

The IPFIX standards describe the use of templates to format the export of specific types of protocol traffic. The Oracle Communications Session Border Controller and the Communications Session Monitor Mediation Engine share ten (10) pre-defined templates that facilitate protocol message exchange, and subsequent processing and analysis by the Communications Session Monitor Engine.

The pre-defined templates are:

- incoming SIP/DNS over UDP
- incoming SIP over TCP
- incoming SIP over SCTP
- incoming DNS over UDP (entire IP and UDP header not included)
- outgoing SIP/DNS over UDP
- outgoing SIP over TCP
- outgoing SIP over SCTP
- outgoing DNS over UDP (entire IP and UDP header not included)
- media qos and flow record
- IPFIX handshake (used for connection establishment)

## Communications Monitor Configuration

Communications Monitor configuration consists of the following steps.

1. Configuration of one or more Oracle Communications Session Border Controller/Palladion exporter/collector pairs.

Configuration of the -config object, which defines common operations across all interfaces, is not required. Default values can be retained to enable standard service.

2. Optional assignment of a TLS profile to an exporter/collector pair.

## Communication Monitor

Use the following procedure to configure communication monitoring:

1. From superuser mode, use the following CLI sequence to access comm-monitor configuration mode. From comm-monitor mode, you establish a connection between the Oracle Communications Session Border Controller, acting as a exporter of protocol message traffic and related data, and an Oracle® Communications Session Monitor Mediation Engine, acting as an information collector.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# system
ACMEPACKET(system)# system-config
ACMEPACKET(system-config)# comm-monitor
ACMEPACKET(comm-monitor)#
```

2. Use the **state** parameter to enable or disable communication monitoring.

## Call Traffic Monitoring

---

Communication monitoring is disabled by default.

```
ACMEPACKET(comm-monitor) # state enabled
ACMEPACKET(comm-monitor) #
```

3. Use the **sbc-group-id** parameter to assign an integer value to the Oracle Communications Session Border Controller, in its role as an information exporter.

Retain the default value (0) or assign another integer value.

```
ACMEPACKET(comm-monitor) # sbc-group-id 5
ACMEPACKET(comm-monitor) #
```

4. If the network interface specified in Step 8 is a media interface, you can optionally use TLS to encrypt the exporter/collector connection.

To enable TLS encryption, use the **tls-profile** parameter to identify a TLS profile to be assigned to the network interface. The absence of an assigned TLS profile (the default state) results in unencrypted transmission.

Refer to [TLS Profile Configuration](#) for configuration details.

```
ACMEPACKET(comm-monitor) # tls-profile commMonitor
ACMEPACKET(comm-monitor) #
```

5. Use the **qos-enable** parameter to enable or disable to export of RTP, SRTP, and QOS data flow information.

```
ACMEPACKET(comm-monitor) # qos-enable enabled
ACMEPACKET(comm-monitor) #
```

6. Use the **monitor-collector** parameter to move to monitor-collector configuration mode.

While in this mode you identify a Communications Session Monitor Mediation Engine collector by IP address and port number.

```
ACMEPACKET(comm-monitor) # monitor-collector
ACMEPACKET(monitor-collector) #
```

7. Use the **address** and **port** parameters to specify the IP address and port number monitored by a Communications Session Monitor Mediation Engine for incoming IPFIX traffic.

Enter an IPv4 address and a port number with the range 1025 through 65535, with a default value of 4739.

```
ACMEPACKET(monitor-collector) # address 172.30.101.239
ACMEPACKET(monitor-collector) # port 4729
ACMEPACKET(monitor-collector) #
```


8. Use the **network-interface** parameter to specify the network interface that supports the TCP connection between the Oracle Communications Session Border Controller to the Communications Session Monitor Mediation Engine.

To specify the wancom0 management interface:

```
ACMEPACKET(comm-monitor) # network-interface wancom0:0
ACMEPACKET(comm-monitor) #
```

To specify a media interface:

```
ACMEPACKET(comm-monitor) # network-interface m01
ACMEPACKET(comm-monitor) #
```

 **Note:** If configuring with a media interface, that interface must belong to a configured realm.

9. Use **done** and **exit** to return to comm-monitor configuration mode.
10. Use **done**, **exit**, and **verify-config** to complete configuration.
11. Repeat Steps 1 through 10 to configure additional as required.

## TSCF Rekey Profile Configuration

Rekeying is a cryptographic technique that enhances security by enforcing the negotiation of existing keys on an ongoing secure connection. Rekeying can be either time-based, in which case new keys are negotiated at the expiration of a timer, or traffic-based, in which case new keys are negotiated when a threshold byte count is exceeded.

Use the following procedure to configure an optional tscf-rekey-profile. Later, you will assign the profile to a specific TSCF interface. If you do not intend to enforce re-keying, this procedure can be safely ignored.

1. From superuser mode, use the following command sequence to access tscf-rekey-profile configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# tscf
ACMEPACKET(tscf)# tscf-rekey-profile
ACMEPACKET(tscf-rekey-profile)#
```

2. Use the **name** parameter to provide a unique identifier for this tscf-rekey-profile.

```
ACMEPACKET(tscf-rekey-profile)# name tscfRekey01
ACMEPACKET(tscf-rekey-profile)#
```

3. Use the **initiator** parameter to identify the rekey initiator.

Supported values are **client** (default) | **server** (the Session Director)

```
ACMEPACKET(tscf-rekey-profile)# initiator client
ACMEPACKET(tscf-rekey-profile)#
```

4. Use the **max-rekey-time** parameter to specify the maximum interval (in minutes) between re-keying operations.

Supported values are **0** (default) | **30 - 1440** (minutes)

The default value, **0**, specifies that time-based rekeying is not enforced; other integer values specify that time-based re-keying must be initiated by the tunnel endpoint designated by the **initiator** parameter.

```
ACMEPACKET(tscf-rekey-profile)# max-rekey-time 30
ACMEPACKET(tscf-rekey-profile)#
```

5. Use the **max-rekey-data** parameter to specify the maximum traffic exchange (measured in Kb) between rekeying operations.

The default value, **0**, specifies that traffic-based rekeying is not enforced; other integer values specify that traffic-based re-keying must be initiated by the tunnel endpoint designated by the **initiator** parameter.

```
ACMEPACKET(tscf-rekey-profile)# max-rekey-data 0
ACMEPACKET(tscf-rekey-profile)#
```

6. Use **done**, **exit**, and **verify-config** to complete tscf-rekey-profile configuration.
7. Repeat Steps 1 through 6 to configure additional tscf-rekey-profiles as required.

## TLS Profile Configuration

Use the following procedure to configure a tls-profile that identifies the cryptographic resources, specifically certificates and protocols, required for the establishment of a secure/encrypted connection between the Oracle Communications Session Border Controller and the Communications Session Monitor Mediation Engine.

1. From superuser mode, use the following command sequence to access tls-profile configuration mode.

```
ACMEPACKET# configure terminal
ACMEPACKET(configure)# security
ACMEPACKET(security)# tls-profile
ACMEPACKET(tls-profile)#
```

2. Use the **name** parameter to provide a unique identifier for this tls-profile.

```
ACMEPACKET(tls-profile)# name commMonitor
ACMEPACKET(tls-profile)#
```

3. Use the required **end-entity-certificate** parameter to specify the name of the certificate-record configuration that identifies the credential (specifically, an X509.v3 certificate) offered by the Oracle Communications Session Border Controller in support of its asserted identity.

```
ACMEPACKET(tls-profile) # end-entity-certificate commMonitor509  
ACMEPACKET(tls-profile) #
```

4. Use the required **trusted-ca-certificates** parameter to compile a list or one or more certificate-record configuration elements referencing trusted Certification Authority (CA) certificates used to authenticate the offered certificate. These referenced certificates are conveyed to the Communications Session Monitor Mediation Engine as part of the TLS exchange.

Provide a comma separated list of existing CA **certificate-record** configuration elements.

```
ACMEPACKET(tls-profile) # trusted-ca-certificates verisignClass3-  
a,verisignClass3-b,baltimore,thawtePremium,acme-CA  
ACMEPACKET(tls-profile) #
```

5. Retain the default value, **all**, for the **cipher-list** parameter.
6. Use the **verify-depth** parameter to specify the maximum number of chained certificates that will be processed while authenticating end-entity certificate received from the Communications Session Monitor Mediation Engine.

Provide an integer within the range 1 through 10 (the default).

The Oracle Communications Session Border Controller supports the processing of certificate chains (consisting of an end-entity certificate and some number of CA certificates) when X.509v3 certificate-based authentication is used. The following process validates a received TLS certificate chain.

- Check the validity dates (Not Before and Not After fields) of the end certificate. If either date is invalid, authentication fails; otherwise, continue chain validation
- Check the maximum length of the certificate chain (specified by **verify-depth**). If the current chain exceeds this value, authentication fails; otherwise, continue chain validation.
- Verify that the Issuer field of the current certificate is identical to the Subject field of the next certificate in the chain. If values are not identical, authentication fails; otherwise, continue chain validation.
- Check the validity dates (Not Before and Not After fields) of the next certificate. If either date is invalid, authentication fails; otherwise, continue chain validation.
- Check the X509v3 Extensions field to verify that the current certificate identifies a CA. If not so, authentication fails; otherwise, continue chain validation.
- Extract the Public Key from the current CA certificate. Use it to decode the Signature field of the prior certificate in the chain. The decoded Signature field yields an MD5 hash value for the contents of that certificate (minus the Signature field).
- Compute the same MD5 hash. If the results are not identical, authentication fails; otherwise, continue chain validation.
- If the hashes are identical, determine if the CA identified by the current certificate is a trust anchor by referring to the **trusted-ca-certificates** attribute of the associated TLS-profile configuration object. If the CA is trusted, authentication succeeds. If not, return to Step 2.

```
ACMEPACKET(tls-profile) # verify-depth 8  
ACMEPACKET(tls-profile) #
```

7. Use the **mutual-authenticate** parameter to **enable** or **disable** (the default) mutual authentication.

Protocol requirements mandate that the server present its certificate to the client application. Optionally, the server can implement mutual authentication by requesting a certificate from the client application, and authenticating the certificate offered by the client.

Upon receiving a server certificate request, the client application must respond with a certificate; failure to do so results in authentication failure.

```
ACMEPACKET(tls-profile) # mutual-authenticate disabled  
ACMEPACKET(tls-profile) #
```

8. Retain the default value, **compatibility**, for the **tls-version** parameter.

9. Retain default values for all other parameters.
10. Use **done**, **exit**, and **verify-config** to complete `tls-profile` configuration.
11. Repeat Steps 1 through 10 to configure additional `tls-profiles` as required.

## Packet Trace

---

The Oracle Communications Session Border Controller's packet trace tool provides the user with the ability to capture traffic from the Oracle Communications Session Border Controller itself.

The user invokes the tool from the CLI, manually specifying:

- How to capture (local vs remote)
- What to capture
- Capture start and stop

There are two capture modes, one that saves traffic locally and one that mirrors traffic to a user-specified target. Software only deployments support local capture only. Proprietary Acme hardware deployments support both local and remote capture.

Local capture supports PCAP filters to specify the type of traffic to capture. Remote capture supports its own syntax to identify the traffic to mirror.

Local packet capture is dependent on access control configuration, not capturing any denied traffic. Remote capture mirrors traffic regardless of access control configuration.

Installed NIUs impact remote packet capture. Fragmented packets that ingress HIFNs or Cavium NIUs include only the outer header within the fragments. As a result, these traces do not appear to be using IPIP encapsulation. This differs from fragmented packets that ingress the Quad port GiGE and Copper PHY NIUs. These traces include inner and outer headers within the fragments.

Do not run `packet-trace` simultaneously with other Oracle Communications Session Border Controller replication features, such as LI, SRS, SIP Monitoring and Trace, and Call Recording. These features may interfere with each other, corrupting each's results.

## Packet Trace Remote

Packet trace remote enables the Oracle Communications Session Border Controller to mirror traffic between two endpoints, or between itself and a specific endpoint to a user-specified target. To accomplish this, the Oracle Communications Session Border Controller replicates the packets sent and received, encapsulates them according to RFC 2003, and sends them to a user-configured target. At the target, the user would capture and analyze the packets.

Currently, the Oracle Communications Session Border Controller supports:

- One configurable trace server (on which you capture/analyze the traffic)
- Sixteen concurrent endpoint traces

To use this feature, the user configures a **capture-receiver** on the Oracle Communications Session Border Controller so that it knows where to send the mirrored packets. Once the **capture-receiver** is configured, the user issues the **packet-trace** command to start, stop and specify filters for traces.

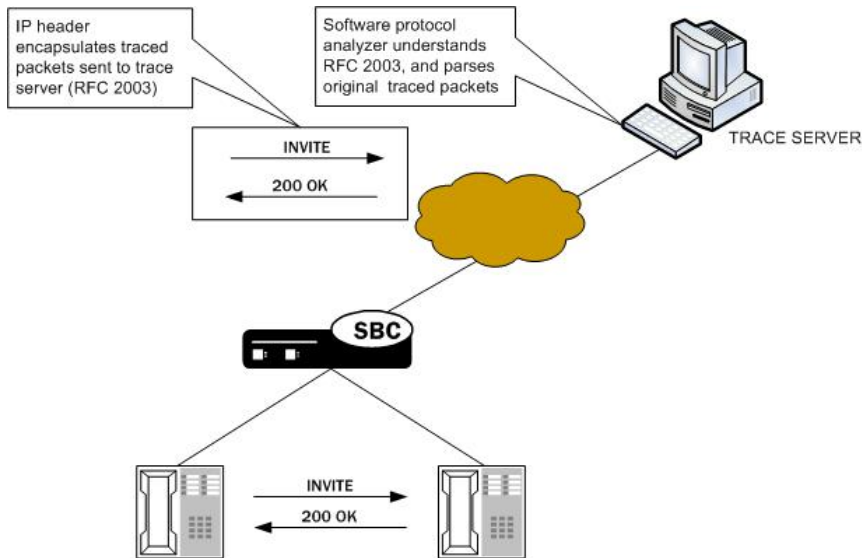
You establish a packet trace filter with the following information:

- Network interface—The name of the network interface on the Oracle Communications Session Border Controller from which you want to trace packets. The user can enter this value as a name or as a name and subport identifier value (name:subportid)
- IP address—IP address of the endpoint to or from which the target traffic goes.
- Local port number—Optional parameter; Layer 4 port number on which the Oracle Communications Session Border Controller receives and from which it sends; if no port is specified or if it is set to 0, then all ports will be traced

## Call Traffic Monitoring

- Remote port number—Optional parameter; Layer 4 port number to which the Oracle Communications Session Border Controller sends and from which it receives; if no port is specified or if it is set to 0, then all ports will be traced.

The Oracle Communications Session Border Controller then encapsulates the original packets in accordance with RFC 2003 (*IP Encapsulation within IP*); it adds the requisite headers, and the payload contains the original packet trace with the Layer 2 header removed. Since software protocol analyzers understand RFC 2003, they can easily parse the original traced packets.



It is possible that—for large frames—when the Oracle Communications Session Border Controller performs the steps to comply with RFC 2003 by adding the requisite header, the resulting packet might exceed Ethernet maximum transmission unit (MTU). This could result in packets being dropped by external network devices, but widespread support for jumbo frames should mitigate this possibility.

If the Oracle Communications Session Border Controller either receives or transmits IP fragments during a packet trace, it only traces the first fragment. The first fragment is likely to be a maximum-sized Ethernet frame.

The Oracle Communications Session Border Controller continues to conduct the packet trace and send the replicated information to the trace server until you instruct it to stop. You stop a packet trace with the CLI **packet-trace remote stop** command. With this command, you can stop either an individual packet trace or all packet traces that the Oracle Communications Session Border Controller is currently conducting.

## Packet Trace Local

Packet Trace Local enables the Oracle Communications Session Border Controller to capture traffic between two endpoints, or between itself and a specific endpoint. To accomplish this, the Oracle Communications Session Border Controller replicates the packets sent and received and saves them to disk in .PCAP format.

The default packet trace filter uses the specified interface to capture both ingress and egress traffic. To specify captured traffic, the user can append the command with a PCAP filter enclosed in quotes. PCAP filter syntax is widely published.

While capturing, the system displays the number of packets it has captured and prevents the user from entering any other CLI commands from that session. The user terminates the capture by pressing Ctrl+C.

By default, the system saves the .PCAP file in /opt/traces, naming it with the applicable interface name as well as the date and time of the capture. Alternatively, the user can specify file name using the system supports the PCAP filter flags -w.

The system rotates the PCAPs created in this directory by size. The last 25 files are kept and are rotated when they reach 100 MB. If there are capture files in the /opt/traces directory when this command is run, the system prompts the user to remove them before running new captures. If preferred, the user can decline this file deletion.

## Packet Trace Scenarios

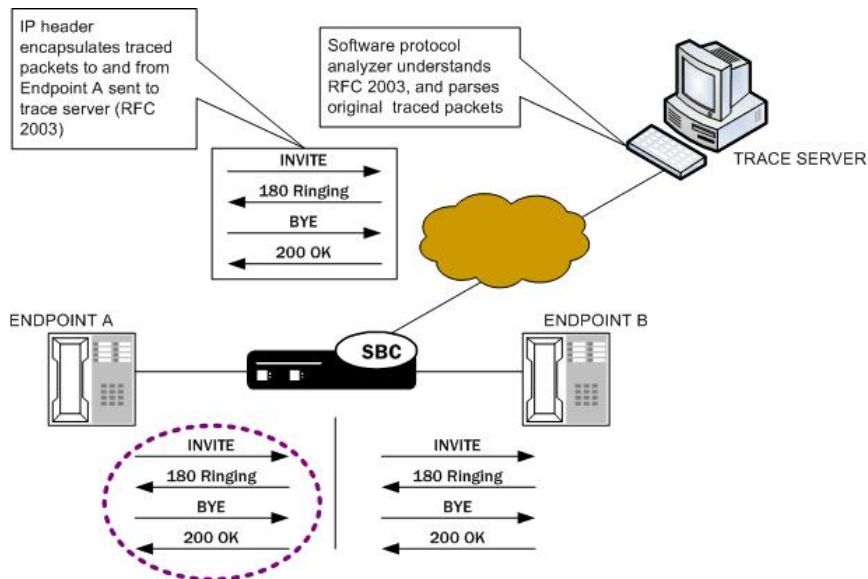
This section describes three possible ways that you might use the packet trace feature. You can examine communications sent to and from one endpoint, sent between two endpoints, or sent between ingress and/or egress Oracle Communications Session Border Controller interfaces to endpoints.

### Packet Trace for One Endpoint

When you use the `packet-trace remote <state>` command, the Oracle Communications Session Border Controller sets up packet tracing for one endpoint. The Oracle Communications Session Border Controller collects and replicates the packets to and from one endpoint. To enable this kind of trace, you set up one packet trace using the `packet-trace` command.

The commands you carry out for `packet-trace remote` would take the following form:

```
ACMEPACKET# packet-trace remote start F01:0 <IP address of Endpoint A>
```



The commands you carry out for `packet-trace local` would take the following form:

```
ACMEPACKET# packet-trace local F01:0 <"host IP address of Endpoint A">
```

### Packet Trace for Both Call Legs

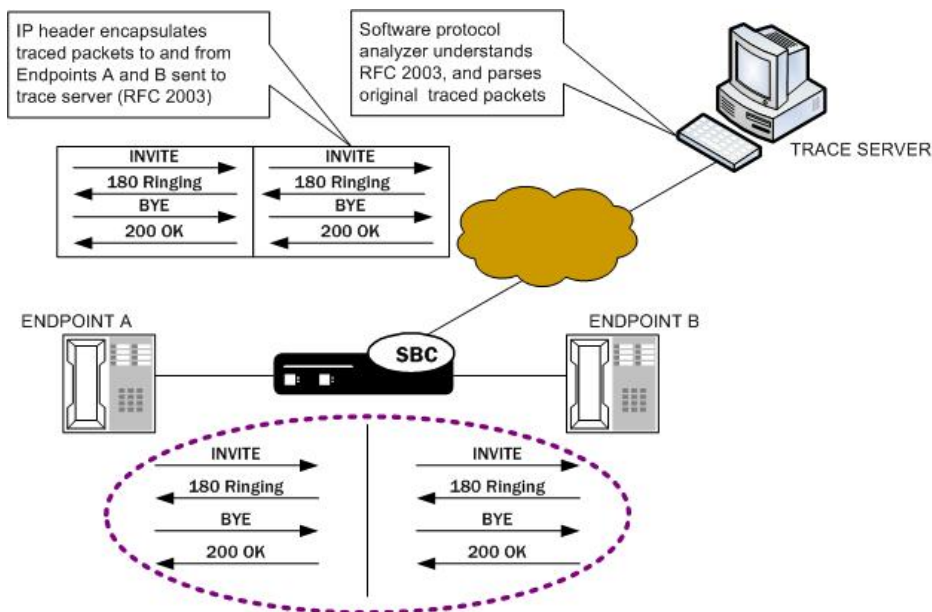
If you want to trace both sides (both call legs), then you must set up individual traces for each endpoint—meaning that you would initiate two packet traces. The results of the trace will give you the communications both call legs for the communication exchanged between the endpoints you specify.

If you initiate a packet trace for both endpoints that captures both signaling and media, the signaling will be captured as usual. However, RTP will only be traced for the ingress call leg. This is because the Oracle Communications Session Border Controller performs NAT on the RTP, which means it cannot be captured on the egress call leg.

The commands you carry out for `packet-trace remote` would take the following form:

```
ACMEPACKET# packet-trace remote start F01:0 <IP address of Endpoint A>
ACMEPACKET# packet-trace remote start F02:0 <IP address of Endpoint B>
```

## Call Traffic Monitoring



The commands you carry out for **packet-trace local** would take the following form:

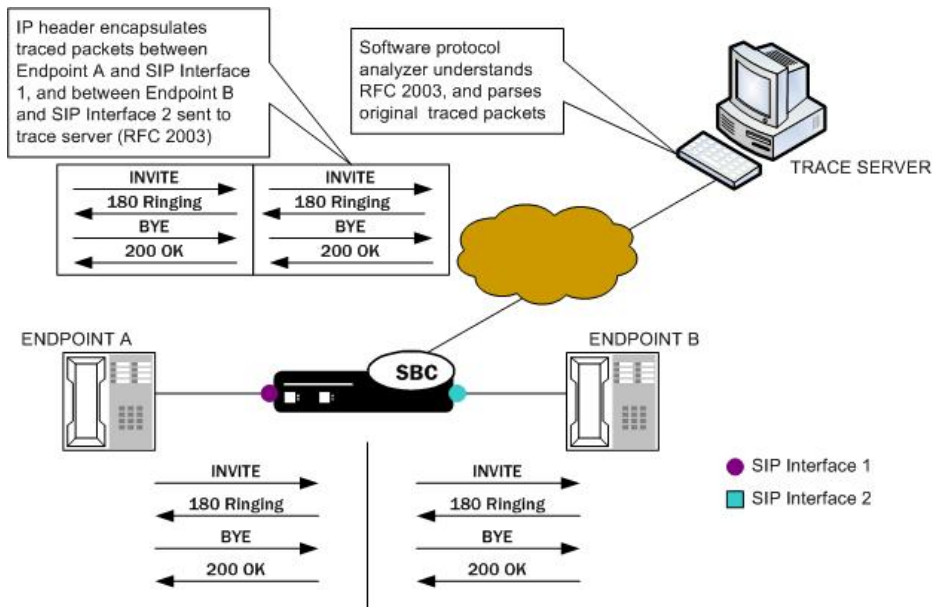
```
ACMEPACKET# packet-trace local F01:0 <"host IP address of Endpoint A">
ACMEPACKET# packet-trace local F02:0 <"host IP address of Endpoint B">
```

### Packet Trace for a Oracle Communications Session Border Controller Signaling Address

You can perform a packet trace for addresses internal to the Oracle Communications Session Border Controller; this can be the address, for example, of a SIP interface. Using signaling interface addresses puts the emphasis on the Oracle Communications Session Border Controller rather than on the endpoints by allowing you to view traffic from specified interfaces.

The commands you carry out for **packet-trace remote** would take the following form:

```
ACMEPACKET# packet-trace remote start F01:0 <IP address of Oracle
Communications Session Border Controller interface1>
ACMEPACKET# packet-trace remote start F02:0 <IP address of Oracle
Communications Session Border Controller interface2>
```





The commands you carry out for **packet-trace local** would take the following form:

```
ACMEPACKET# packet-trace local F01:0 <"host IP address of Oracle
Communications Session Border Controller interfacel">
ACMEPACKET# packet-trace local F02:0 <"host IP address of Oracle
Communications Session Border Controller interface2">
```

## Running Packet Trace

There are four steps you take when you use the **packet-trace remote** feature. For **packet-trace local**, there are only two.

- Configuring the Oracle Communications Session Border Controller with the trace server information so that the Oracle Communications Session Border Controller knows where to send replicated data. (**packet-trace remote** only)
- Setting up the capture filter ip proto 4 in your software protocol analyzer if you only want to see the results of the Oracle Communications Session Border Controller packet trace(s). (**packet-trace remote** only)
- Starting a packet trace.
- Stopping a packet trace.

This section provides information about how to perform all tasks.

### Configuring a Trace Server

Trace servers only apply to **packet-trace remote**. You need to configure a trace server on the Oracle Communications Session Border Controller; this is the device to which the Oracle Communications Session Border Controller sends replicated data. The Oracle Communications Session Border Controller supports one trace server.

To configure a trace server on your Oracle Communications Session Border Controller:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter.

```
ACMEPACKET(configure)# system
ACMEPACKET(system)#
```

3. Enter **capture-receiver** and press Enter.

```
ACMEPACKET(system)# capture-receiver
ACMEPACKET(capture receiver)#
```

4. **state**—Type **enabled** so that you can use the trace server to which you want to send the mirrored packets for calls you are packet tracing. The default is **disabled**. The valid values are:

- enabled | disabled

Disable capture receivers you are not actively using for traces to prevent potential service outages caused by the capture's system resource utilization.

5. **address**—Enter the IP address of the trace server; there is no default.
6. **network-interface**—Enter the name and subport of the Oracle Communications Session Border Controller network interface from which the Oracle Communications Session Border Controller is to send mirrored packets. Your entry needs to take the form name:subport. The default is **:0**.
7. Save and activate your configuration.

### Starting a Remote Packet Trace

You use the start a remote packet trace by entering the appropriate CLI command with these pieces of information:

- Network interface (name:subport ID combination)
- IP address to be traced; if you do not enter local and/or remote ports when you start the trace, the Oracle Communications Session Border Controller will trace all ports

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- (Optional) Local UDP/TCP port on which the Oracle Communications Session Border Controller sends and receives traffic to be traced
- (Optional) Remote UDP/TCP port to which the Oracle Communications Session Border Controller sends traffic, and from which it receives traffic to be traced; you cannot enter the remote port without specifying a local port

To start a packet trace with local and remote ports specified:

Note that the system supports packet trace on the Acme Packet 3820, Acme Packet 4500 and Acme Packet 4600, but not on the Acme Packet 1100 and VM Edition.

Enter the ACLI **packet-trace remote** command followed by a Space, and the word **start**. After another Space, type in the name and subport ID for the network interface followed by a Space, the IP address to be traced followed by a Space, the local port number followed by a Space, and then optionally the remote port number. Then press Enter.

```
ACMEPACKET# packet-trace remote start core:0 192.168.10.99 5060 5060
Trace started for 192.168.10.99
```

### Stopping a Remote Packet Trace

You stop a remote packet trace by entering the appropriate ACLI command with these pieces of information:

- Network interface (name:subport ID combination)
- IP address to be traced
- (Optional) Local UDP/TCP port on which the Oracle Communications Session Border Controller sends and receives traffic to be traced
- (Optional) Remote UDP/TCP port to which the Oracle Communications Session Border Controller sends traffic, and from which it receives traffic to be traced

If the packet trace you want to stop has no entries for local and/or remote ports, then you do not have to specify them.

1. To stop a packet trace with local and remote ports specified, enter the ACLI **packet-trace remote** command followed by a Space, and the word **stop**. After another Space, type in the name and subport ID for the network interface followed by a Space, the IP address to be traced followed by a Space, the local port number followed by a Space, and then optionally the remote port number. Then press Enter.

```
ACMEPACKET# packet-trace remote stop core:0 192.168.10.99 5060 5060
```

2. To stop all packet traces on the Oracle Communications Session Border Controller, enter the ACLI **packet-trace remote** command followed by a Space, and the word **stop**. After another Space, type the word **all** and press Enter.

```
ACMEPACKET# packet-trace remote stop all
```

### Starting a Local Packet Trace

You use the start a packet trace by entering the appropriate ACLI command with these pieces of information:

- Network interface (name:subport ID combination)
- (Optional) Enter a tcpdump command line within quotes

Note that the system supports local packet trace on all platforms.

Enter the ACLI **packet-trace local** command followed by a Space. Type in the name and subport ID for the network interface followed by a Space, the IP address to be traced followed by a Space, the local port number followed by a Space, and then optionally the remote port number. Then press Enter.

```
ACMEPACKET# packet-trace local s0p0 "host 192.168.12.12"
Files found in trace directory. Remove [y/n]?: y
File: /opt/traces/s0p0_0_00001_20150723095442.pcap
Packets: 5 Packets dropped: 0
```

The ACLI session does not accept use input while the **packet-trace local** command is running.

## Stopping a Local Packet Trace

Type Ctrl-C to stop a local packet trace. This also re-enables the command line session.

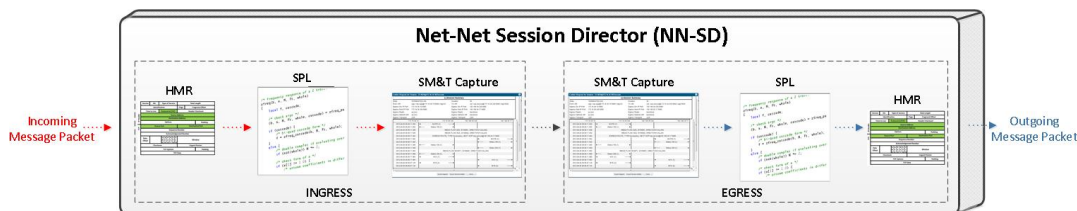
## SIP Monitor and Trace

The SIP monitoring and tracing feature provides the ability to set filters on the Oracle Communications Session Border Controller for filtering SIP session data, and displaying the results in a Web-based graphical user interface (GUI). You can use the data for the purpose of troubleshooting your Oracle Communications Session Border Controller(s).

This section provides information about how SIP monitoring and tracing works, configuring filters for filtering SIP session data, and how you can view/analyze monitored SIP session information via the Web-based GUI.

The SIP Monitor and Trace feature allows SIP sessions on the Oracle Communications Session Director(s) in your network to be monitored. When the Oracle Communications Session Border Controller receives an incoming message packet, it applies the configured Header Manipulation Rules (HMR), applies the message-receive Session Plug-in Language (SPL), and then captures the data from the message and displays it in the SIP Monitor and Trace application. The Oracle Communications Session Border Controller then processes the SIP information it received, and sends it out from the SIP Monitor and Trace application, applies the message-send SPL, and then applies HMR before sending the message out from the Oracle Communications Session Border Controller.

Refer to the illustration below for the message in/message out process for SIP Monitor and Trace.



You configure the monitoring process to use filters to filter the active session data from original ingress or final egress SIP session messages. These filters are based on Acme Packet's Command Line Interface (ACLI)-configured filters matching criteria or dynamic events that occur, and are used for the purpose of troubleshooting SIP sessions on the network.

As a system monitors active sessions, it captures data using the following filters:

- **Static filters** - Configured filters that filter the data on ingress and egress requests and responses in the SIP session dialogs. You configure these filters via the ACLI as part of the Oracle Communications Session Border Controller configuration. The configured filters save to the current configuration (after saving and activating the configuration).
- **Dynamic filters** - Filters you specify that match information in the ingress/egress SIP messages according to the filters you dynamically specified. You use the ACLI to specify these filters, but there is no change to the current configuration. The filters take effect immediately, and do not require the use of the "Save" and Activate commands. Using dynamic filters is recommended if you want to set specific filters but make no changes to the current configuration.

For more information about configuring static filters and dynamic filters, see [Filters to Configure](#) and [Dynamic Filters](#)

When a filter configuration is enabled, the system matches the values in the configured filters to the headers of messages before it applies any changes. If no match is found in the headers during monitoring, the system uses the filter defaults in the system configuration to perform the filtering. The system logs the filter results along with any additional call details and displays the results in the Web-based GUI.

The following illustration shows the SIP monitoring and tracing flow process.



## Filters to Configure

This section provides information about enabling the use of SIP monitoring and tracing filters you can configure on the Oracle Communications Session Border Controller. It includes a description and examples of the filter objects and attributes you can set to monitor specific SIP session data on the Oracle Communications Session Border Controller.

### Filter Objects

The Oracle Communications Session Border Controller provides filter objects you can configure on the Oracle Communications Session Border Controller for enabling/disabling filtering, and for creating customized filters for SIP Monitoring and Tracing. When filtering is enabled, the system can monitor and filter specific SIP session data and display it to the Web-based GUI. The filter objects you can configure include:

Filters	Description
Filters	Description
filter-config	Object that allows you to create custom filters to use for SIP monitoring and tracing. You can then configure session agents (SA) and/or realms to use these filters, or set sip-monitoring to use the filters on a global basis.  For more information, see .
sip-monitoring	Object that allows you to configure SIP monitoring and tracing features.  Note: You must configure the sip-monitoring object to enable filtering. A session agent and/or realm must also be configured, or you must set filtering on a global basis, for monitoring and tracing to occur.
state	Attribute that enables/disables SIP monitoring and tracing.  For more information, see .
monitoring-filters	Attribute that allows you to specify the name of the custom filter(s) to use on a global basis. This value is based on the filter(s) created in “filter-config”. You can also specify an * (asterisk) as a value for this attribute, which monitors all session data on the Oracle Communications Session Border Controller.  For more information, see <a href="#">Using Filters to Monitor on a Global-Basis</a> .
interesting-events	Object that allows you to configure the following attributes:  type - Sets the interesting events to monitor (short-session, local-rejection)  trigger-threshold - Sets the number of interesting events that must occur within the time set by the trigger-window value. If the number of events reaches the trigger-threshold during the trigger-window time, monitoring is started.  trigger-timeout - Sets the amount of time, in seconds, that the trigger is active once monitoring has started. If no interesting events occur in the time frame set for this value, monitoring never starts. For example, if trigger-timeout is set to 40, and no interesting events occur in 40 seconds, then monitoring never starts.  Note: Interesting Events are always enabled on a global-basis on the Oracle Communications Session Border Controller.

Filters	Description
	For more information, see <a href="#">Configuring Interesting Events</a> .

The following paragraphs provide information and procedures for configuring these features.

## Creating Custom Filters

You can create single or multiple custom, session filters on the Oracle Communications Session Border Controller for monitoring and tracing purposes. These filters allow incoming and outgoing session data to be filtered with specific information and then displayed to the Web-based GUI. You can use the custom filter(s) during monitoring on a global basis, or when monitoring session agent (SAs) and/or realms.

You create custom filters using the **filter-config** object at the path **Configure terminal > session-router > filter-config** in the ACLI.

The following table identifies the attributes you can configure for each filter.

Filters	Description
filter-config	Object that allows you to create a custom filter(s) to be used for monitoring and tracing on the Oracle Communications Session Border Controller.
name	Name of the custom filter.  Note: You specify this filter name when configuring global monitoring, SA monitoring, and/or realm monitoring.
address	IP address to be filtered. Depending on the value you specify for this attribute, filtering matches the IP address or IP address and netmask, in the message header. For example:  1.1.1.1 is <IP address>  1.1.1.1/24 is <IP address>/<Netmask>
user	Phone number or user-part to be filtered. Depending on the value you specify for this attribute, filtering matches the phone number string or the user-part with the following header information if it exists in the message: From URI, To URI, Request URI, P-Preferred URI, P-Asserted Identity, P-Associated URI, P-Called Party URI.

You can define a single or multiple filters with specific names and then specify the filter name(s) to use for global monitoring, session agent monitoring, and/or realm monitoring.

### Creating a Custom Filter

Use the following procedure to create a custom filter on the Oracle Communications Session Border Controller.

To configure a filter(s):

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related objects.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **filter-config** and press Enter to access the filter configuration-related attributes.

```
ACMEPACKET(session-router)# filter-config
ACMEPACKET(filter-config)#
```

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**name**—Enter a name to assign to this filter. Valid values are alpha-numeric characters. Default is blank.

```
ACMEPACKET(filter-config)# name FILTER1
```



**Note:** You can use this filter name when configuring monitoring on a global-basis, or when monitoring session-agents and/or realms.

**address**—Enter the IP address to apply to this filter. You can specify netmask if required. IP Address must be entered in dotted decimal format (0.0.0.0). Default is 0.0.0.0.

```
ACMEPACKET(filter-config)# address 1.1.1.1      (filters on IP address)
ACMEPACKET(filter-config)# address 1.1.1.1/24   (filters on IP address and
netmask)
```

**user**—Enter a phone number or user-part to apply to this filter. Valid values are numeric characters. Default is blank.

```
ACMEPACKET(filter-config)# user 5551212
```

You must specify either the phone number OR user part for the user attribute. If you want both the phone number AND user part to be filtered, you must create separate filters to set each value.

4. Enter **done** to save the filter.

```
ACMEPACKET(filter-config)# done
```

5. Enter **exit** to exit the filter configuration.

```
ACMEPACKET(filter-config)# exit
```

6. Enter **exit** to exit the session-router configuration.

```
ACMEPACKET(session-router)# exit
```

7. Enter **exit** to exit the configure mode.

```
ACMEPACKET(configure)# exit
```

8. Enter **save-config** to save the filter configuration.

```
ACMEPACKET# save-config
```

9. Enter **activate-config** to activate the filter configuration.

```
ACMEPACKET# activate-config
```

### Multiple Custom Filter Examples

The following examples show three custom filters (FILTER1, FILTER2, and FILTER3) created for SIP monitoring and tracing on the Oracle Communications Session Border Controller.

#### Filter 1

```
ACMEPACKET(filter-config)# name FILTER1
```

```
ACMEPACKET(filter-config)# address 1.1.1.1
```

```
ACMEPACKET(filter-config)# user 5551212
```

#### Filter 2

```
ACMEPACKET(filter-config)# name FILTER2
```

```
ACMEPACKET(filter-config)# address 3.3.3.3/24
```

```
ACMEPACKET(filter-config)# user 1781
```

#### Filter 3

```
ACMEPACKET(filter-config)# name FILTER3
```

```
ACMEPACKET(filter-config)# user sip
```

You can specify the Oracle Communications Session Border Controller monitoring process to use FILTER1, FILTER2, and/or FILTER3 for global monitoring, or for monitoring SAs and/or realms. However, before you apply the custom filters, you can enable/disable SIP monitoring on the Oracle Communications Session Border Controller.

To enable/disable SIP monitoring, see .

To use a custom filter(s) on a global basis, see [Using Filters to Monitor on a Global-Basis](#).


To use a custom filter(s) when monitoring SAs, see [Using Filters when Monitoring Session Agents](#).

To use a custom filter(s) when monitoring realms, see [Using Filters when Monitoring Realms](#).

## Enabling Disabling SIP Monitoring & Tracing

You can enable or disable the Oracle Communications Session Border Controller to perform SIP monitoring using the state parameter at the path Configure terminal->session-router->sip-monitoring in the ACLI.

Use the following procedure to enable/disable SIP monitoring on the Oracle Communications Session Border Controller.

 **Note:** You must enable the sip-monitoring object for monitoring and filtering to occur on the Oracle Communications Session Border Controller. With sip-monitoring enabled, you can configure a filter(s) on a global basis, as well as for a session agent and/or a realm. You can also initiate dynamic filter commands.

To enable/disable sip-monitoring:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related objects.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **sip-monitoring** and press Enter to access the SIP monitoring-related attributes.

```
ACMEPACKET(session-router)# sip-monitoring
ACMEPACKET(sip-monitoring)#
```

**state**—Enter whether or not to enable the sip monitoring on the Oracle Communications Session Border Controller. Default is enabled. Valid values are:

<ul style="list-style-type: none"> <li>• enabled (default)</li> </ul>
<ul style="list-style-type: none"> <li>• disabled</li> </ul>

4. Enter **done** to save the setting.

```
ACMEPACKET(sip-monitoring)# done
```

5. Enter **exit** to exit the sip-monitoring configuration.

```
ACMEPACKET(sip-monitoring)# exit
```

6. Enter **exit** to exit the session-router configuration.

```
ACMEPACKET(session-router)# exit
```

7. Enter **exit** to exit the configure mode.

```
ACMEPACKET(configure)# exit
```

8. Enter **save-config** to save the filters.

```
ACMEPACKET# save-config
```

9. Enter **activate-config** to activate the filters in the current configuration.

```
ACMEPACKET# activate-config
```

10. Configure global filters, or assign filters to a session agent and/or realm. For more information, see the following:

- [Using Filters to Monitor on a Global-Basis](#)

- [Using Filters when Monitoring Session Agents](#)
- [Using Filters when Monitoring Realms](#)

With sip-monitoring enabled, you can also initiate dynamic filter commands if required.

### Using Filters to Monitor on a Global-Basis

The Oracle Communications Session Border Controller allows you to filter SIP session data on a global-basis using the **monitoring-filters** object at the path **Configure terminal > session-router > sip-monitoring > monitoring-filters** in the ACLI. You can apply a single or multiple custom filter for global monitoring. For more information about creating a custom filter, see [Creating a Custom Filter](#).

To configure filtering on a global basis:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related objects.

```
ACMEPACKET(configure)# session-router  
ACMEPACKET(session-router)#
```

3. Type **sip-monitoring** and press Enter to access the SIP monitoring-related attributes.


```
ACMEPACKET(session-router)# sip-monitoring  
ACMEPACKET(sip-monitoring)#
```

4. Type **select** and press Enter to select the sip-monitoring objects.

```
ACMEPACKET(sip-monitoring)# select  
ACMEPACKET(sip-monitoring)#
```

**monitoring-filters**—Enter the custom filter name(s) you want to use when monitoring on a global-basis. You can enter multiple filter names in a comma-separated list (with no spaces) if required. To add to an existing filter list, use the “+” before the filter name you are adding. Use a “-” to remove filter names. Enter an \* (asterisk) to filter all session data.

```
ACMEPACKET(sip-monitoring)# monitoring-filters FILTER1,FILTER2  
ACMEPACKET(sip-monitoring)# monitoring-filters FILTER1,FILTER2 +FILTER3  
ACMEPACKET(sip-monitoring)# monitoring-filters FILTER1,FILTER2 -FILTER3  
ACMEPACKET(sip-monitoring)# monitoring-filters *
```

 **Note:** If you enter the \* with a filter name, the filter name is ignored and the Oracle Communications Session Border Controller uses the \* to filter all session data.

5. Enter **done** to save the configuration.

```
ACMEPACKET(sip-monitoring)# done
```

6. Enter **exit** to exit the sip-monitoring configuration.

```
ACMEPACKET(sip-monitoring)# exit
```

7. Enter **done** to save the sip-monitoring configuration.

```
ACMEPACKET(session-router)# done
```

8. Enter **exit** to exit the session-router configuration.

```
ACMEPACKET(session-router)# exit
```

9. Enter **exit** to exit the configure mode.

```
ACMEPACKET(configure)# exit
```

10. Enter **save-config** to save the configuration.

```
ACMEPACKET# save-config
```

11. Enter **activate-config** to activate the configuration.

```
ACMEPACKET# activate-config
```



## Using Filters when Monitoring Session Agents

You can configure the Oracle Communications Session Border Controller to perform filtering of SIP session data for session agent (SA) configurations. You must specify the hostname of the SA and the filter to use to perform the filtering, at the path **Configure terminal > session-router > session-agent** in the ACLI. For more information about creating a custom filter, see [Creating a Custom Filter](#).

To configure filtering for a Session Agent:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related objects.

```
ACMEPACKET(configure)# session-router
ACMEPACKET(session-router)#
```

3. Type **session-agent** and press Enter to access the session agent-related attributes.

```
ACMEPACKET(session-router)# session-agent
ACMEPACKET(session-agent)#
```

4. Type **select** and press Enter.

```
ACMEPACKET(session-agent)# select
ACMEPACKET(session-agent)#
```

**hostname**—Specify the hostname of the session agent to which you want to apply the custom filter(s).

```
ACMEPACKET(session-agent)# hostname SA1
```

**monitoring-filters**—Enter the custom filter name(s) you want to use when monitoring on a global-basis. You can enter multiple filter names in a comma-separated list (with no spaces) if required. To add to an existing filter list, use the “+” before the filter name you are adding. Use a “-” to remove filter names. Enter an \* (asterisk) to filter all SIP session data.

```
ACMEPACKET(session-agent)# monitoring-filters FILTER1,FILTER2
ACMEPACKET(session-agent)# monitoring-filters FILTER1,FILTER2 +FILTER3
ACMEPACKET(session-agent)# monitoring-filters FILTER1,FILTER2 -FILTER3
ACMEPACKET(session-agent)# monitoring-filters *
```



**Note:** If you enter the \* with a filter name, the filter name is ignored and the Oracle Communications Session Border Controller uses the \* to filter all session data.

5. Enter **done** to save the configuration.

```
ACMEPACKET(session-agent)# done
```

6. Enter **exit** to exit the session-agent configuration.

```
ACMEPACKET(session-agent)# exit
```

7. Enter **done** to save the configuration.

```
ACMEPACKET(session-router)# done
```

8. Enter **exit** to exit the session-router configuration.

```
ACMEPACKET(session-router)# exit
```

9. Enter **exit** to exit the configure mode.

```
ACMEPACKET(configure)# exit
```

10. Enter **save-config** to save the configuration.

```
ACMEPACKET# save-config
```

11. Enter **activate-config** to activate the configuration.

```
ACMEPACKET# activate-config
```

## Using Filters when Monitoring Realms

You can configure the Oracle Communications Session Border Controller to perform filtering of SIP session data for realm configurations. You must specify the realm identifier and the filter to use to perform the filtering, at the path **Configure terminal > media-manager > realm-config** in the CLI. For more information about creating a custom filter, see [Creating a Custom Filter](#).

To configure filtering for a realm:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **media-manager** and press Enter to access the media manager-related objects.

```
ACMEPACKET (configure) # media-manager
ACMEPACKET (media-manager) #
```

3. Type **realm-config** and press Enter to access the realm configuration-related attributes.

```
ACMEPACKET (media-manager) # realm-config
ACMEPACKET (realm-config) #
```

4. Type **select** and press Enter.

```
ACMEPACKET (realm-config) # select
ACMEPACKET (realm-config) #
```

**identifier**—Specify the identifier of the realm to which you want to apply the custom filter(s).

```
ACMEPACKET (realm-config) # identifier REALM1
```

**monitoring-filters**—Enter the custom filter name(s) you want to use when monitoring on a global-basis. You can enter multiple filter names in a comma-separated list (with no spaces) if required. To add to an existing filter list, use the “+” before the filter name you are adding. Use a “-” to remove filter names. Enter an \* (asterisk) to filter all SIP session data.

```
ACMEPACKET (realm-config) # monitoring-filters FILTER1,FILTER2
ACMEPACKET (realm-config) # monitoring-filters FILTER1,FILTER2 +FILTER3
ACMEPACKET (realm-config) # monitoring-filters FILTER1,FILTER2 -FILTER3
ACMEPACKET (realm-config) # monitoring-filters *
```



**Note:** If you enter the \* with a filter name, the filter name is ignored and the Oracle Communications Session Border Controller uses the \* to filter all session data.

5. Enter **done** to save the configuration.

```
ACMEPACKET (realm-config) # done
```

6. Enter **exit** to exit the realm-config configuration.

```
ACMEPACKET (realm-config) # exit
```

7. Enter **done** to save the configuration.

```
ACMEPACKET (media-manager) # done
```

8. Enter **exit** to exit the media-manager configuration.

```
ACMEPACKET (media-manager) # exit
```

9. Enter **exit** to exit the configure mode.

```
ACMEPACKET (configure) # exit
```

10. Enter **save-config** to save the configuration.

```
ACMEPACKET# save-config
```

11. Enter **activate-config** to activate the configuration.

```
ACMEPACKET# activate-config
```

## Global SA and Realm Filter Examples

The following are examples of global, session agent, and realm filters configured on the Oracle Communications Session Border Controller. These examples assume that FILTER1, FILTER2, and FILTER3 have been pre-configured as custom filters.

### Global Filter

```
ACMEPACKET (sip-monitoring) # monitoring-filters FILTER1, FILTER3
```

This filter captures the SIP session data based on the filter settings in FILTER1 and FILTER3 only, for all sessions on the Oracle Communications Session Border Controller.

### Session Agent Filters


```
ACMEPACKET (session-agent) # hostname SA1
ACMEPACKET (session-agent) # monitoring-filters FILTER2
ACMEPACKET (session-agent) # hostname SA2
ACMEPACKET (session-agent) # monitoring-filters FILTER2, FILTER3
```

These filters capture the SIP session data for SA1 only, based on the filter settings in FILTER2, and the SIP session data for SA2 only, based on the filter settings in FILTER2 and FILTER3.

### Realm Filters

```
ACMEPACKET (realm-config) # identifier REALM1
ACMEPACKET (realm-config) # monitoring-filters *
ACMEPACKET (realm-config) # identifier REALM2
ACMEPACKET (realm-config) # monitoring-filters FILTER1
```

These filters capture all SIP session data for REALM1, and the SIP session data for REALM2 only, based on the filter settings in FILTER1.

 **Note:** If you leave a monitoring-filter field blank, no monitoring takes place for that object.

## Configuring Interesting Events


Interesting events on the Oracle Communications Session Border Controller are those events that are considered “interesting” for the purpose of troubleshooting SIP sessions in your network. You can specify the type of interesting event you want to filter using the object, **interesting-events** at the path, **Configure terminal > session-router > sip-monitoring > interesting-events** in the ACLI.

Currently, there are two types of interesting events that the Oracle Communications Session Border Controller can monitor:

- **short-session** (short session events on the Oracle Communications Session Border Controller)
- **local-rejection** (local rejection events on the Oracle Communications Session Border Controller)

You can use the following trigger attributes to specify time provisioning for the interesting events:

- **trigger-threshold**
- **trigger-timeout**

 **Note:** You can also set a trigger-window object to support these trigger attributes. For more information, see [Configuring a Trigger Window](#).

The following table identifies the attributes you can set for the interesting-events object.

Filter	Description
interesting-events	Allows you to configure trigger attributes that apply to the filters you set on the Oracle Communications Session Border Controller. You can configure the following interesting-event attributes:

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Filter	Description
	Note: Interesting Events are always enabled on a global-basis on the Oracle Communications Session Border Controller.
type	Sets the interesting events to monitor (short-session, local-rejection)
trigger-threshold	Sets the number of interesting events that must occur within the time set by the trigger-window value. If the number of events reaches the trigger-threshold during the trigger-window time, monitoring is started.
trigger-timeout	Sets the amount of time, in seconds, that the trigger is active once monitoring has started. If no interesting events occur in the time frame set for this value, monitoring never starts. For example, if trigger-timeout is set to 40, and no interesting events occur in 40 seconds, then monitoring never starts.

The Oracle Communications Session Border Controller considers “short session” and local rejection interesting events. A session is viewed as a "short session" if the length of time, in seconds, is equal to or below the “short-session-duration” value configured at the path **Configure terminal > session-router > session-router-config > short-session-duration**. A local rejection can occur when sessions are locally rejected at the Oracle Communications Session Border Controller for any reason (for example, Session Agent (SA) unavailable, no route found, SIP signalling error, etc.)

If a short session or local rejection event occurs, the Oracle Communications Session Border Controller uses the values configured for the trigger attributes to determine when to start filtering the SIP session data.

If a short session event occurs when the Oracle Communications Session Border Controller is NOT monitoring, the event information is taken from the last BYE that occurred in the session; therefore, only some parts of the call flow may display in the Web-based GUI. If a local rejection event occurs when the Oracle Communications Session Border Controller is NOT monitoring, it displays only the information in the last rejected transaction.

## Interesting Events Configuration

Use the following procedure to configure interesting events for SIP monitoring and tracing on the Oracle Communications Session Border Controller.

To configure interesting events:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related objects.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **session-router** again and press Enter to access the session router configuration-related attributes.

```
ACMEPACKET (session-router) # session-router  
ACMEPACKET (session-router-config) #
```

**short-session-duration**—Enter the maximum session duration, in seconds, to be considered a short session. Default is 0 (disabled). Valid values are 0 to 999999999.

```
ACMEPACKET (session-router-config) # short-session-duration 30
```

4. Enter **done** to save the filters.

```
ACMEPACKET (session-router-config) # done  
ACMEPACKET (session-router-config) #
```

5. Enter **exit** to exit the interesting-events configuration.

```
ACMEPACKET (session-router-config) # exit  
ACMEPACKET (session-router) #
```

6. Type **sip-monitoring** and press Enter to access the SIP monitoring-related attributes.

```
ACMEPACKET(session-router) # sip-monitoring
ACMEPACKET(sip-monitoring) #
```

7. Type **select** and press Enter.

```
ACMEPACKET(sip-monitoring) # select
ACMEPACKET(sip-monitoring) #
```

8. Type **interesting-events** and press Enter to access the interesting events-related attributes.

```
ACMEPACKET(sip-monitoring) # interesting-events
ACMEPACKET(interesting-events) #
```

**type**—Enter the type of interesting event you for which you want to filter. Default is blank and disables this filter. Valid values are:

- short-session
- local-rejection

```
ACMEPACKET(interesting-events) # type short-session
```

**trigger-threshold** — (optional) Enter the number of interesting events that must occur within the time set by the trigger window value. If the number of events reaches the trigger-threshold during the trigger-window time, monitoring is started. Default is 0 (disabled). Valid values are 0 to 999999999.

```
ACMEPACKET(interesting-events) # trigger-threshold 50
```

**trigger-timeout** —Sets the amount of time, in seconds, that the trigger is active once monitoring has started. If no interesting events occur in the time frame set for this value, monitoring never starts. Default is 0 (trigger always on). Valid values are 0 to 999999999.

```
ACMEPACKET(interesting-events) # trigger-timeout 30
```

9. Enter **done** to save the filters.

```
ACMEPACKET(interesting-events) # done
```

10. Enter **exit** to exit the interesting-events configuration.

```
ACMEPACKET(interesting-events) # exit
```

11. Enter **exit** to exit the sip-monitoring configuration.

```
ACMEPACKET(sip-monitoring) # exit
```

12. Enter **done** to save the configuration.

```
ACMEPACKET(session-router) # done
```

13. Enter **exit** to exit the session-router configuration.

```
ACMEPACKET(session-router) # exit
```

14. Enter **save-config** to save the filters.

```
ACMEPACKET # save-config
```

15. Enter **activate-config** to activate the filters in the current configuration.

```
ACMEPACKET # activate-config
```

## Configuring a Trigger Window

The **trigger-window** attribute specifies the amount of time, in seconds, for the window of time that the “trigger-threshold” value must reach before monitoring begins. For example, if “interesting-event” type is set to short-session, “trigger-threshold” is set to 2, and trigger-window is set to 60, monitoring begins when the Oracle Communications Session Border Controller has discovered 2 short-session events in a 60 second window.

Use the following procedure to configure a trigger window.

To configure a trigger window:

## Call Traffic Monitoring

---

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **session-router** and press Enter to access the session router-related objects.

```
ACMEPACKET (configure) # session-router  
ACMEPACKET (session-router) #
```

3. Type **sip-monitoring** and press Enter to access the SIP monitoring-related attributes.

```
ACMEPACKET (session-router) # sip-monitoring  
ACMEPACKET (sip-monitoring) #
```

4. Type **select** and press Enter.

```
ACMEPACKET (sip-monitoring) # select  
ACMEPACKET (sip-monitoring) #
```

**trigger-window**—Enter the amount of time, in seconds, for the window of time that the trigger-threshold value must reach before monitoring begins. Default is 30. Valid values are 0 to 999999999. Zero (0) disables this the trigger-window parameter.

```
ACMEPACKET (sip-monitoring) # trigger-window 50
```

5. Enter **done** to save the filters.

```
ACMEPACKET (sip-monitoring) # done
```

6. Enter **exit** to exit the sip-monitoring configuration.

```
ACMEPACKET (sip-monitoring) # exit
```

7. Enter **done** to save the configuration.

```
ACMEPACKET (session-router) # done
```

8. Enter **exit** to exit the session-router configuration.

```
ACMEPACKET (session-router) # exit
```

9. Enter **exit** to exit the configure mode.

```
ACMEPACKET (configure) # exit
```

10. Enter **save-config** to save the filters.

```
ACMEPACKET# save-config
```

11. Enter **activate-config** to activate the filters in the current configuration.

```
ACMEPACKET# activate-config
```

### Example

The following is an example filter configuration, filtering interesting events with a trigger window on the Oracle Communications Session Border Controller. These parameters perform filtering on a global basis.

#### Monitoring Enabled on a Global Basis

```
ACMEPACKET (sip-monitoring) # state enabled
```

#### Short-Session Configured

```
ACMEPACKET (interesting-events) # type short-session  
ACMEPACKET (interesting-events) # trigger-threshold 2  
ACMEPACKET (interesting-events) # trigger-timeout 60
```

#### Local-Rejection Configured

```
ACMEPACKET (interesting-events) # type >local-rejection  
ACMEPACKET (interesting-events) # trigger-threshold 1  
ACMEPACKET (interesting-events) # trigger-timeout 0
```

#### Trigger-Window Configured

```
ACMEPACKET (sip-monitoring) # trigger-window 120
```

The configuration above has global SIP monitoring **enabled** and is set to capture interesting events that are **short-session** and local-rejection events.

Per the triggers for the short-session configuration, if **2** (trigger-threshold) short-session events occur in a window of **120** seconds (trigger-window), then monitoring is started. If no short-session events occur after **60** seconds (trigger-timeout), no monitoring is started.

Per the triggers for the local-rejection configuration, if more that **1** (trigger-threshold) local-rejection event occurs in a window of **120** seconds (trigger-window), then monitoring is started. The value of **0** (trigger-timeout) indicates that monitoring is always enabled for this event.


## Dynamic Filters

The SIP monitoring and tracing feature provides a time-saving feature of adding filters dynamically, and turning the filters ON and OFF as required. The filtering process performs on a dynamic basis dependant on the filters you specify.

### Dynamic Filter Commands

You can use the ACLI to initiate the following dynamic filtering commands:

- **capture start** - starts the filters you specify in the filter syntax
- **capture stop** - stops the filters you specify in the filter syntax

 **Note:** Initiating these commands does NOT change the values set in the ACLI-configured filters on the Oracle Communications Session Border Controller. The Oracle Communications Session Border Controller uses the dynamic filters until you initiate a stop command.

The syntax for the dynamic filter commands are:

capture start <main filter> <subfilter(s)>

capture stop <main filter> <subfilter(s)>

You MUST enter a “<main filter>” AND a “<subfilter(s)>” when initiating the “capture start” and capture stop commands.

The following table identifies the values you can use for each attribute in the command syntax.

Syntax Attribute	Values
<main filter>	global - monitors and captures all realm <realm name> - monitors and captures everything matching realm session-agent <session-agent name> - monitors and captures everything matching session agent. int-ev <short-session   local-rejection> - monitors and captures everything matching a short- session and/or local-rejection.
[<subfilter(s)>]	* - monitors and captures all sessions. user <Phone Number or User Part URI> - monitors and captures everything that matches this phone number or user part. addr-prefix <IP address or IP address and netmask> - monitors and captures everything that matches this address or address prefix.

## Call Traffic Monitoring

---

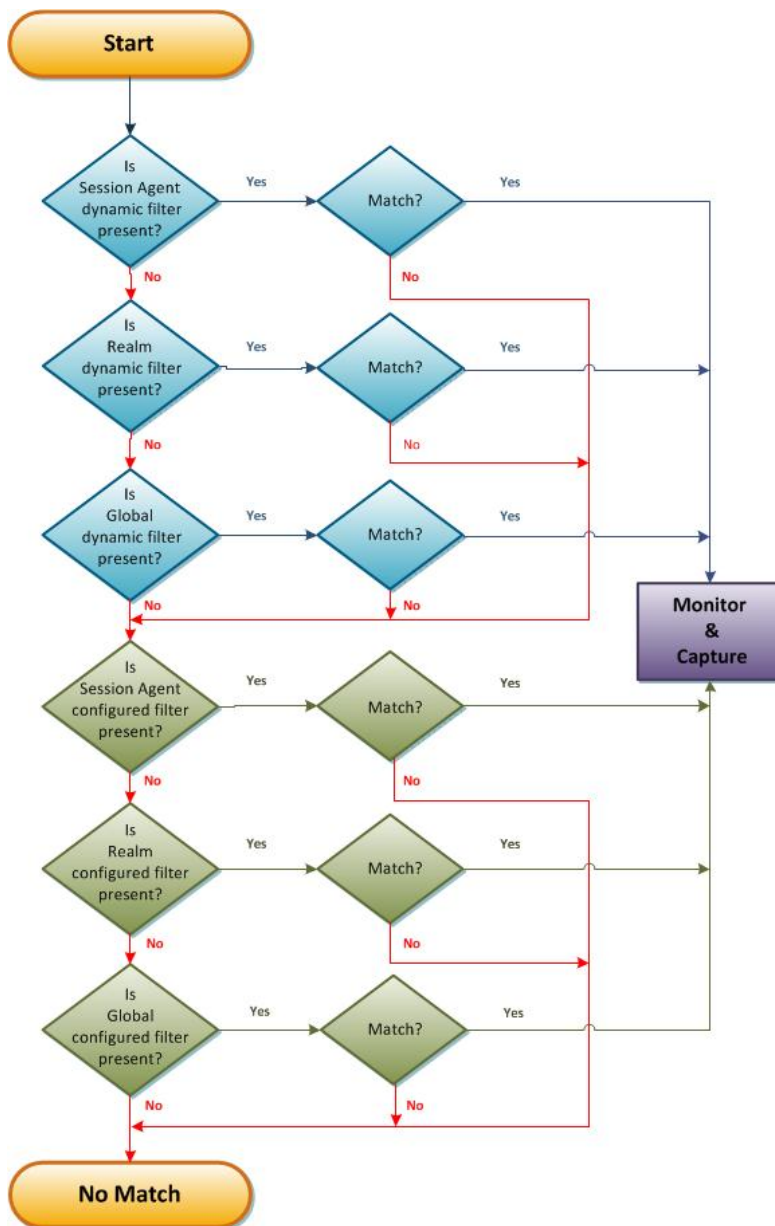
### Examples


The following table provides examples for using the dynamic filter commands.

Example	Description
capture start global *	Captures all session data.
capture start global user USER1	Captures all session data for USER1.
capture start global addr-prefix 1.1.1.1	Captures all session data for IP address 1.1.1.1.
capture start global addr-prefix 1.1.1.1/24	Captures all session data for IP address 1.1.1.1 using netmask of 24.
capture start session-agent 172.1.1.1 addr-prefix 10.10.10.10	Captures session data for SA 172.1.1.1 at IP address 10.10.10.10.
capture start int-ev local-rejection	Captures session data for interesting events that occur that are of type local-rejection.
capture start int-ev short session	Captures session data for interesting events that occur that are of type short-session.

The following flow chart shows the dynamic filter process.





 **Note:** Dynamic filters are removed after a save/activate command, and after a reboot/switchover of the Oracle Communications Session Border Controller.

Issuing another dynamic command may or may not affect previous dynamic commands that were already initiated. If you issue a dynamic command with a <main filter> object, and then issue another command with the same <main filter> object, the new command tasks precedence. If you issue a dynamic command with a different <main filter> object, then the Oracle Communications Session Border Controller uses both <main filter> commands to monitor traffic.

For example, if you enter the following dynamic command:

```
ACMPACKET# capture start realm1 user 123
```

and then enter:

```
ACMPACKET# capture start realm2
```

The Oracle Communications Session Border Controller monitors realm1 AND realm2 with user 123.

The command **show monitoring dynamic-monitoring-filters summary** indicates the dynamic filters currently being used. For example:

## Call Traffic Monitoring

---

```
ACMPACKET# show monitoring dynamic-monitoring-filters summary

Active Filters for realm [realm1]
Monitoring Filters: userPart=123

Active Filters for realm [realm2]
Monitoring Filters: userPart=123
```



**Note:** For more information about the `show monitoring dynamic-monitoring-filters summary` command, see.

To stop dynamic filter commands, you can initiate the `capture stop <main filter>` command. For example:

```
ACMPACKET# capture stop realm1 user 123
```



**Note:** To stop configured filters, you must manually remove them from the ACLI configuration.

## Clearing all Dynamic Filters

You can clear all dynamic filters using the following command:

- **reset monitoring dynamic-commands** - clears all dynamic filters previously initiated

The Oracle Communications Session Border Controller maintains a record of all dynamically initiated active filters. When you initiate this reset command, the Oracle Communications Session Border Controller searches through all of the filters and resets all the dynamic filters for each main filter (realm, session-agent, session-group, interesting event).

### Example

The following command is an example of using the reset command to clear all dynamic capture filters.

```
ACMEPACKET# reset monitoring dynamic-commands
```

The following message displays:

```
Reset all dynamically created monitoring capture commands....
```

## Clearing Event Monitoring Records

You can clear all records stored in the event monitoring in-memory database using the following command:

- **reset monitoring records** - clears all event monitoring records from the in-memory database.

Use the following procedure to clear all event monitoring records.

To clear event monitoring records:

1. At the prompt, type **reset monitoring records**, and press Enter.

```
ACMEPACKET# reset monitoring records
```

The following prompt displays:

```
All in-memory event monitoring records will be deleted [y/n]?:
```

2. Type **y** and press Enter.

```
All in-memory event monitoring records will be deleted [y/n]?: y
```

The following message displays.

```
Deleting the in-memory event records.
```

If you enter **n** for Step 2, the following message displays.

```
 Cancelling the reset.
```

No event monitoring records are deleted.



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## Web-based GUI

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### Introduction

This chapter provides information and procedures for using the embedded Web-based GUI on your Oracle Communications Session Border Controller. This GUI displays results of filtered SIP session data from one or multiple Oracle Communications Session Border Controllers, and provides traces in a common log format for local viewing or for exporting to your PC. The Web-based GUI client provides local viewing support of SIP sessions only.

---

### Web Browser Support

You can use any of the following Web browsers to access Oracle Communications Web-based GUI for viewing SIP session data:

- Internet Explorer versions 9.0 and higher
- Mozilla Firefox versions 12.0 and higher
- Google Chrome versions 19.0.1084.46m and higher

---

### Overview

The Web-based GUI allows you to view information about monitored SIP sessions on the Oracle Communications Session Border Controller(s). It supports four types of summary reports:

- **Sessions**
- **Registrations**
- **Subscriptions**
- **Notable events (interesting events)**

Each report summarizes the applicable information, displays it on easy-to-read pages, and allows you to sort and customize columns within each report. You can also perform searches to find specific information in a report.

Each type of report provides sorting, searching, and paging functionality and provides action buttons to use to display additional information or perform a task.

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### Logging in

You can login to the Web-based GUI using your Web browser for browsers that support this GUI). The Web interface supports authentication functionality similar to a user logging in via TELNET, Secure Shell (SSH), File Transfer

## Web-based GUI

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Protocol (FTP), and SSH File Transfer Protocol (SFTP). SIP Monitoring and Tracing also supports RADIUS authentication.

 **Note:** Regardless of the privilege level set via authentication (user or admin), logging into the Web-based GUI gives user-level access privileges.


Up to 5 users can log into the same Web-based GUI interface. If more than 5 users attempt to log in, the following error message displays:

User limit reached. Please try again later.

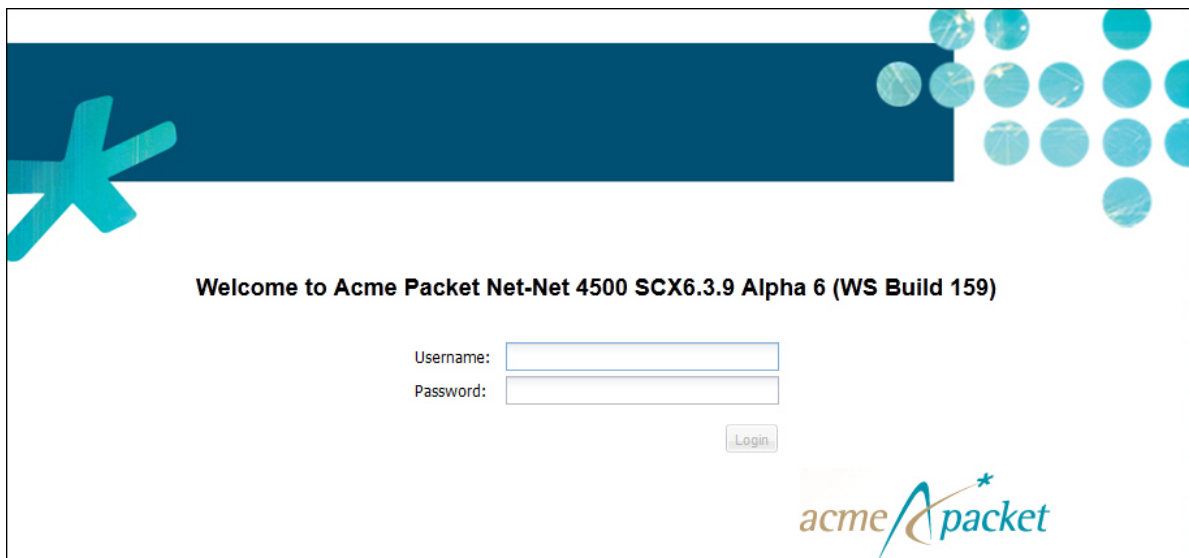
To login to the Web-based GUI:

1. On your PC, open an Internet Browser.
2. Start the Web-based GUI by using either the HTTP or HTTPS login:

```
http://<Server IP address>  
https://<Server IP address>
```

 **Note:** Logging in using HTTP and/or HTTPS is dependant on the setting that was made during the Web-based GUI installation wizard. You can change these settings using the ACLI. For more information, see Configuring the Web Server.

3. Enter your Web-based GUI username and password. Default username is “**user**” or “**admin**”. Default password is packet.
4. Click <Login>.



5. The Web-based GUI opens.

SIP monitoring and tracing reports

Logout

acme packet

Monitoring and Tracing

Sessions

Registrations

Subscriptions

Notable Events

SIP Session Summary

Search Criteria: All

Search Show all

Page Size 50 Page 1 of 1 Displaying 1 - 15 of 15

Start Time	State	Call ID	Request URI	From URI	To URI	Ingress Realm	Egress Realm
2012-06-14 15:46:34.915	TERMINATED-200	5-17902@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:46:33.914	TERMINATED-200	4-17902@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:46:32.914	TERMINATED-200	3-17902@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:46:31.914	TERMINATED-200	2-17902@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:46:30.914	TERMINATED-200	1-17902@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:45:35.557	FAILED-408	5-17609@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:45:34.557	FAILED-408	4-17609@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:45:33.558	FAILED-408	3-17609@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:45:32.559	FAILED-408	2-17609@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:45:31.559	TERMINATED-0	1-17609@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:45:14.210	TERMINATED-200	5-17548@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:45:13.211	TERMINATED-200	4-17548@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core
2012-06-14 15:45:12.210	TERMINATED-200	3-17548@192.168.200.226	sip:192.168.204.71:5060	*2273630@tel781-414-23...	sut+esp:kam@192.168.204.7...	access	core

Refresh Ladder Diagram Export Session Details Export Summary

6. Click on the report you want to view in the slider on the left side of the page. On all Sessions, Registrations, Subscriptions, and Notable Events report pages, you can perform the following:

- Customize the page display
- Perform a search for specific data
- Display a ladder diagram
- Export session details
- Export session summary

The following paragraphs describe each of these tasks.

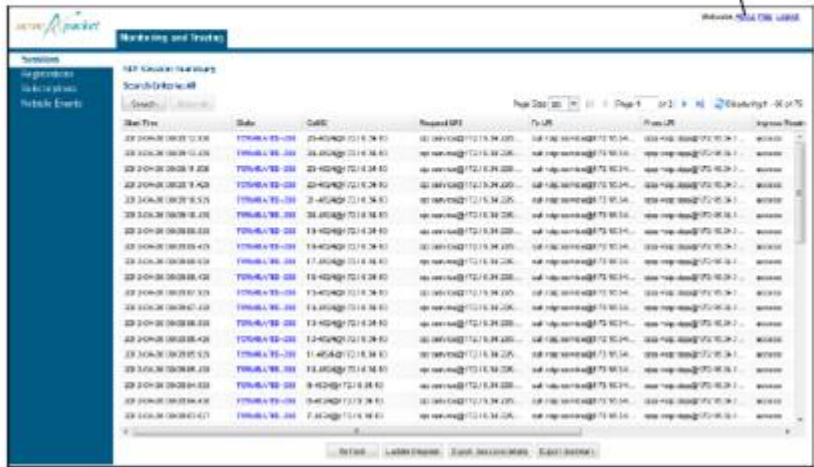
7. To logout of the Web-based GUI, click **Logout** in the upper right corner of the page.

## About Information

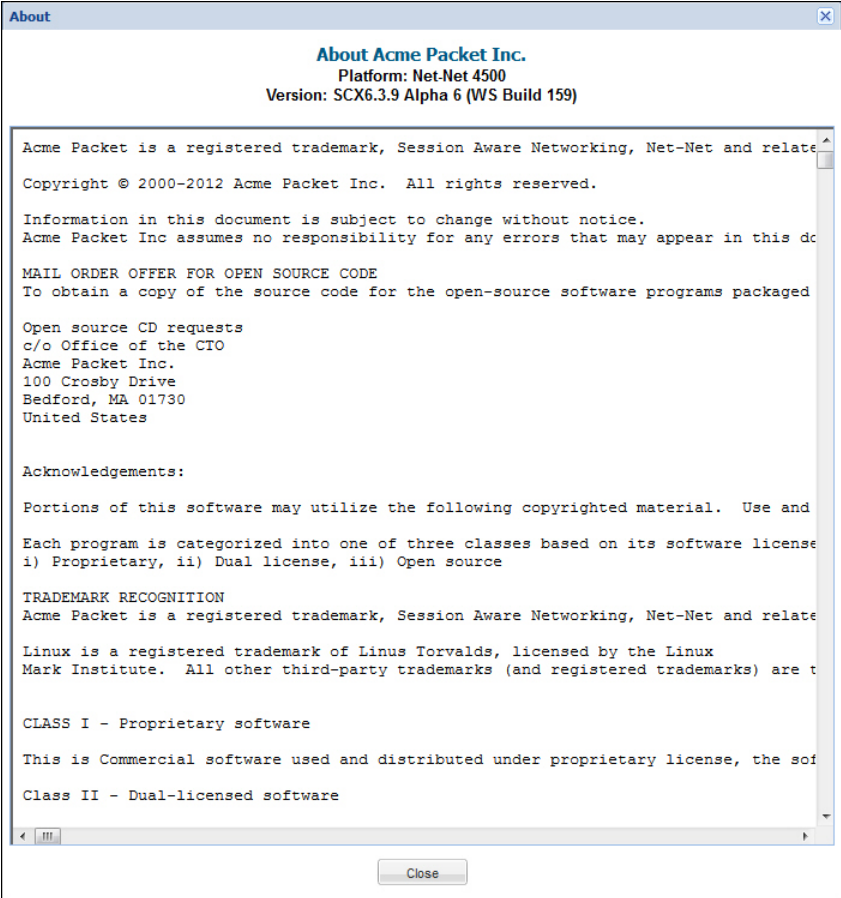
You can display information about the Net-Net SBC you are currently logged into via the Web-based GUI. You can access About from any main report summary page.

To display the About information:

1. From any main report summary page, click the About link in the upper right corner on the page.



The following information displays.



This box displays the following about the Net-Net SBC you are currently logged into:

- Net-Net Platform type (3800, 4500, Server Edition, or Virtual Machine Edition)



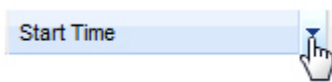
- Version Number of software (including Build Number)
  - Trademark Recognition
  - Copyright
  - Open Source Mailing Address
  - Acknowledgements
  - Relevant Proprietary and Open Source Licenses
2. Click <Close> to close the box.

## Customizing the Page Display

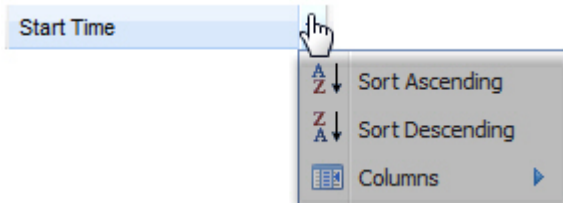
You can customize the data presented in the SM& T pages by changing whether or not specific columns display and how they display. You can also sort the order of item entries.

To customize the page display within any report:

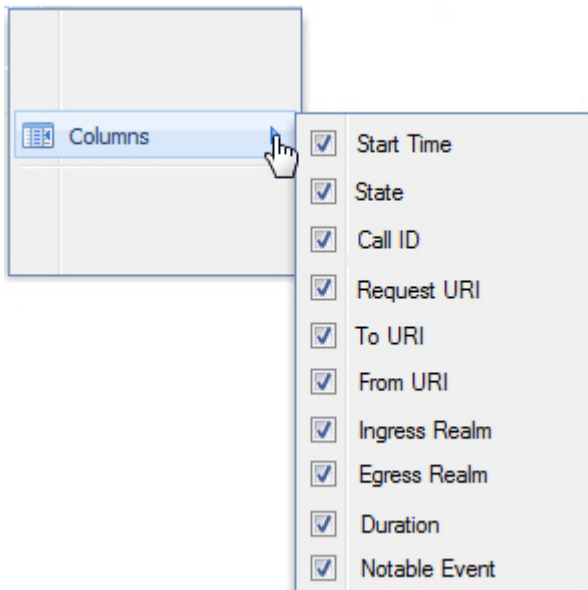
1. Position the cursor over a column heading. A pointer displays on the right hand side of the box. For example:



2. Click the down arrow to display the menu. For example:



3. Click Sort Ascending to sort the data in the table in ascending order.
4. Click Sort Descending to sort the data in the table in descending order.
5. Click Columns to access and customize a list of column names. For example:



6. Place a checkmark in the box to display that heading/column in the window. Remove the checkmark to hide the heading/column in the window.
7. Click Group By This Field to group all items in the list according to the column head you currently have selected. For example, if you select the “Site” column heading, and then select Group By This Field, all items in the list are grouped by Site.

## Web-based GUI

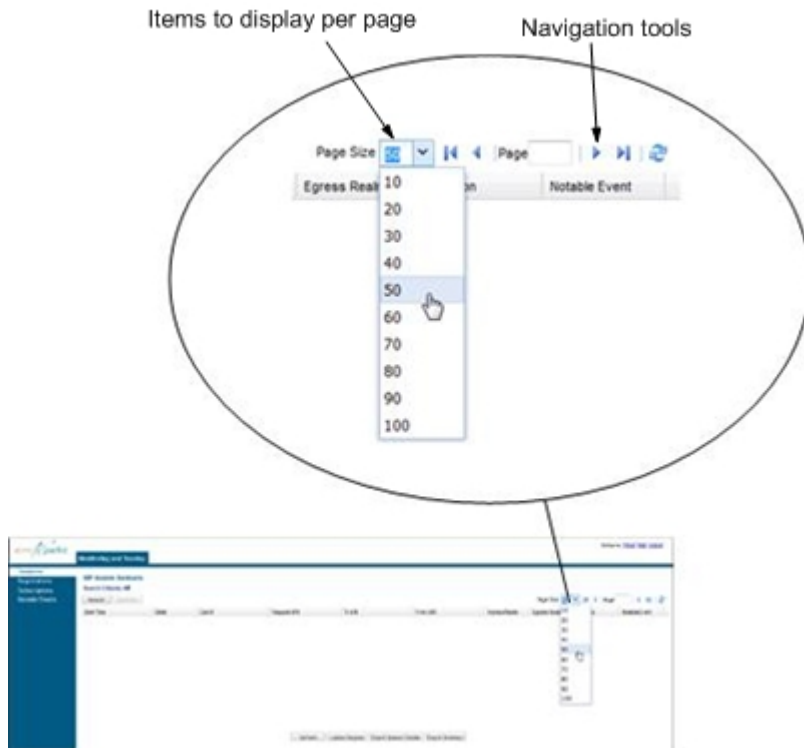
- For the “**Show in Groups**” option, place a checkmark in the box to show the current heading in the group when you select Group By This Field. Remove the checkmark to hide the current heading when Group By This Field is selected.
- To close the heading menus, click anywhere in the window.

### Changing Number of Data Items on the Page

By default, 50 data items are shown per page. You can change the number of items viewed for all tables in Trunk Manager.

To change the number of data items displayed:

- At the top right corner of the window, click the down arrow next to **Size**. The drop down list of values appears.

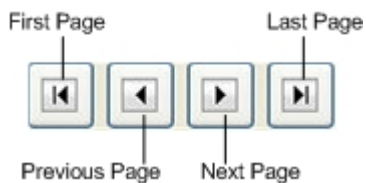


- Click the number of data items you want to display per page. Default is 50. Valid values are 10 to 100 in increments of 10.

### Navigating Pages

To navigate through multiple pages:

- Use the navigation arrows located at the top right corner of the window to navigate through multiple pages.
- Click the navigation icons to display the desired page, such as the first page, previous page, next page, and the last page of Events list view.




### Refresh

To refresh the page:

Click the <Refresh> button at the bottom of the page.

or

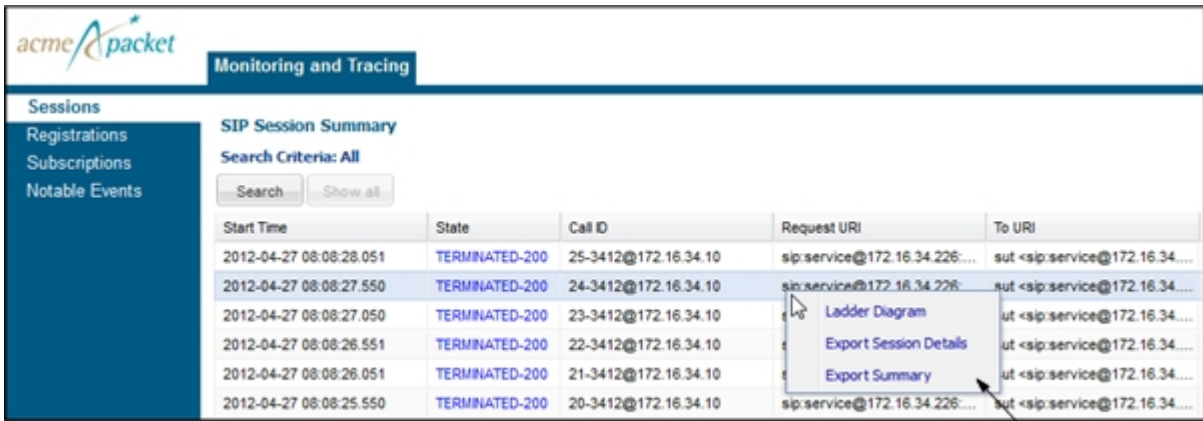
Click the  in the upper right corner of the page.

## Pop-up Context Menu

All reports (Sessions, Registrations, Subscriptions, and Notable Events) support a pop-up context menu that you can use to select a Ladder diagram, Export Session Details, and Export Session Summary. You display the pop-up context menu by selecting a record and right-clicking to show a menu.

To display the pop-up context menu in any report:

1. Click on a record in the report page.



The screenshot shows the 'acme packet' logo and 'Monitoring and Tracing' header. A sidebar on the left lists 'Sessions', 'Registrations', 'Subscriptions', and 'Notable Events'. The main content area is titled 'SIP Session Summary' with search criteria set to 'All'. A table with columns 'Start Time', 'State', 'Call ID', 'Request URI', and 'To URI' is shown. A context menu is open over a row, with options: 'Ladder Diagram', 'Export Session Details', and 'Export Summary'. An arrow points to the menu with the label 'Pop-up Context Menu'.

Start Time	State	Call ID	Request URI	To URI
2012-04-27 08:08:28.051	TERMINATED-200	25-3412@172.16.34.10	sip.service@172.16.34.226....	sut < sip.service@172.16.34....
2012-04-27 08:08:27.550	TERMINATED-200	24-3412@172.16.34.10	sip.service@172.16.34.226....	sut < sip.service@172.16.34....
2012-04-27 08:08:27.050	TERMINATED-200	23-3412@172.16.34.10	...	sut < sip.service@172.16.34....
2012-04-27 08:08:26.551	TERMINATED-200	22-3412@172.16.34.10	...	sut < sip.service@172.16.34....
2012-04-27 08:08:26.051	TERMINATED-200	21-3412@172.16.34.10	...	sut < sip.service@172.16.34....
2012-04-27 08:08:25.550	TERMINATED-200	20-3412@172.16.34.10	sip.service@172.16.34.226....	sut < sip.service@172.16.34....

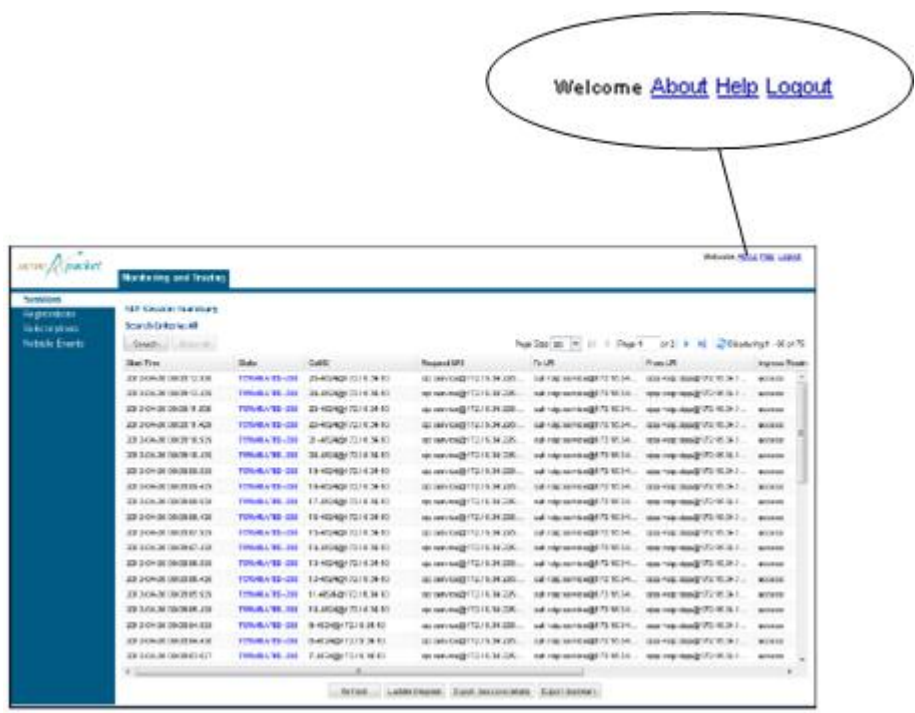
2. Right-click the mouse to display the pop-up context menu.
3. Drag the mouse to an option on the menu and left-click the mouse to select it. The applicable page displays or the export begins, depending on the option you selected (Ladder Diagram, Export Session Details, or Export Summary).

## Help Menu

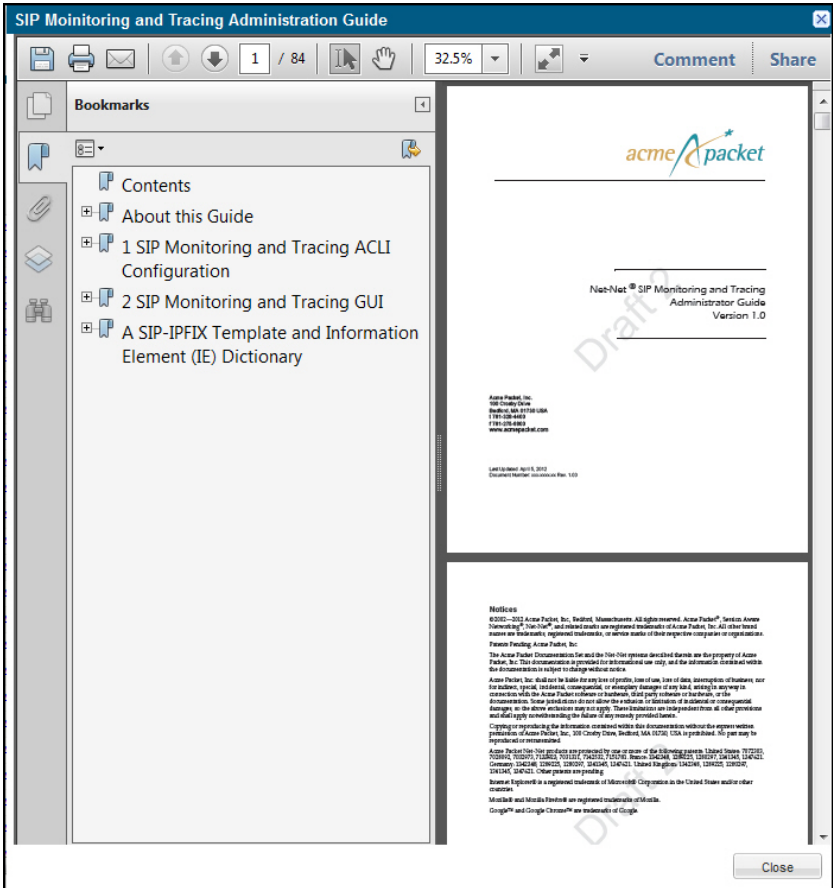
The Web-based GUI provides online help for all of the pages within the GUI. You can access help from any page by clicking the Help link in the upper right corner of the page.

To display Help:

1. From any main report summary page, click the Help link in the upper right corner on the page.



The applicable help for the current page displays in a pop-up window.



2. Click **<Close>** to close the box.

or

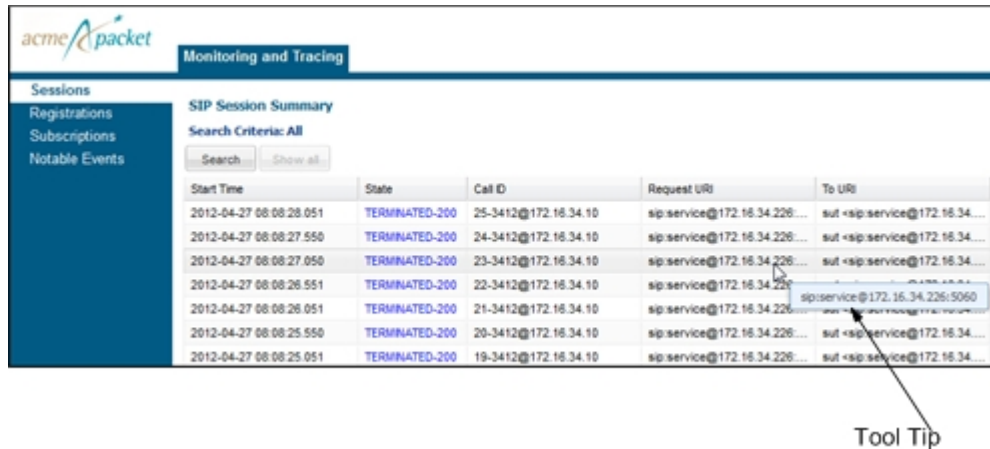
Click the X in the upper right corner of the pop-up window.

## Tool-Tip Help

Throughout the Web-based GUI, you can scroll over a record within a summary report and display quick information about that record in a temporary pop-up box. This allows you to view a summarized version of help for that record.

To view a tool-tip help:

1. From any main report summary page, use the mouse to scroll over a record in a summary report.



A blue-shaded box displays allowing you to view quick help about the record.

2. To close the tool-tip, scroll off of the record. Alternatively, click at another location within the summary report.



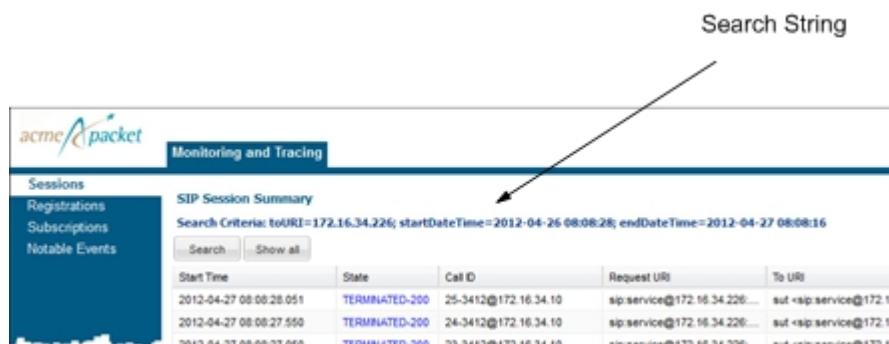
**Note:** Ladder diagrams for a specific record also display tool tip summary information. For more information, see [Ladder Diagram](#).

## Search for a Record

The **<Search>** button at the top of the report page allows you to perform a search to find a specific record(s) within a report (Sessions, Registrations, Subscriptions, Notable Events). It also allows you to specify criteria on which to perform the search.




After defining a search criteria in the Search Filter dialog box, clicking **<Search>**, automatically populates the report page with the records that match the criteria you specified. The search performs the filtering process of criteria dependent on the report page from which you are running the search.

For example, performing a search from the Sessions report page displays only the reports pertaining to call sessions. If you performed a search on the Registration report page, only the reports pertaining to call registrations displays on the report page. The search string containing the criteria on which you performed the search, displays in the top left corner of the page.



The search criteria is saved until you click **<Reset>** in the dialog box, or until you log out of the HTTP session.

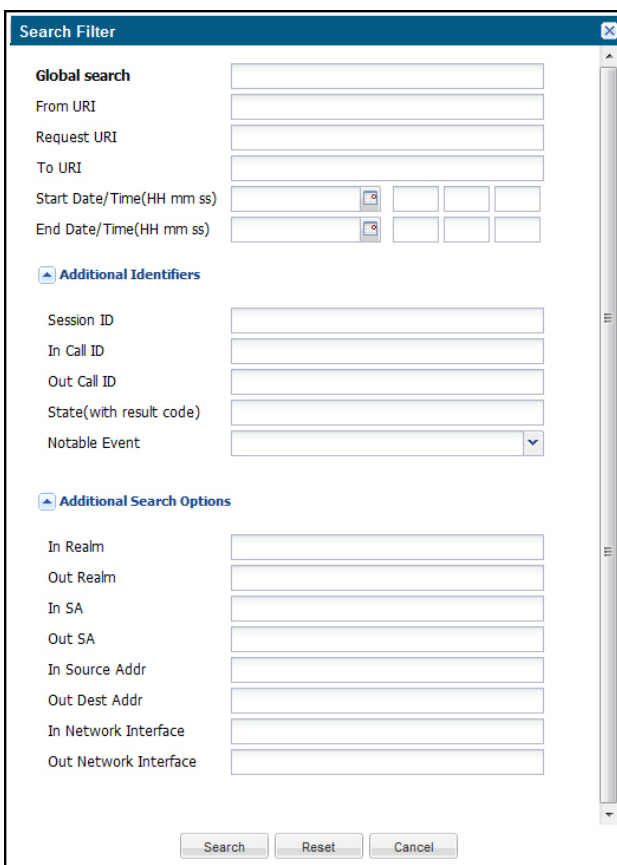
To perform a search:

-  **Note:** You can specify a value for any or all of the fields in the Search box. The search process searches for records with all of the values you specify and displays only the records with these values. If you perform a “Global Search”, AND specify values in other fields, the search process searches the other specified fields first and then filters on the “Global Search” field.
-  **Note:** If you specify a “\*” in a search string, the search is performed on that exact string. For example, if you search for “123\*45”, the search shows results for all strings containing “123\*45”.
-  **Note:** You can use quotes (“ ”) to specify a search. For example, you can enter Smith and the search finds all of the records that match Smith, such as:


```
John Smith field<sip:sipp@192.168.1.70:5070>;tag=12260SIPpTag001
```

If you enter a space before or after a quotation mark, (for example, "Smith "), the search returns no data.

1. In any reports page, click <**Search**>.
2. The following dialog box displays.

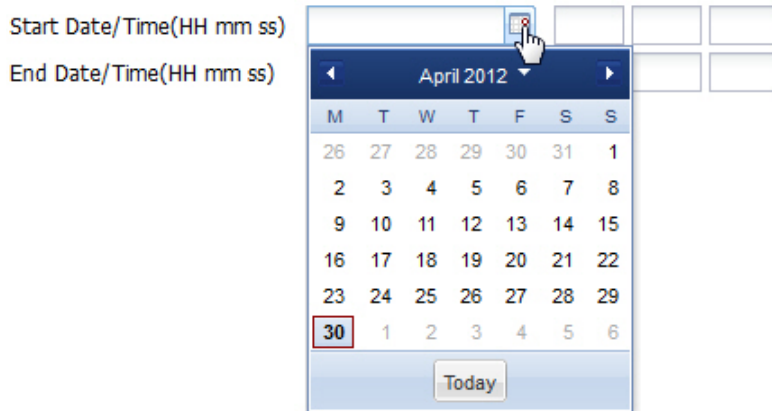


3. In the Global Search field, specify a string to search all parameters in all records. Valid values are alpha-numeric characters.

 **Note:** The Global Search option searches all parameters in all the session records stored in memory. All values you specify in other fields are searched before the value specified in the Global Search field is used.

4. In the From URI field, enter the URI formatted string of the call source information you are searching. Valid values are alpha-numeric characters. For example, sipp<sip:sipp@172.16.34.10:5060;tag=24.
5. In the Requested URI field, enter the URI formatted string that contains a protocol, name, location, or any other applicable characteristic, that is sent by the Oracle Communications Session Border Controller in the REQUEST header. Valid values are alpha-numeric characters. For example, sip:service@172.16.34.226:5060.

6. In the To URI field, enter URI formatted string of the call destination information you are searching. Valid values are alpha-numeric characters. For example, sut<sip:service@172.16.34.226:5060;tag=99.
7. In the Start Date/Time (HH mm ss) field, enter a starting date to search on in the first text box in the format YYYY-MM-DD (where Y =year, M=month, and D=day). Alternatively, click on the calendar icon in this field to display a calendar from which you can select a date. Navigate the calendar to find the date you want and click on it to enter it into this field, or click <Today> to enter today's date. For example, 2012-04-15 would search for all records starting on April 15, 2012. Valid values are numeric characters only. Enter a start time to search on in the last three text boxes in the format HH mm ss (where H=hour, m=minutes, and s=seconds). For example, 01 30 45 would search for all records starting at 1:30 and 45 seconds. Valid value are numeric characters only.



8. In the End Date/Time (HH mm ss) field, repeat the process of entering a date and time as provided in Step 7.
9. To search on additional parameters, click on the Additional Identifiers down arrow to expand the dialog box.

**Additional Identifiers**

Session ID

In Call ID

Out Call ID

State(with result code)

Notable Event

10. In the Session Id field, enter the ID of the call session you want to search. Valid values are alpha-numeric characters. For example, 22-3412@172.16.34.1.
11. In the In Call ID field, enter the ID of the incoming call (phone number and source IP address). Valid values are alpha-numeric characters. For example, 25-3412@172.16.34.10.
12. In the Out Call ID field, enter the ID of the outgoing call (phone number and IP address). Valid values are alpha-numeric characters. For example, 14-3412@172.14.54.6.
13. In the State (with result code) field, enter the status of the call session with the result code for which you want to search. Valid values are (case-sensitive):
  - INITIAL-<result code>
  - EARLY-<result code>
  - ESTABLISHED-<result code>
  - TERMINATED-<result code>
  - FAILED-<result code>

Result codes can range from 1xx to 5xx. For example, terminated-200, or failed-400.

14. In the Notable Event field, select the notable event for which you want to search. Valid values are:
  - any-event - search displays any notable event that was stored in memory.

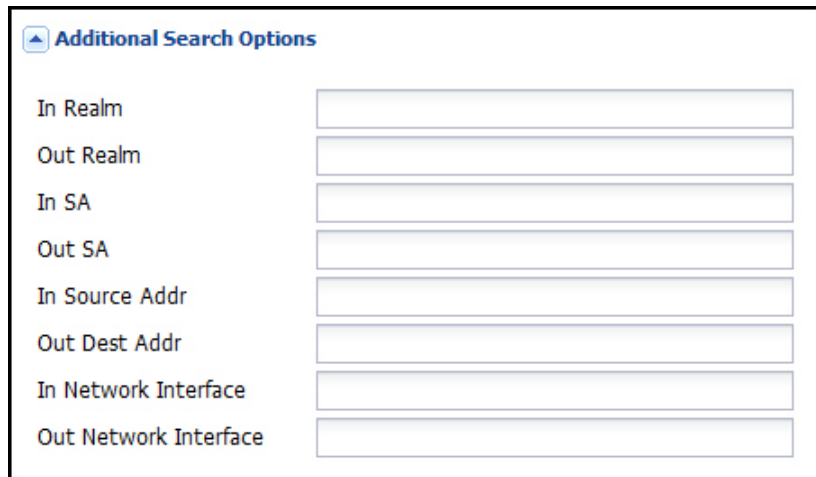


## Web-based GUI

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- short-session - search displays only records that indicate a short-session duration has occurred.
- local-rejection - search displays only records that indicate a local-rejection has occurred.

15. To search on additional parameters, click on the Additional Search Options down arrow to expand the dialog box.



The screenshot shows a dialog box titled "Additional Search Options" with a blue arrow icon on the left. It contains eight text input fields, each with a label to its left:

- In Realm
- Out Realm
- In SA
- Out SA
- In Source Addr
- Out Dest Addr
- In Network Interface
- Out Network Interface

16. In the “**In Realm**” field, enter the name of the realm for which the incoming call belongs. Valid values are alpha-numeric characters. For example, access.
17. In the “**Out Realm**” field, enter the name of the realm for which the outgoing call belongs. Valid values are alpha-numeric characters. For example, backbone.
18. In the “**In SA**” field, enter the name of the session agent (SA) on the incoming call session. Valid values are alpha-numeric characters. For example, SA1.
19. In the “**Out SA**” field, enter the name of the session agent (SA) on the outgoing call session. Valid values are alpha-numeric characters. For example, SA2.
20. In the “**In Source Addr**” field, enter the source IP address of the SA that accepted the incoming call session. IP Address must be entered in dotted decimal format (0.0.0.0). For example, 172.45.6.7.
21. In the “**Out Dest Addr**” field, enter the destination IP address of the SA that accepted the outgoing call session. IP Address must be entered in dotted decimal format (0.0.0.0). For example, 172.64.56.7.
22. In the In Network Interface field, enter the incoming core network interface that connects the Oracle Communications Session Border Controller to your network. IP Address must be entered in dotted decimal format (0.0.0.0). For example, 192.45.6.7.
23. In the Out Network Interface field, enter the outgoing network interface that connects your Oracle Communications Session Border Controller to the outside network. IP Address must be entered in dotted decimal format (0.0.0.0). For example, 192.45.6.8.
24. Click **<Search>** to perform the search with the values you specified. A list of the records that the search process filtered, display in the window.

The Web-based GUI saves the search specifications until you click **<Reset>** in the search dialog box, OR until you log out of the Web-based GUI.

## Exporting Information to a Text File

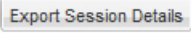
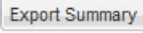

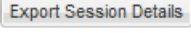
The Web-based GUI allows you to export information to a text file from the Sessions, Registrations, Subscriptions and Notable Events Reports, as well as from a specific ladder diagram, or from a page containing the results of a search. The data exports to a file that you can open and view as required.

You can export any of the following:

- All information from each report
- Information from a specific record only
- Information from a search result
- Information from a Ladder Diagram



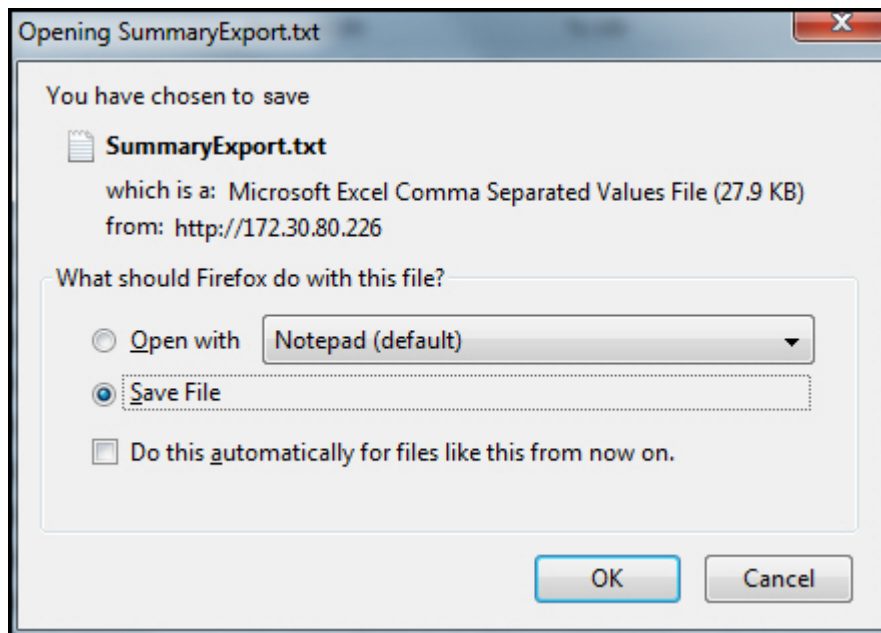
The following table identifies the buttons to use to export specific information from the Web-based GUI. All the export buttons in the GUI export to text files.

Button	Description
From the Sessions, Registrations, Subscriptions, and Notable Events Reports:	
	Exports the SIP messages and media events associated with the selected session, to a file in text format on the local machine.
	Exports all logged session summary records to a file in text format on the local machine. Note: This button exports ALL call session summary records or the records that matched a search criteria to the file.
From the Ladder Diagram:	
	Exports all of the information in the Ladder Diagram (Session Summary, SIP Message Details, and QoS statistics), to an HTML file format on the local machine.
	Exports detailed information about the SIP messages and media events in the Ladder Diagram associated with the selected session, to a file in text format on the local machine.


The following procedure is an example of using the export buttons in the Web-based GUI:

To export information to a text file:

1. In the applicable Report window or Ladder Diagram window, click the appropriate Export button. The following example illustrates that the <Export Summary> button was clicked on a report page. The following prompt displays.



The Web-based GUI assigns the file name for the text file (shown as SummaryExport.txt in the illustration above.)

 **Note:** The Web-based GUI exports Ladder Diagrams as HTML files.

2. Click “Open with” and select the application for which to open the resulting text file. Alternatively, click Save File to save the text file to your local PC.
3. Click <OK> to export the session or media information to the text file.

## Web-based GUI

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The following illustration shows a partial summary of the Sessions Report exported to a text file and opened using Microsoft Word. For additional examples of exported files, see Appendix B.

Export Summary:

```
-----Session Summary-----
Startup Time: 2012-04-26 08:40:44.624
State: TERMINATED-200
Duration: 9
From URI: sipp &lt;sip:sipp@172.16.34.10:5060&gt;;tag=25
To URI: sut &lt;sip:service@172.16.34.226:5060&gt;;tag=3453
Ingress Src Address: 172.16.34.10
Ingress Src Port: 5060
Ingress Dest Address: 172.16.34.226
Ingress Dest Port: 5060
Egress Source Address: 192.168.34.226
Egress Source Port: 5060
Egress Destination Address: 192.168.34.17
Egress Destination Port: 5060
Ingress Realm: access
Egress Realm: backbone
Ingress NetworkIf: access
Egress NetworkIf: backbone

-----Session Summary-----
Startup Time: 2012-04-26 08:40:43.624
State: TERMINATED-200
Duration: 9
From URI: sipp &lt;sip:sipp@172.16.34.10:5060&gt;;tag=24
To URI: sut &lt;sip:service@172.16.34.226:5060&gt;;tag=3452
Ingress Src Address: 172.16.34.10
Ingress Src Port: 5060
Ingress Dest Address: 172.16.34.226
Ingress Dest Port: 5060
Egress Source Address: 192.168.34.226
Egress Source Port: 5060
```

## Session Reports

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The Session Report is a SIP session summary of all logged call sessions on the Oracle Communications Session Border Controller. The data that displays in this table is dependent on the filters set via the ACLI on the Oracle Communications Session Border Controller.



**Note:** For setting filters to capture applicable data, see [Filters to Configure](#).

The columns that display on the Session Report page are dependent on the columns you selected in the procedure, [Customizing the Page Display](#).

Start Time	State	Call ID	Request URI	From URI	To URI	Ingress Realm	Egress Realm
2012-06-14 15:46:34.915	TERMINATED-200	5-17902@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:46:33.914	TERMINATED-200	4-17902@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:46:32.914	TERMINATED-200	3-17902@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:46:31.914	TERMINATED-200	2-17902@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:46:30.914	TERMINATED-200	1-17902@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:45:35.557	FAILED-408	5-17609@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:45:34.557	FAILED-408	4-17609@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:45:33.558	FAILED-408	3-17609@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:45:32.559	FAILED-408	2-17609@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:45:31.559	TERMINATED-0	1-17609@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:45:14.210	TERMINATED-200	5-17548@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:45:13.211	TERMINATED-200	4-17548@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core
2012-06-14 15:45:12.210	TERMINATED-200	3-17548@192.168.200.226	sip:test@192.168.204.71:5060	*+2273636* <tel:781-414-23...	sut <sip:kam@192.168.204.7...	access	core


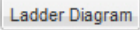
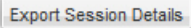
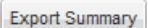
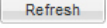
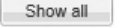
The following table describes the columns on this page.

Heading	Description
Start Time	Timestamp of the first SIP message in the call session.
State	Status of the call or media session. Valid values are: INITIAL Session for which an INVITE or SUBSCRIBE was forwarded. EARLY Session received the first provisional response (1xx other than 100). ESTABLISHED Session for which a success (2xx) response was received. TERMINATED Session that has ended by receiving or sending a BYE for an “Established” session or forwarding an error response for an “Initial” or Early session. The session remains in the terminated state until all the resources for the session are freed up. FAILED Session that has failed due to a 4xx or 5xx error code.
Call ID	Identification of the call source. Includes the phone number and source IP address.
Request URI	Uniform Resource Identifier (URI) formatted string that identifies a resource via a protocol, name, location, and any other applicable characteristic, and is sent by the Oracle Communications Session Border Controller in REQUEST headers.
To URI	URI formatted string that identifies the call destination information.
From URI	URI formatted string that identifies the call source information.
Ingress Realm	Incoming realm name.
Egress Realm	Outgoing realm name.
Duration	Amount of time, in seconds, that the call or media event was active.
Notable Event	Indicates if a notable event has occurred on the call session. Valid values are: short session - Sessions that don’t meet a minimum configurable duration threshold; Session dialogue, captured media information and termination signalling; Any event flagged as a short session interesting event local rejection - Sessions locally rejected at the Oracle Communications Session Border Controller for any reason (for example, Session Agent (SA) unavailable, no route found, SIP signalling error, etc.); Session dialogue, capture media information and termination signalling; Any event flagged as a local rejection interesting event

## Web-based GUI

Heading	Description
Session ID	Identification assigned to the call session.
Ingress Src Addr	Source IP address of the incoming call or media event.
Egress Dest Addr	Destination IP address of the outgoing call or media event.
Calling Pkts	Number of packets that occurred during outbound calling media traffic.
Called Pkts	Number of packets that occurred during inbound called media traffic.
Calling R Factor	Average Quality of Service (QoS) factor observed during the incoming call session. Quality of service shapes traffic to provide different priority and level of performance to different data flows. R-factors are metrics in VoIP, that use a formula to take into account both user perceptions and the cumulative effect of equipment impairments to arrive at a numeric expression of voice quality. This column defines the call or transmission quality expressed as an R factor.
Called RFactor	QoS factor observed during the outgoing call session. Quality of service shapes traffic to provide different priority and level of performance to different data flows. R-factors are metrics in VoIP, that use a formula to take into account both user perceptions and the cumulative effect of equipment impairments to arrive at a numeric expression of voice quality. This column defines the call or transmission quality expressed as an R factor.
Calling MOS	A measure of voice quality for an incoming media stream.
Called MOS	A measure of voice quality for an outgoing media stream.

The following table describes the buttons on this page.

Button	Description
	Performs a refresh of the page and updates it with the latest data.
	Displays a Ladder Diagram of a specific record in the table. The Ladder Diagram displays detailed information about a call session or media event.
	Exports the SIP messages and media events associated with the selected session, to a file in text format on the local machine.
	Exports all logged session summary records to a file in text format on the local machine.
	Allows you to specify parameters for performing a search for specific session summary records within the current report.
	Displays all of the session summary records in the Session Report.

For information about the Export buttons on this page, see [Exporting Information to a Text File](#).

## Session Report Configuration

To display a Session Report:

1. After logging into the Web-based GUI, click Sessions in the left column. The session summary report displays.
2. Use the buttons on the bottom of the page to view information about the records in this report.

For information about Refresh, see [Refresh](#).

For information about the Ladder Diagram, see [Ladder Diagram](#).

For information about exporting information, see [Exporting Information to a Text File](#).

## Ladder Diagram

Ladder diagrams in the Web-based GUI are logical schematics that show the call and media flow of packets on ingress and egress routes via the Oracle Communications Session Border Controller.

Ladder diagrams for the Session Report display the following session summary information:

- Quality of Service (QoS) statistics for call sessions
- SIP messages and media events in time sequence

To display a ladder diagram for a specific record in the Session Report, you can double-click a record in the summary table OR click **<Ladder Diagram>** on the Session Report page.

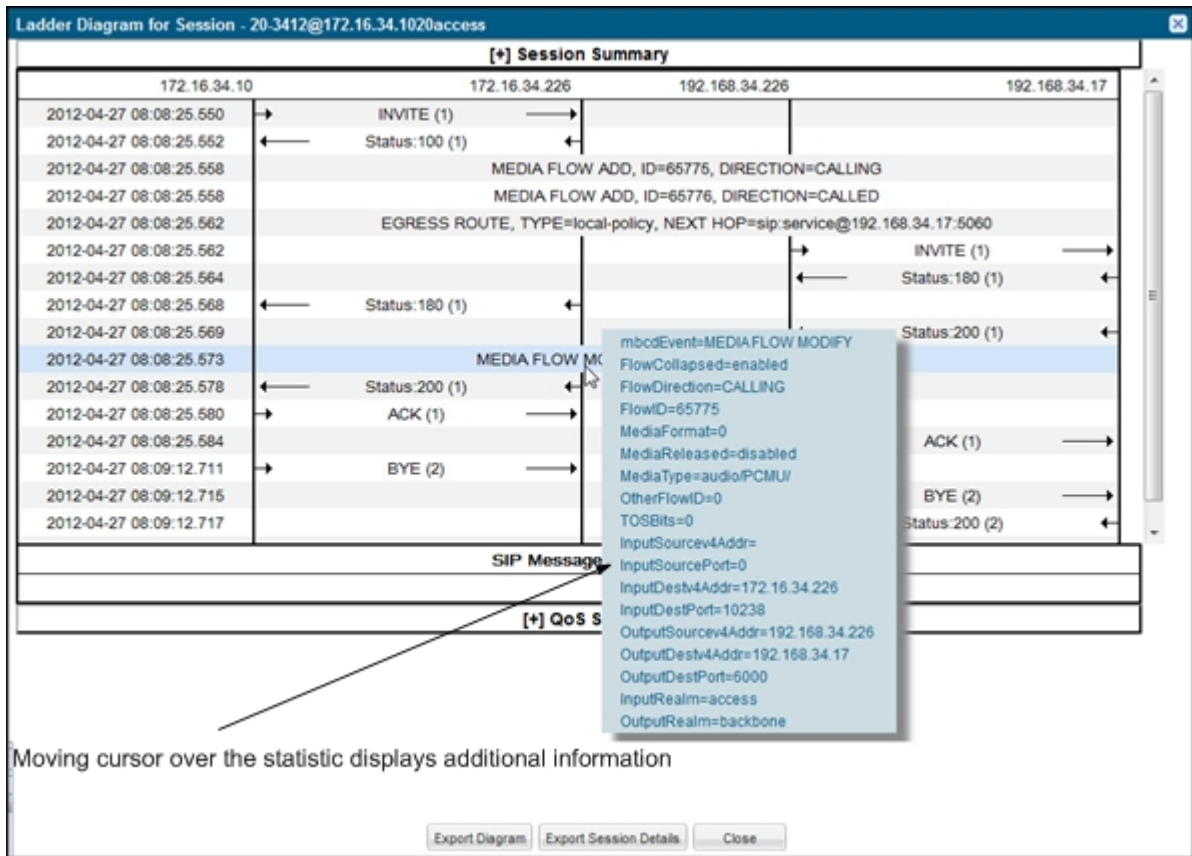
Double-click to display ladder diagram

Or click **<Ladder Diagram>**

Start Time	State	Call ID	Request URI	From URI	To URI	Ingress Realm	Egress Realm
2012-06-14 15:46:34.915	TERMINATED-200	5-17962@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:46:33.914	TERMINATED-200	4-17962@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:46:32.914	TERMINATED-200	3-17962@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:46:31.914	TERMINATED-200	2-17962@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:46:30.914	TERMINATED-200	1-17962@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:45:35.557	FAILED-498	5-17609@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:45:34.557	FAILED-498	4-17609@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:45:33.558	FAILED-498	3-17609@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:45:32.559	FAILED-498	2-17609@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:45:31.559	TERMINATED-0	1-17609@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:45:14.210	TERMINATED-200	5-17548@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:45:13.211	TERMINATED-200	4-17548@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core
2012-06-14 15:45:12.210	TERMINATED-200	3-17548@192.168.200.226	sip:test@192.168.204.71:5060	"2273636" <tel:781-414-23...	sut+ospkam@192.168.204.7...	access	core

To display a ladder diagram:

On the Sessions Report page, click **<Ladder Diagram>**. Alternatively, select a record in the table and double-click on that record. The following is an example of the ladder diagram that displays.



The Session Record Ladder Diagram consists of the following information:

- **Session Summary** - summary information about the call or media session in focus. For more information, see [Session Summary](#).
- **SIP Message Details** - SIP message and call flow information about the call or media session in focus. For more information, see [SIP Message Details](#).
- **QoS Statistics** - Quality of Service (QoS) statistic information about the call or media session in focus. For more information, see [QoS Statistics](#).

You can move your mouse over any statistic in the Ladder Diagram to view additional parameters and associated values for the statistic in a pop-up window.

The following table describes the buttons in this Ladder Diagram window.

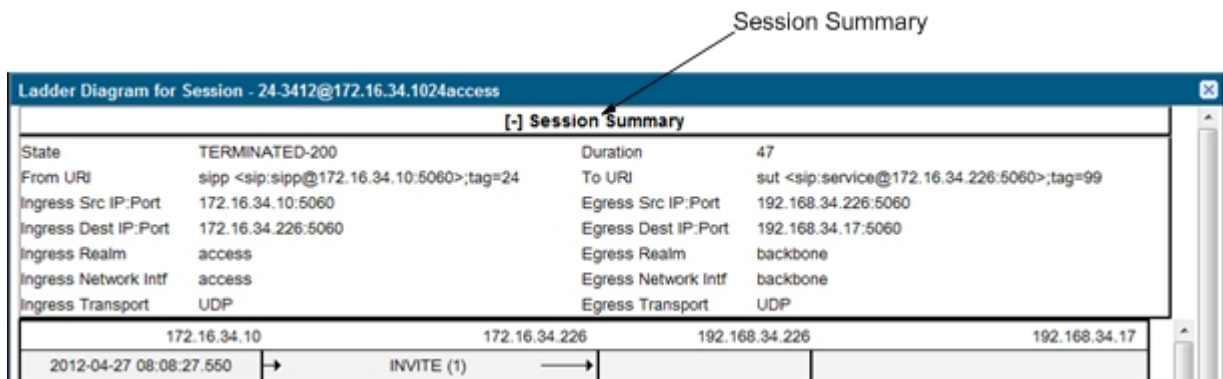
Button	Description
	Exports all of the information in the Ladder Diagram (Session Summary, SIP Message Details, and QoS statistics), to a file in text format on the local machine.
	Exports detailed information about the SIP messages and media events associated with the session in focus, to a file in text format on the local machine.
	Closes the Ladder Diagram window.

**Note:** For information about the Export buttons on this page, see [Exporting Information to a Text File](#).

### Session Summary

The Session Summary window in the Ladder Diagram displays an overall summary of the call or media session in focus.





To display the Session Summary:

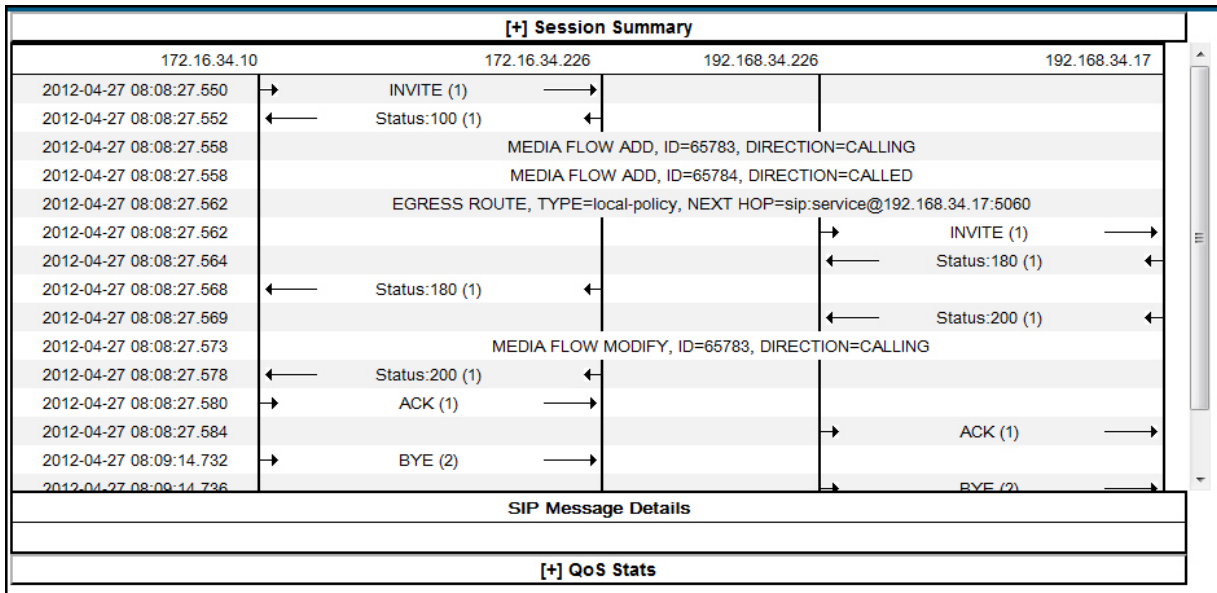
1. In the Ladder Diagram, click the **[+]** next to Session Summary at the top of the Ladder Diagram window. The Session Summary window expands. This window displays a summary of information about the call or media session in focus. The following table describes each field in the Session Summary window.

Heading	Description
State	Status of the call or media session. Valid values are: INITIAL Session for which an INVITE or SUBSCRIBE was forwarded. EARLY Session received the first provisional response (1xx other than 100). ESTABLISHED Session for which a success (2xx) response was received. TERMINATED Session that has ended by receiving or sending a BYE for an “Established” session or forwarding an error response for an “Initial” or Early session. The session remains in the terminated state until all the resources for the session are freed up. FAILED Session that has failed due to a 4xx or 5xx error code.
Duration	Amount of time, in seconds, that the call or media session was active.
From URI	URI formatted string that identifies the call source information.
To URI	URI formatted string that identifies the call destination information.
Ingress Src IP:Port	Source IP address and port number of the incoming call or media session.
Egress Src IP: Port	Source IP address and port number of the outgoing call or media session.
Ingress Dest IP:Port	Destination IP address and port number of the incoming call or media session.
Egress Dest IP: Port	Destination IP address and port number of the outgoing call or media session.
Ingress Realm	Incoming realm name.
Egress Realm	Outgoing realm name.
Ingress Network Intf	Name of the incoming network interface on the Oracle Communications Session Border Controller.
Egress Network Intf	Name of the outgoing network interface on the Oracle Communications Session Border Controller.
Ingress Transport	Protocol type used on the incoming call or media session. Valid values are User Datagram Protocol (UDP) or Transport Control Protocol (TCP).
Egress Transport	Protocol type used on the outgoing call or media session. Valid values are User Datagram Protocol (UDP) or Transport Control Protocol (TCP).

2. Click [-] to close the Session Summary window.

**SIP Message Details**

The SIP Message Detail window displays detailed information and data flow (ingress and egress) about the call or media event.



To display SIP Message Details:

On the Sessions Report page, click <Ladder Diagram>.

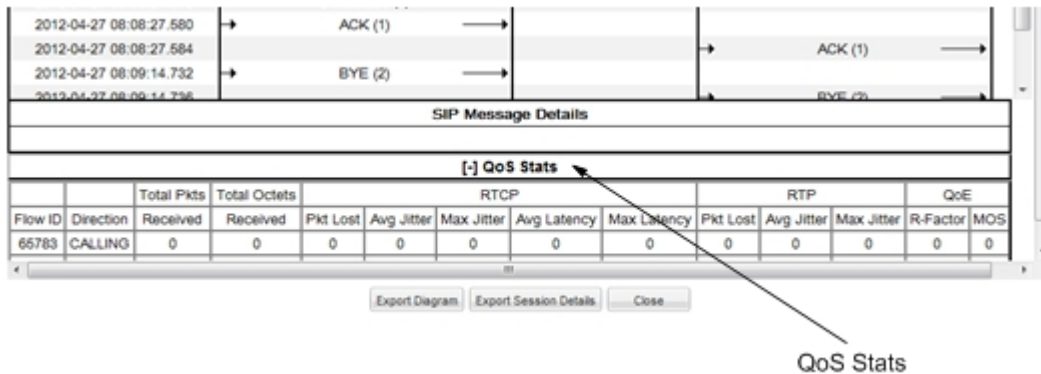
or

Select a record in the table and double-click on that record. The SIP Message Details window displays.

This window displays the messages and status codes that occurred during the active call session or media event. You can use the information to troubleshoot calls/media events that failed or timed out when trying to connect.

**QoS Statistics**

The Quality of Service (QoS) window displays information about the quality of the service used on the call session or media event when the call or event was active.



To display QoS Statistics:



1. In the Ladder Diagram, click the [+] next to QoS Stats at the bottom of the Ladder Diagram window. The QoS window expands. This window displays the QoS statistics for the call session or media event in focus. The following table describes each field in the QoS Statistics window.


Heading	Description
Flow ID	ID number assigned to the call session or media event flow of data.
Direction	The direction of the call or media event flow. Valid values are: CALLING (egress direction) CALLED (ingress direction)
Total Pkts Received	Total number of data packets received on the interface during the active call session or media event.
Total Octets Received	Total number of octets received on the interface during the active call session or media event. An octet is a unit of digital information that consists of eight bits.
RTCP	Real-time Transport Control Protocol - used to send control packets to participants in a call.
Pkts Lost	Number of RTCP data packets lost on the interface during the active call session or media event.
Avg Jitter	Average measure of the variability over time of the RTCP packet latency across a network. A network with constant latency has no variation (or jitter). Jitter is referred to as Packet Delay Variation (PDV). It is the difference in the one-way end-to-end delay values for packets of a flow. Jitter is measured in terms of a time deviation from the nominal packet interarrival times for successive packets.
Max Jitter	Maximum measure of the variability over time of the RTCP packet latency across a network. A network with constant latency has no variation (or jitter).
Avg Latency	Average observed one-way signaling latency during the active window period. This is the average amount of time the signaling travels in one direction.
Max Latency	Maximum observed one-way signaling latency during the sliding window period. This is the maximum amount of time the signaling travels in one direction.  Note: For more information about a sliding window period, see the <i>Oracle Communications Historical Data Recording Resource Guide</i> .
RTP	Real-Time Transport Protocol - a standard packet format for delivering audio and video over the internet.
Pkts Lost	Number of RTP data packets lost on the interface during the active call session or media event.
Avg Jitter	Average measure of the variability over time of the RTP packet latency across a network. A network with constant latency has no variation (or jitter). Jitter is referred to as Packet Delay Variation (PDV). It is the difference in the one-way end-to-end delay values for packets of a flow. Jitter is measured in terms of a time deviation from the nominal packet interarrival times for successive packets.
Max Jitter	Maximum measure of the variability over time of the RTP packet latency across a network. A network with constant latency has no variation (or jitter).
QoE	Quality of Experience - measurement used to determine how well the network is satisfying the end user's requirements.
R-Factor	Average Quality of Service (QoS) factor observed during the active window period. Quality of service shapes traffic to provide different priority and level of

Heading	Description
	performance to different data flows. R-factors are metrics in VoIP, that use a formula to take into account both user perceptions and the cumulative effect of equipment impairments to arrive at a numeric expression of voice quality. This statistic defines the call or transmission quality expressed as an R factor.
MOS	Mean Opinion Score (MOS) score. MOS is a measure of voice quality. MOS gives a numerical indication of the perceived quality of the media received after being transmitted and eventually compressed using Codecs.

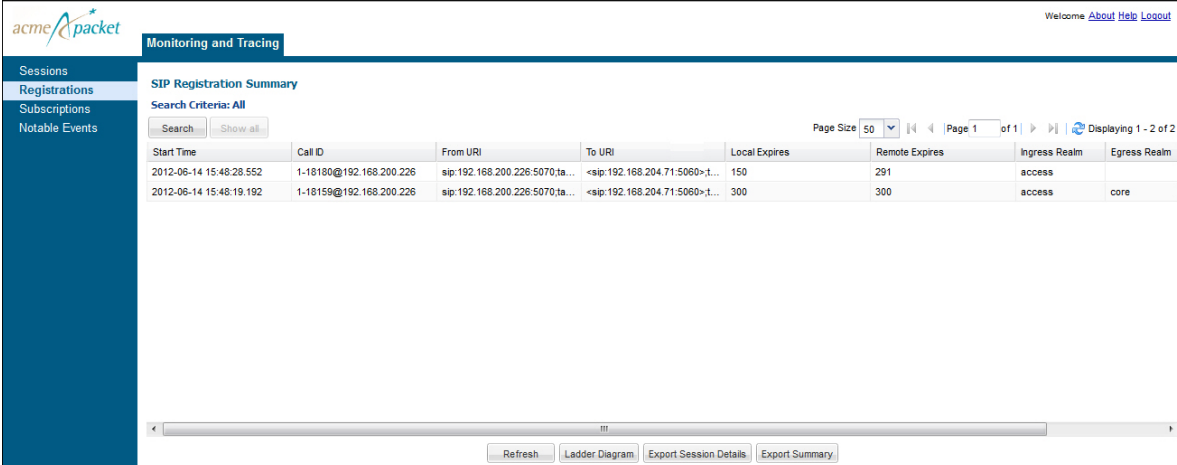
2. Click [-] to close the QoS Stats window.

## Registration Reports

The Registration Report is a summary of all logged SIP registrations sessions on the Oracle Communications Session Border Controller. The data that displays in this table is dependent on the filters set via the ACLI on the Oracle Communications Session Border Controller.

 **Note:** For setting filters to capture applicable data, see Filters to Configure.

The columns that display on the Registration Report page are dependent on the columns you selected in the procedure, [Customizing the Page Display](#).



The screenshot shows the 'acme packet' web interface. The main content area is titled 'SIP Registration Summary' and contains a table with the following data:

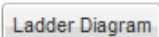
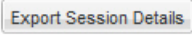
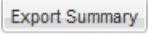
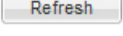
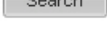
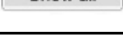
Start Time	Call ID	From URI	To URI	Local Expires	Remote Expires	Ingress Realm	Egress Realm
2012-06-14 15:48:28.552	1-18180@192.168.200.226	sip:192.168.200.226:5070.ta...	<sip:192.168.204.71:5060>t...	150	291	access	
2012-06-14 15:48:19.192	1-18159@192.168.200.226	sip:192.168.200.226:5070.ta...	<sip:192.168.204.71:5060>t...	300	300	access	core

The following table describes the columns on this page.

Heading	Description
Start Time	Timestamp of the first SIP message in the call session.
Call ID	Identification of the call source. Includes the phone number and source IP address.
To URI	URI formatted string that identifies the call destination information.
From URI	URI formatted string that identifies the call source information.
Local Expires	The current setting for the expiration of a registration request sent from the Integrated Media Gateway (IMG) to a Remote SIP User Agent. The default is 3600 sec.
Remote Expires	The current setting for the expiration of a registration request sent from the Remote SIP User Agent to the Integrated Media Gateway (IMG). The default is 3600 sec.
Ingress Realm	Incoming realm name.

Heading	Description
Egress Realm	Outgoing realm name.
Notable Event	Indicates if a notable event has occurred on the call session. Valid value is: local rejection - Sessions locally rejected at the Oracle Communications Session Border Controller for any reason (for example, Session Agent (SA) unavailable, no route found, SIP signalling error, etc.); Session dialogue, capture media information and termination signalling; Any event flagged as a local rejection interesting event
Session ID	Identification assigned to the call session.
Ingress Src Addr	Source IP address of the incoming call or media event.
Egress Dest Addr	Destination IP address of the outgoing call or media event.
Request URI	Uniform Resource Identifier (URI) formatted string that identifies a resource via a protocol, name, location, and any other applicable characteristic, and is sent by the system in REQUEST headers.

The following table describes the buttons on this page.


Button	Description
	Displays a Ladder Diagram of a specific record in the table. The Ladder Diagram displays detailed information about a call registration.
	Exports the call registration information associated with the selected record, to a file in text format on the local machine.
	Exports all logged call registration records to a file in text format on the local machine.
	Performs a refresh of the page and updates it with the latest data.
	Allows you to specify parameters for performing a search for specific call registration records within the current report.
	Displays all of the call registration records in the Registrations Report.

To display a Registrations Report:

1. After logging into the Web-based GUI, click Registrations in the left column. The registrations summary report displays.
2. Use the buttons on the bottom of the page to view information about the records in this report. For information about Refresh, see [Refresh](#). For information about the Ladder Diagram, see [Ladder Diagram](#). For information about exporting information, see [Exporting Information to a Text File](#).

## Subscription Reports

The Subscription Report is a summary of all logged SIP subscription sessions on the Oracle Communications Session Border Controller. The data that displays in this table is dependent on the filters set via the ACLI on the Oracle Communications Session Border Controller.

 **Note:** For setting filters to capture applicable data, see [Filters to Configure](#).

The columns that display on the Subscription Report page are dependent on the columns you selected in the procedure, [Customizing the Page Display](#).

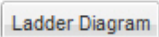
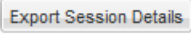
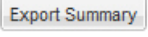
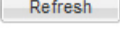
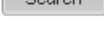

The screenshot shows the 'Monitoring and Tracing' section of the acme packet GUI. It displays a 'SIP Subscription Summary' table with columns for Start Time, Call ID, From URI, To URI, Events, Expires, Ingress Realm, Egress Realm, and Notable Event. The table contains 13 rows of data, all showing 'access' events in the 'core' realm. The interface includes a search bar, a 'Show all' button, and pagination controls indicating 'Page 1 of 1' and 'Displaying 1 - 13 of 13'.

The following table describes the columns on this page.

Heading	Description
Start Time	Timestamp of the first SIP message in the call session.
Call ID	Identification of the call source. Includes the phone number and source IP address.
From URI	URI formatted string that identifies the call source information.
To URI	URI formatted string that identifies the call destination information.
Events	<p>Specific subscribe event package that was sent from an endpoint to the destination endpoint. Applicable event packages can be:</p> <ul style="list-style-type: none"> <li>conference - Event package that allows users to subscribe to a conference Uniform Resource Identifier (URI).</li> <li>consent-pending additions - Event package used by SIP relays to inform user agents about the consent-related status of the entries to be added to a resource list.</li> <li>dialog - Event package that allows users to subscribe to another user, and receive notifications about the changes in the state of the INVITE-initiated dialogs in which the user is involved.</li> <li>kpml - Event package that enables monitoring of dual-tone multi-frequency (DTMF) signals, and uses XML documents called Key Press Markup Language (KPML).</li> <li>message-summary - Event package that carries message-waiting status and message summaries from a messaging system to an interested User Agent (UA).</li> <li>presence - Event package that conveys the ability and willingness of a user to communicate across a set of devices. A presence protocol is a protocol for providing a presence service over the Internet or any IP network.</li> <li>reg - Event package that provides a way to monitor the status of *all* the registrations for a particular Address of Record (AoR).</li> <li>refer - Event package that provides a mechanism to allow the party sending the REFER to be notified of the outcome of a referenced request.</li> <li>.winfo - Event package for watcher information. It tracks the state of subscriptions to a resource in another package.</li> <li>vq-rtcp - Event package that collects and reports the metrics that measure quality for RTP sessions.</li> </ul>

Heading	Description
Local Expires	The current setting for the expiration of a registration request sent from the Integrated Media Gateway (IMG) to a Remote SIP User Agent. The default is 3600 sec.
Remote Expires	The current setting for the expiration of a registration request sent from the Remote SIP User Agent to the Integrated Media Gateway (IMG). The default is 3600 sec.
Ingress Realm	Incoming realm name.
Egress Realm	Outgoing realm name.
Notable Event	Indicates if a notable event has occurred on the call session. Valid value is: local rejection - Sessions locally rejected on the system for any reason (for example, Session Agent (SA) unavailable, no route found, SIP signalling error, etc.); Session dialogue, capture media information and termination signalling; Any event flagged as a local rejection interesting event
Session ID	Identification assigned to the call session.
Ingress Src Addr	Source IP address of the incoming call or media event.
Egress Dest Addr	Destination IP address of the outgoing call or media event.
Request URI	Uniform Resource Identifier (URI) formatted string that identifies a resource via a protocol, name, location, and any other applicable characteristic, and is sent by the system in REQUEST headers.

The following table describes the buttons on this page.


Button	Description
	Displays a Ladder Diagram of a specific record in the table. The Ladder Diagram displays detailed information about a call subscription.
	Exports the subscription information associated with the selected record, to a file in text format on the local machine.
	Exports all logged subscription records to a file in text format on the local machine.
	Performs a refresh of the page and updates it with the latest data.
	Allows you to specify parameters for performing a search for specific subscription records within the current report.
	Displays all of the subscription records in the Subscriptions Report.

To display a Subscriptions Report:

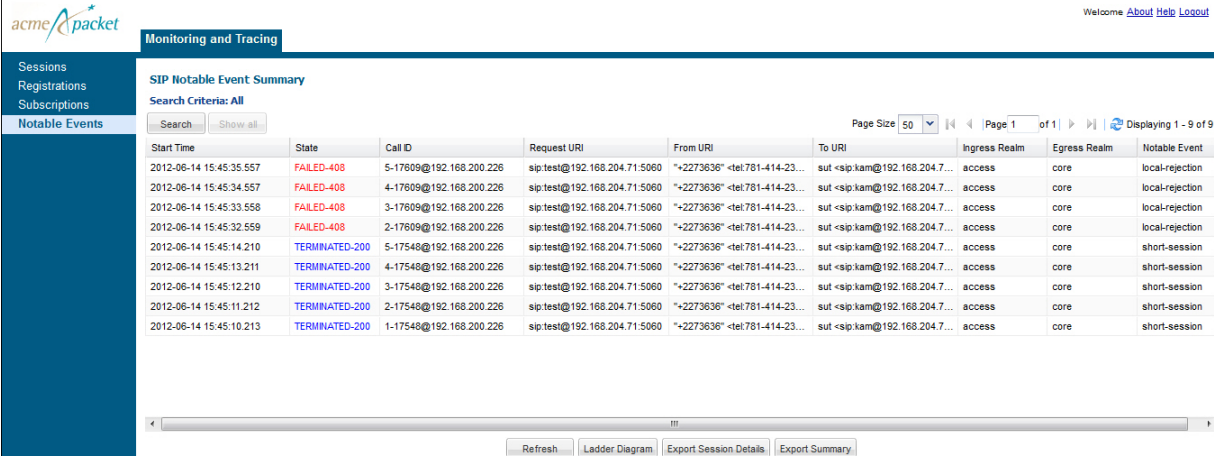
1. After logging into the Web-based GUI, click Subscriptions in the left column. The subscriptions summary report displays.
2. Use the buttons on the bottom of the page to view information about the records in this report. For information about Refresh, see [Refresh](#). For information about the Ladder Diagram, see [Ladder Diagram](#). For information about exporting information, see [Exporting Information to a Text File](#).

## Notable Event Reports

The Notable Events Report contains all logged sessions that have a notable event associated with the session on the Oracle Communications Session Border Controller. The data that displays in this table is dependent on the filters set via the ACLI on the Oracle Communications Session Border Controller.

 **Note:** For setting filters to capture applicable data, see [Filters to Configure sessions that have a nota.](#)

The columns that display on the Notable Events Report page are dependent on the columns you selected in the procedure, [Customizing the Page Display.](#)



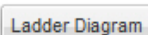

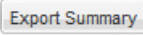
Start Time	State	Call ID	Request URI	From URI	To URI	Ingress Realm	Egress Realm	Notable Event
2012-06-14 15:45:35.557	FAILED-408	5-17609@192.168.200.226	sip.test@192.168.204.71:5060	*+2273636* <tel781-414-23...	sut <sp.kam@192.168.204.7...	access	core	local-rejection
2012-06-14 15:45:34.557	FAILED-408	4-17609@192.168.200.226	sip.test@192.168.204.71:5060	*+2273636* <tel781-414-23...	sut <sp.kam@192.168.204.7...	access	core	local-rejection
2012-06-14 15:45:33.558	FAILED-408	3-17609@192.168.200.226	sip.test@192.168.204.71:5060	*+2273636* <tel781-414-23...	sut <sp.kam@192.168.204.7...	access	core	local-rejection
2012-06-14 15:45:32.559	FAILED-408	2-17609@192.168.200.226	sip.test@192.168.204.71:5060	*+2273636* <tel781-414-23...	sut <sp.kam@192.168.204.7...	access	core	local-rejection
2012-06-14 15:45:14.210	TERMINATED-200	5-17548@192.168.200.226	sip.test@192.168.204.71:5060	*+2273636* <tel781-414-23...	sut <sp.kam@192.168.204.7...	access	core	short-session
2012-06-14 15:45:13.211	TERMINATED-200	4-17548@192.168.200.226	sip.test@192.168.204.71:5060	*+2273636* <tel781-414-23...	sut <sp.kam@192.168.204.7...	access	core	short-session
2012-06-14 15:45:12.210	TERMINATED-200	3-17548@192.168.200.226	sip.test@192.168.204.71:5060	*+2273636* <tel781-414-23...	sut <sp.kam@192.168.204.7...	access	core	short-session
2012-06-14 15:45:11.212	TERMINATED-200	2-17548@192.168.200.226	sip.test@192.168.204.71:5060	*+2273636* <tel781-414-23...	sut <sp.kam@192.168.204.7...	access	core	short-session
2012-06-14 15:45:10.213	TERMINATED-200	1-17548@192.168.200.226	sip.test@192.168.204.71:5060	*+2273636* <tel781-414-23...	sut <sp.kam@192.168.204.7...	access	core	short-session

The following table describes the columns on this page.

Heading	Description
Start Time	Timestamp of the first SIP message in the call session.
State	Status of the call or media event session. Valid values are: INITIAL Session for which an INVITE or SUBSCRIBE was forwarded. EARLY Session received the first provisional response (1xx other than 100). ESTABLISHED Session for which a success (2xx) response was received. TERMINATED Session that has ended by receiving or sending a BYE for an “Established” session or forwarding an error response for an “Initial” or Early session. The session remains in the terminated state until all the resources for the session are freed up. FAILED Session that has failed due to a 4xx or 5xx error code.
Call ID	Identification of the call source. Includes the phone number and source IP address.
Request URI	Uniform Resource Identifier (URI) formatted string that identifies a resource via a protocol, name, location, and any other applicable characteristic, and is sent by the system in REQUEST headers.
To URI	URI formatted string that identifies the call destination information.
From URI	URI formatted string that identifies the call source information.
Events	Specific subscribe event package that was sent from an endpoint to the destination endpoint. Applicable event packages can be: conference - Event package that allows users to subscribe to a conference Uniform Resource Identifier (URI).


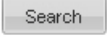
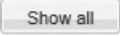
Heading	Description
	<p>consent-pending additions - Event package used by SIP relays to inform user agents about the consent-related status of the entries to be added to a resource list.</p> <p>dialog - Event package that allows users to subscribe to another user, and receive notifications about the changes in the state of the INVITE-initiated dialogs in which the user is involved.</p> <p>kpml - Event package that enables monitoring of dual-tone multi-frequency (DTMF) signals, and uses XML documents called Key Press Markup Language (KPML).</p> <p>message-summary - Event package that carries message-waiting status and message summaries from a messaging system to an interested User Agent (UA).</p> <p>presence - Event package that conveys the ability and willingness of a user to communicate across a set of devices. A presence protocol is a protocol for providing a presence service over the Internet or any IP network.</p> <p>reg - Event package that provides a way to monitor the status of *all* the registrations for a particular Address of Record (AoR).</p> <p>refer - Event package that provides a mechanism to allow the party sending the REFER to be notified of the outcome of a referenced request.</p> <p>.winfo - Event package for watcher information. It tracks the state of subscriptions to a resource in another package.</p> <p>vq-rtcpx - Event package that collects and reports the metrics that measure quality for RTP sessions.</p>
Ingress Realm	Incoming realm name.
Egress Realm	Outgoing realm name.
Notable Event	<p>Indicates if an interesting event has occurred on the call session. Valid values are:</p> <p>short session - A session is viewed as a "short session" if the length of time, in seconds, is equal to or below the short-session-duration value configured at the path Terminal-&gt;session-router-&gt;session-router-config-&gt;short-session-duration in the ACLI.</p> <p>local rejection - Sessions that are locally rejected at the Oracle Communications Session Border Controller for any reason (for example, Session Agent (SA) unavailable, no route found, SIP signalling error, etc.).</p>
Session ID	Identification assigned to the call session.
Ingress Src Addr	Source IP address of the incoming call or media event.
Egress Dest Addr	Destination IP address of the outgoing call or media event.

The following table describes the buttons on this page.

Button	Description
	Displays a Ladder Diagram of a specific record in the table. The Ladder Diagram displays detailed information about a notable event.
	Exports the notable event information associated with the selected record, to a file in text format on the local machine.
	Exports all logged notable event records to a file in text format on the local machine.



## Web-based GUI

Button	Description
	Performs a refresh of the page and updates it with the latest data.
	Allows you to specify parameters for performing a search for specific notable event records within the current report.
	Displays all of the notable event records in the Notable Events Report.

To display a Notable Events Report:

1. After logging into the Web-based GUI, click Notable Events in the left column. The notable events summary report displays.
2. Use the buttons on the bottom of the page to view information about the records in this report. For information about Refresh, see [Refresh](#). For information about the Ladder Diagram, see [Ladder Diagram](#). For information about exporting information, see [Exporting Information to a Text File](#).



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# Web Server Configuration

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## Introduction

The Oracle Communications Session Border Controller includes an embedded Web-based GUI that can display the SIP session data results from a filtered session. This GUI displays SIP sessions data on one or multiple Oracle Communications Session Border Controllers, and provides traces in a common log format for local viewing or for exporting to your PC.

This chapter provides the information required to prepare the Web Server for running the Web-based GUI.

Topics include:

- [Configuring the Web Server](#)
- [Configuring TLS for the Web Server](#)
- [Management Commands for the Web Server](#)

 **Note:** The web server may only be reached over an IPv4 address. IPv6-based web server is unsupported.

---

## Configuring the Web Server

To use the Web-based GUI via your browser, you must first enable the GUI using the ACLI.

Use the following procedure to configure the Web server:

To configure the Web server:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **system** and press Enter to access the system-related objects.

```
ACMEPACKET (configure)# system  
ACMEPACKET (system)#
```

3. Type **web-server-config** and press Enter to access the event monitoring-related attributes.

```
ACMEPACKET (system)# web-server-config  
ACMEPACKET (web-server-config)#
```

**state**—Enter whether or not to enable the Web-based GUI. Default is enabled. Valid values are:


## Web Server Configuration

- |                                                                     |
|---------------------------------------------------------------------|
| <ul style="list-style-type: none"><li>• enabled (default)</li></ul> |
| <ul style="list-style-type: none"><li>• disabled</li></ul>          |

```
ACMEPACKET(web-server-config) # state enabled
```

**inactivity-timeout** —Enter the amount of time, in minutes, that the Web-based GUI must have remained inactive before it ends Web session. For example, if this timeout value is set as 5, after 5 minutes of no activity, the Web session disconnects. Default is 5. Valid values are 0 to 20.

```
ACMEPACKET(web-server-config) # inactivity-timeout 5
```

 **Note:** The following `http-state`, `http-port`, `https-state`, and `https-port` parameters may have already been set via the Web-based GUI installation wizard on your Oracle Communications Session Border Controller. You can edit these parameters if required using the ACLI.

**http-state** —Enter whether or not to enable HTTP for accessing the Web server. Default is enabled. Valid values are:

- |                                                                     |
|---------------------------------------------------------------------|
| <ul style="list-style-type: none"><li>• enabled (default)</li></ul> |
| <ul style="list-style-type: none"><li>• disabled</li></ul>          |

```
ACMEPACKET(web-server-config) # http-state enabled
```

**http-port** —Enter the HTTP port to use to connect to the Web server. Default is 80. Valid values are 1 to 65535.

```
ACMEPACKET(web-server-config) # http-port 80
```

**https-state** —Enter whether or not to enable HTTPS (secure connection) for accessing the Web server. Default is disabled. Valid values are:

- |                                                                      |
|----------------------------------------------------------------------|
| <ul style="list-style-type: none"><li>• enabled</li></ul>            |
| <ul style="list-style-type: none"><li>• disabled (default)</li></ul> |

```
ACMEPACKET(web-server-config) # https-state enabled
```

**https-port** —Enter the HTTPS port to use to connect to the Web server. Default is 443. Valid values are 1 to 65535.

```
ACMEPACKET(web-server-config) # https-port 443
```

**tls-profile** —Enter the Transport Layer Security (TLS) Protocol profile name to use with HTTPS. Default is blank. Valid values are **alpha-numeric characters**.

```
ACMEPACKET(web-server-config) # tls-profile tlsSM&T
```

If you specify a `tls-profile`, and HTTP is enabled, the system checks against the TLS profile table for a match. If there is no match, the applicable errors display during the verification of the configuration.

To create a TLS profile, see [Configuring a TLS Profile](#).

4. Enter **exit** to exit the Web server configuration.

```
ACMEPACKET(web-server-config) # exit
```

5. Enter **exit** to exit the system configuration.

```
ACMEPACKET(system) # exit
```

6. Enter **exit** to exit the configure mode.

```
ACMEPACKET(configure) # exit
```

7. Enter **save-config** to save the configuration.


```
ACMEPACKET# save-config
```

8. Enter **activate-config** to activate as the current configuration.

```
ACMEPACKET# activate-config
```

## Configuring TLS for the Web Server

The SIP monitoring and tracing feature supports the use of HTTP over Transport Layer Security (TLS) using the TLS Protocol. TLS is a cryptographic protocol that provides communication security over the Internet. It encrypts the segments of network connections at the Transport Layer, using asymmetric cryptography for key exchange, symmetric encryption for privacy, and message authentication codes for message integrity.

 **Note:** For more information about setting up security on your Oracle Communications Session Border Controller, refer to the Security section of the *ACLI Configuration Guide*.

To use TLS with SIP monitoring and tracing, you must configure a TLS certificate and a TLS profile using the ACLI at the path Configure Terminal->Security. This configuration stores the information required to run SIP over TLS.

If you enable TLS on the active system, the Web-based GUI interface on the standby system is disabled.

### Process Overview

In summary, you need to take the following steps to enable your Oracle Communications Session Border Controller for TLS.

1. Make sure that your Oracle Communications Session Border Controller has the appropriate hardware installed and that you have obtained an enabled the licenses related to TLS support. (Note that the Acme Packet 4250 does not require an additional license for TLS support.)
2. Configure certificates.
3. Configure the specific parameters related to TLS.

### Configuring Certificates


Configuring certificates is a three-step process:

1. Create a certificate record configuration on the Oracle Communications Session Border Controller
2. Generate a certificate request by the Oracle Communications Session Border Controller and save the configuration
3. Import the certificate record into the Oracle Communications Session Border Controller and save the configuration

#### Configuring the Certificate Record

The certificate record configuration represents either the end-entity or the Certificate Authority (CA) certificate on the Oracle Communications Session Border Controller. If it is used to present an end-entity certificate, a private key should be associated with this certificate record configuration using the ACLI security certificate request command.

No private key should be associated with the certificate record configuration if it was issued to hold a CA certificate. A certificate can be imported to a certificate record configuration using the ACLI security certificate import command.

 **Note:** There is no need to create a certificate record when importing a CA certificate or certificate in pkcs12 format.

To configure a certificate:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **security** and press Enter to access the security-related objects.

```
ACMEPACKET (configure) # security
```

3. Type **certificate-record** and press Enter to access the certificate record parameters.

```
ACMEPACKET (security) # certificate-record
ACMEPACKET (certificate-record) #
```

**name**—Enter the name of the certificate record. This is a key field, and you must enter a value for it. For example, acmepacket.

**country**—Enter the name of the country. The default is US.

**state**—Enter the name of the state of for the country. The default is MA.

**locality**—Enter the name of the locality for the state. The default is Burlington.

**organization**—Enter the name of the organization holding the certificate. The default is Engineering.

**unit**—Enter the name of the unit for the holding the certificate within the organization.

**common-name**—Enter the common name for the certificate record.

**key-size**—Enter the size of the key for the certificate. Use the default of 1024, or change it to one of the other supported values: 512, 2048, or 4096.

**alternate-name**—Enter the alternate name of the certificate holder.

**key-usage-list**—Enter the usage extensions you want to use with this certificate record. This parameter can be configured with multiple values, and it defaults to the combination of digitalSignature and keyEncipherment. For a list of possible values and their descriptions, see the section Key Usage Control in the *ACLI Configuration Guide*.

**extended-key-usage-list**—Enter the extended key usage extensions you want to use with this certificate record. The default is serverAuth. For a list of possible values and their descriptions, see the section Key Usage Control in the *ACLI Configuration Guide*.

4. Enter **done** to save the certificate-record configuration.

```
ACMEPACKET (certificate-record) # done
```

5. Enter **exit** to exit the certificate-record configuration.

```
ACMEPACKET (certificate-record) # exit
```

6. Enter **y** at the prompt to save the configuration.

```
Save Changes [y|n]?: y
```

7. Enter **exit** to exit the security configuration.

```
ACMEPACKET (security) # exit
```

8. Enter **exit** to exit the configure mode.


```
ACMEPACKET (configure) # exit
```

9. Enter **save-config** to save the configuration.

```
ACMEPACKET# save-config
```


10. Enter **activate-config** to activate as the current configuration.

```
ACMEPACKET# activate-config
```

 **Note:** For verifying a certificate record, see the Security section of the *ACLI Configuration Guide*.

### Generating a Certificate Request

Using the ACLI security certificate request command allows you to generate a private key and a certificate request in PKCS10 PEM format.

 **Note:** You can only perform this step once you have configured a certificate record.

The Oracle Communications Session Border Controller stores the private key that is generated in the certificate record configuration in 3DES encrypted form with an internally generated password. The PKCS10 request is displayed on the screen in PEM (Base64) form.

You use this command for certificate record configurations that hold end-entity certificates. If you have configured the certificate record to hold a CA certificate, then you do not need to generate a certificate request because the CA publishes its certificate in the public domain. You import a CA certificate by using the ACLI **security certificate request import** command.

This command sends information to the CA to generate the certificate, but you cannot have Internet connectivity from the Oracle Communications Session Border Controller to the Internet. You can access the internet through a browser such as Internet Explorer if it is available, or you can save the certificate request to a disk and then submit it to the CA.

To run the applicable command, you must use the value you entered in the name parameter of the certificate record configuration. You run the command from the main Superuser mode command line, and then save and activate the configuration.

```
ACMEPACKET# security certificate request acmepacket
Generating Certificate Signing Request. This can take several
minutes....
-----BEGIN CERTIFICATE REQUEST-----
MIIB2jCCAUMCAQAwYTELMAkGAlUEBhMCdXMxCzAJBgNVBAGTAklBMRMwEQYDVQQH
EwpCdXJsaW5ndG9uMRQwEgYDVQQKEwtFbmdpbmVlcmluZzEMMAoGA1UECzMdYWJj
MQwwCgYDVQQDEwNhYmMwZ8wDQYJKoZIhvcNAQEBBQADgY0AMIGJAoGBALOMLHo8
/qIOddIDVuqot0Y72l/BfH8lolRkmhZQ4e7sS+zZHzbG8phzmzhfOSECnZiA2bEo
f+Nti7e7Uof4lLwiYl9fvhURfzhENOKThAPKPiJCzBBglTITHTYa100Cq2fj5A8B
ZcuAHj7Vp5wP2zpz6EUTFpqTDMLVdwJGJrElAgMBAAGgOTAMBgNVHRExBRMDZGVm
MCkGAlUdDzEiEYBkaWdpdGFsU2lnbmF0dXJlLGtleUVuY2lwaGVybWVudDANBgkq
hkiG9w0BAQUFAAOBgQAtel4ZSLI8gggMzodbYwgUHUGqTGeDzQDhJV5fKUXWemFz
JsTmWn5Gy/kR4+Nq274G14fnk00fTafMtgQ5aL3gM43TqaPOTZjJ6qgwuRKhobPI
7hkovkgAxHge7wClghiAp/ELdl7tQ515k04Bmd5f/fxG7nNiu8iEg7P000IBgg==
-----END CERTIFICATE REQUEST-----
WARNING: Configuration changed, run "save-config" command.
ACMEPACKET# save config
copying file /code/config/dataDoc.gz -> /code/config/dataDoc_3.gz
copying file /code/config/tmp/editing/dataDoc.gz ->
/code/config/dataDoc.gz
Save complete
ACMEPACKET# activate config
activate complete
```

### Importing a Certificate Using the ACLI

For an end-entity certificate, once a certificate is generated using the ACLI security certificate request command, that request should be submitted to a CA for generation of a certificate in PKCS7 or X509v3 format. When the certificate has been generated, you can import it into the Oracle Communications Session Border Controller using the security certificate import command.

The syntax is:

```
ORACLE # security certificate import [try-all | pkcs7 | pkcs12 |
x509] [certificate-record file-name]
```

To import a certificate:

1. When you use the security certificate import command, you can specify whether you want to use PKCS7, PKCS12, X509v3 format, or try all. In the command line, you enter the command, the format specification, and the name of the certificate record. Next, the system will prompt you to enter the certificate in PEM format. Paste the certificate in the ACLI at this point. For example:

```
ACMEPACKET# security certificate import try-all acmepacket
```

The following displays:

## Web Server Configuration

```
Please enter the certificate in the PEM format.
  Terminate the certificate with ";" to exit.....
  -----BEGIN CERTIFICATE-----

VMIIDHzCCAoigAwIBAgIIAhMCUACEAHEwDQYJKoZIhvcNAQEFBQAwdELMAkGA1UE
BhMCMCVVMxEzARBgNVBAGTCkNhbGlm3JuaWEExETAPBgNVBACTCFNhbiBkb3NlMQ4w
DAYDVQQKEwVzaXBpdDEpMCCGA1UECXMgU2lwaXQgVGZzdCBDZXJ0aWZpY2F0ZSBB
dXR0b3JpdHkwHhcNMDUwNDEzMjEzNzQzWhcNMDgwNDEyMjEzNzQzWjBUMQswCQYD
VQQGEwJVUzELMAkGA1UECBMCTUExEzARBgNVBACTCk1cmxpbmd0b24xFDASBgNV
BAoTC0VuZ2luZWVyaW5nMQ0wCwYDVQQDEwRhY21lMIGfMA0GCSqGSIb3DQEBAQUA
    A4GNADCBiQKBgQCXjIeOyFKAUB3rKkKk/+59LT+r1GuW7Lgc1V6+hfTSr0co
+ZsQ
bHFUWAA15qXUUBTLJG13QN5Vfg96f7gGAbWayfOS9Uymold3JPCUDoGgb2E7m8iu
    vtq7gwjSeKNXAw/y7yWy/
c04FmUD2U0pZX0CNIR3Mns5OAxQmq0bNYDhawIDAQAB
o4HdMIHaMBEGA1UdEQQKMAiCBnBrdW1hcjAJBgNVHRMEAjAAMB0GA1UdDgQWBBTG
tpodxa6Kmmn04L3Kg62t8BZJHTCBmgYDVR0jBIGSMIGPgBRrRhcU6pR2JYBUbNU
2qHjVBShtqF0pHIwcDELMAkGA1UEBhMCMCVVMxEzARBgNVBAGTCkNhbGlm3JuaWEEx
ETAPBgNVBACTCFNhbiBkb3NlMQ4wDAYDVQQKEwVzaXBpdDEpMCCGA1UECXMgU2lwa
aXQgVGZzdCBDZXJ0aWZpY2F0ZSBBdXR0b3JpdHhMAQAwDQYJKoZIhvcNAQEFBQAD
    gYEAbEs8nUCi+cA2hC/lM49Sitvh8QmpL81KONApsoC4Em24L
+DZwz3uInoWjbjJ
QhefcUfteNYkbuMH7LAK0hnDPvW+St4rQGvK6LJhZj7/
yeLXmYWIPUY3Ux40GVrd
    2UgV/B2SOqH9Nf+FQ+mNZ0lL7EuF4IxSz9/69LuYlXqKsG4=
-----END CERTIFICATE-----;
Certificate imported successfully....
WARNING: Configuration changed, run "save-config" command.
```

2. Enter **save-config** to save the configuration.

```
ACMEPACKET# save-config
copying file /code/config/dataDoc.gz ->
/code/config/dataDoc_3.gz
copying file /code/config/tmp/editing/dataDoc.gz ->
/code/config/dataDoc.gz
Save complete
```

3. Enter **activate-config** to activate as the current configuration.

```
ACMEPACKET# activate-config
activate complete
```



**Note:** For importing a certificate using FTP, see the Security section of the *ACLI Configuration Guide*.

## Importing a Certificate Using FTP

You can also put the certificate file in the directory /opt and then execute the **import-certificate** command, or you can paste the certificate in PEM/Base64 format into the ACLI. If you paste the certificate, you may have to copy and paste it a portion at a time, rather than pasting the whole certificate at once.

To import the certificate using FTP:

1. FTP the certificate file on to the system (directory /opt), let us say the name of the certificate file is cert.pem.
2. Once the certificate is successfully transferred to the system, run the **import-certificate** command.

The syntax is:

```
ORACLE# import-certificate [try-all|pkcs7|x509] [certificate-record file-name]
```

Using the command will look like this when you have used FTP.

```
ORACLE# import-certificate try-all acme cert.pem
Certificate imported successfully...
WARNING: Configuration changed, run "save-config" command.
```

1. Save your configuration.

```
ORACLE# save-config
Save-Config received, processing.
waiting 1200 for request to finish
Request to 'SAVE-CONFIG' has Finished,
Save complete
Currently active and saved configurations do not match!
To sync & activate, run 'activate-config' or 'reboot activate'.
```

2. Synchronize and activate your configurations.

```
ORACLE# activate-config
Activate-Config received, processing.
waiting 120000 for request to finish
Add LI Flows
LiSysClientMgr::handleNotifyReq
H323 Active Stack Cnt: 0
Request to 'ACTIVATE-CONFIG' has Finished,
Activate Complete
ACMEPACKET#
```

## Configuring a TLS Profile

To configure a TLS profile:

1. In Superuser mode, type **configure terminal** and press Enter.

```
ACMEPACKET# configure terminal
```

2. Type **security** and press Enter to access the security-related objects.

```
ACMEPACKET(configure)# security
```

3. Type **tls-profile** and press Enter to access the TLS profile-related parameters.

```
ACMEPACKET(security)# tls-profile
ACMEPACKET(tls-profile)#
```

**name**—Enter the name of the TLS profile. This parameter is required; you cannot leave it empty.

```
ACMEPACKET(tls-profile)# name tls-prof1
```


**end-entity-certificate**—Enter the name of the entity certification record.

```
ACMEPACKET(tls-profile)# end-entity-certificate cert1
```

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**trusted-ca-certificates**—Enter the names of the trusted CA certificate records.

```
ACMEPACKET(tls-profile)# trusted-ca-certificates cert1
```

 **Note:** To create and import certificate records to be used on the Web Server, see [Configuring Certificates](#).

**cipher-list**—Not supported for SIP monitoring and tracing. The Session Director ignores any value you enter for this parameter.

- AES256-SHA (**TLS\_RSA\_WITH\_AES\_256\_CBC\_SHA**) - Firefox (version 12) and Chrome (version 19.0.1084.46m) only
- AES128-SHA (**TLS\_RSA\_WITH\_AES\_128\_CBC\_SHA**) - Firefox (version 12) and Chrome (version 19.0.1084.46m) only
- DES-CBC-SHA (**SSL\_RSA\_WITH\_DES\_CBC\_SHA** or **TLS\_RSA\_WITH\_DES\_CBC\_SHA**) - Internet Explorer (Version 9) only

**verify-depth**—Not supported for SIP monitoring and tracing

**mutual-authenticate**—Not supported for SIP monitoring and tracing

**tls-version**—Enter the TLS version you want to use with this TLS profile. Default is compatibility. Valid values are:

- TLSv1
- SSLv3
- compatibility (default)

```
ACMEPACKET(tls-profile)# tls-version TLSv1
```

**cert-status-check**—Not supported for SIP monitoring and tracing

**cert-status-profile-list**—Not supported for SIP monitoring and tracing

**ignore-dead-responder**—Not supported for SIP monitoring and tracing

4. Enter **done** to save the tls-profile configuration.

```
ACMEPACKET(tls-profile)# done
```

5. Enter **exit** to exit the TLS profile configuration.

```
ACMEPACKET(tls-profile)# exit
```

6. Enter **exit** to exit the security configuration.

```
ACMEPACKET(security)# exit  
ACMEPACKET(configure)#
```

7. Enter **exit** to exit the configure mode.

```
ACMEPACKET(configure)# exit
```

8. Enter **save-config** to save the configuration.

```
ACMEPACKET# save-config
```

9. Enter **activate-config** to activate as the current configuration.

```
ACMEPACKET# activate-config
```

## Management Commands for the Web Server

The following commands allow you to display information for managing the Web server used for accessing the Web-based GUI.

Command	Description
show ip connections	Displays information about the server connections.



Command	Description
show users	Displays information about users logged into a session on the server.
kill <index>	Terminates a session on the server.

## Show ip connections Command

The show ip connections command allows you to display information about active server Transport Control Protocol (TCP) and/or User Datagram Protocol (UDP) connections. For example, this command can show the sockets tied to an HTTPS connection. The following is an example of the show ip connections command output.

```

ACMEPACKET# show ip connections
Active Internet connections (including servers)
PCB      Proto Recv-Q Send-Q  Local Address           Foreign Address
(state)
-----
75059a0  TCP      0      0  172.30.80.231.1538     172.30.0.39.58497
TIME_WAIT
7506420  TCP      0      0  172.30.80.231.443     10.1.20.14.51006
TIME_WAIT
75044a0  TCP      0      0  172.30.80.231.443     10.1.20.14.51000
TIME_WAIT
7504f20  TCP      0      0  172.30.80.231.443     10.1.20.14.50997
TIME_WAIT
7503f60  TCP      0      0  127.0.0.1.3000        127.0.0.1.1064
ESTABLISHED
7503a20  TCP      0      0  127.0.0.1.3000        127.0.0.1.1063
ESTABLISHED
75034e0  TCP      0      0  127.0.0.1.3000        127.0.0.1.1062
ESTABLISHED
7502fa0  TCP      0      0  127.0.0.1.3000        127.0.0.1.1061
ESTABLISHED
7502a60  TCP      0      0  127.0.0.1.3000        127.0.0.1.1060
ESTABLISHED
7502520  TCP      0      0  127.0.0.1.1063        127.0.0.1.3000
ESTABLISHED
7501fe0  TCP      0      0  127.0.0.1.1062        127.0.0.1.3000
ESTABLISHED
7501aa0  TCP      0      0  127.0.0.1.1061        127.0.0.1.3000
ESTABLISHED

```

The following table describes each column in the above output.


Column Heading	Description
PCB	Printed circuit board in the server that is active on the connection.
Proto	Protocol used on this connection. Valid values are: TCP - Transport Control Protocol UDP - User Datagram Protocol
Recv-Q	Receiving queue - pertains to the queue on the server that receives packets from the Internet. This column should always display a zero (0). Packets should not be piling up in this queue.
Send-Q	Sending queue - pertains to the queue on the server that sends out packets to the Internet. This column should always display a zero (0). Packets should not be piling up in this queue.
Local Address	Local server's IP address and port number, or IP address and the name of a service.

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Column Heading	Description
Foreign Address	Hostname and service, or IP address and port number to which you are connected. The asterisk is a placeholder for IP addresses, which of course cannot be known until a remote host connects.
(state)	Current state of the TCP or UDP connection. TCP states can be: LISTEN waiting to receive a connection ESTABLISHED a connection is active TIME_WAIT a recently terminated connection; this should last only a minute or two, then change back to LISTEN. The socket pair cannot be re-used as long the TIME_WAIT state persists. UDP is stateless, so the "State" column is always blank.

## Show users Command

The **show users** command displays information about users currently logged into a session on the server. Each user is indicated by an index number. The following is an example of the show users command output.

 **Note:** The index number for Web sessions always begins at 31.

```
ACMEPACKET# show users
Index task-id      remote-address      IdNum duration type      state      User
-----
0 0x33a4c394          0 00:01:20 console user * console
1 0x33a68858      172.30.0.39:39385   1 00:00:08 telnet  user  user
31 NA                10.1.20.14:51218   31 00:00:55 http   user  user
32 NA                10.1.20.14:443    32 00:00:29 https  user  user
```

The following table describes each column in the above output.

Column Heading	Description
Index	Number that the server assigns to the user as an identification of that user. Note: The index for Web sessions always begins at index 31.
Task-id	Alpha-numeric number that the server assigns to the task currently being performed. This is the session ID assigned to the task at log-in time. This field is not applicable to the Web server.
Remote-address	IP address and port number for which the server is connected.
IdNum	Identification number of the user currently logged into the server. This number is the same as the Index number.
Duration	Amount of time, in hours, minutes, and seconds, that the user has currently been logged into a session on the server. Format is HH:MM:SS.
Type	Type of service that the user is currently using for connection to the server. Valid values can be: Console User is connected to the server via a local console. Telnet User is connected to the server via a Telnet session. FTP User is connected to the server using FTP. HTTP User is connected to the server using a Web HTTP service.

Column Heading	Description
	HTTPS User is connected to the server using a secure Web HTTPS service.
State	Current state of the connection on the server. Valid values are: admin user Note: An "*" indicates a current connection.
User	Current user type logged into the server. Valid values are: console user admin

### Kill index Command

The **kill <index>** command terminates a session on the server. The following example uses the "show users" command to display the index number to use with the kill <index> command.

```
ORACLE# show users
Index task-id      remote-address      IdNum duration type      state      User
-----
0 0x33a4c394
1 0x33a68858      172.30.0.39:39385  1 00:00:08 telnet  user      user
31 NA              10.1.20.14:51218  31 00:00:55 http   user      user
32 NA              10.1.20.14:443    32 00:00:29 https  user      user
ORACLE# kill 31
```

The above kill 31 command terminates the Web server session number 31.

After setting the Web server configuration, you can view the stored monitored data from your Oracle Communications Session Border Controller(s) using the Web-based GUI via your Internet browser. For more information about the Web-based GUI, see [Introduction](#).



## RTC Support

This appendix summarizes real-time configuration (RTC) support status for the Oracle Communications Session Border Controller. The table below lists which configuration elements are supported by RTC and which are not.

ACLI Configuration Elements	Parameter Details
Access Control	
Accounting Config	
Authentication	
Certificate Record	
Class Profile	
Codec Policy	
DNS ALG Service	
DNS Config	
Enum	
External Policy Server	
H.323	<p>The following H.323 stack subelement parameters are not RTC supported in that, if you save and activate a configuration, calls already in progress will be dropped. A reboot is required.</p> <ul style="list-style-type: none"> <li>• state</li> <li>• isgateway</li> <li>• realm-id</li> <li>• assoc-stack</li> <li>• options</li> <li>• proxy-mode</li> <li>• local-ip</li> <li>• max-calls</li> <li>• max-channels</li> <li>• registration-ttl</li> <li>• terminal-alias</li> </ul>

## RTC Support

ACLI Configuration Elements	Parameter Details
	<ul style="list-style-type: none"> <li>• prefixes</li> <li>• ras-port</li> <li>• q931-port</li> <li>• auto-gk-discovery</li> <li>• multicast</li> <li>• gatekeeper</li> <li>• h245-tunneling</li> <li>• gk-identifier</li> <li>• alternate-transport</li> <li>• q931-max-calls</li> <li>• filename</li> </ul>
Host Route	
IPSEC	
IWF	
Licensing	
Local Policy	
Local Response Map	
Local Routing Config	
Media Manager	<p>The Media Manager element is supported with the exception of the following parameters:</p> <ul style="list-style-type: none"> <li>• red-flow-port</li> <li>• red-max-trans</li> <li>• red-sync-start-time</li> <li>• red-sync-comp-time</li> </ul>
Media Policy	
Media Profile	
Network Interface	
Net Management Control	
Network Parameters	<p>The Network Parameters element is supported with the exception of the following parameters:</p> <ul style="list-style-type: none"> <li>• SCTP parameters</li> </ul>
NTP Sync	
Packet Trace Config	
Q850 SIP Map	
Realm Config	
Redundancy Config	<p>The Redundancy Config element is supported with the exception of the following parameters:</p>

ACLI Configuration Elements	Parameter Details
	<ul style="list-style-type: none"> <li>state</li> <li>port</li> <li>cfg-port</li> <li>cfg-max-trans</li> <li>cfg-sync-start-time</li> <li>cfg-sync-comp-time</li> </ul>
Session Agent	
Session Group	
Session Router	
Session Constraints	
Session Translation	
SIP Config	<p>The SIP Config element is supported with the exception of the following parameters:</p> <ul style="list-style-type: none"> <li>red-sip-port</li> <li>red-max-trans</li> <li>red-sync-start-time</li> <li>red-sync-stop-time</li> </ul>
SIP Feature	
SIP Interface	collect>boot-state
SIP Manipulation	
SIP NAT	<p>The SIP NAT element is supported with the exception of the following parameters:</p> <ul style="list-style-type: none"> <li>ext-address</li> </ul>
SIP Response Map	
SNMP	
Static Flow	
Steering Pool	
Surrogate Agent	
System	<p>The System Config element is supported with the exception of the following parameters:</p> <ul style="list-style-type: none"> <li>options</li> </ul>
Test Pattern Rule	
Test Policy	
Test Translation	
TLS Global	<p>When configured to support MSRP over TLS, this configuration element is not RTC enabled. If a single TLS profile is used by both SIP and MSRP, RTC changes will be applied for the SIP TLS configuration ONLY.</p>

## RTC Support

---

ACLI Configuration Elements	Parameter Details
TLS Profile	
Translation Rules	
Trap Receiver	



---

## Format of Exported Text Files

### Introduction

---

This Appendix provides a sample and format of each type of exported file from the Web-based GUI. Sample information in these files are provided as a reference for your convenience.

Exported file examples include:

- Session Summary exported file (text format)
- Session Details exported file (text format)
- Ladder Diagram exported file (HTML format)



**Note:** Oracle recommends you open an exported text file using an application that provides advanced text formatting to make it easier to read.

### Exporting Files

---

The Web-based GUI allows you to export Monitor and Trace information to a text file from the Sessions, Registrations, Subscriptions and Notable Events Reports, as well as from a specific ladder diagram, or from a page containing the results of a search. The data exports to a file that you can open and view as required.

You can export any of the following to a file:

From the Sessions, Registrations, Subscriptions, and Notable Events Reports:

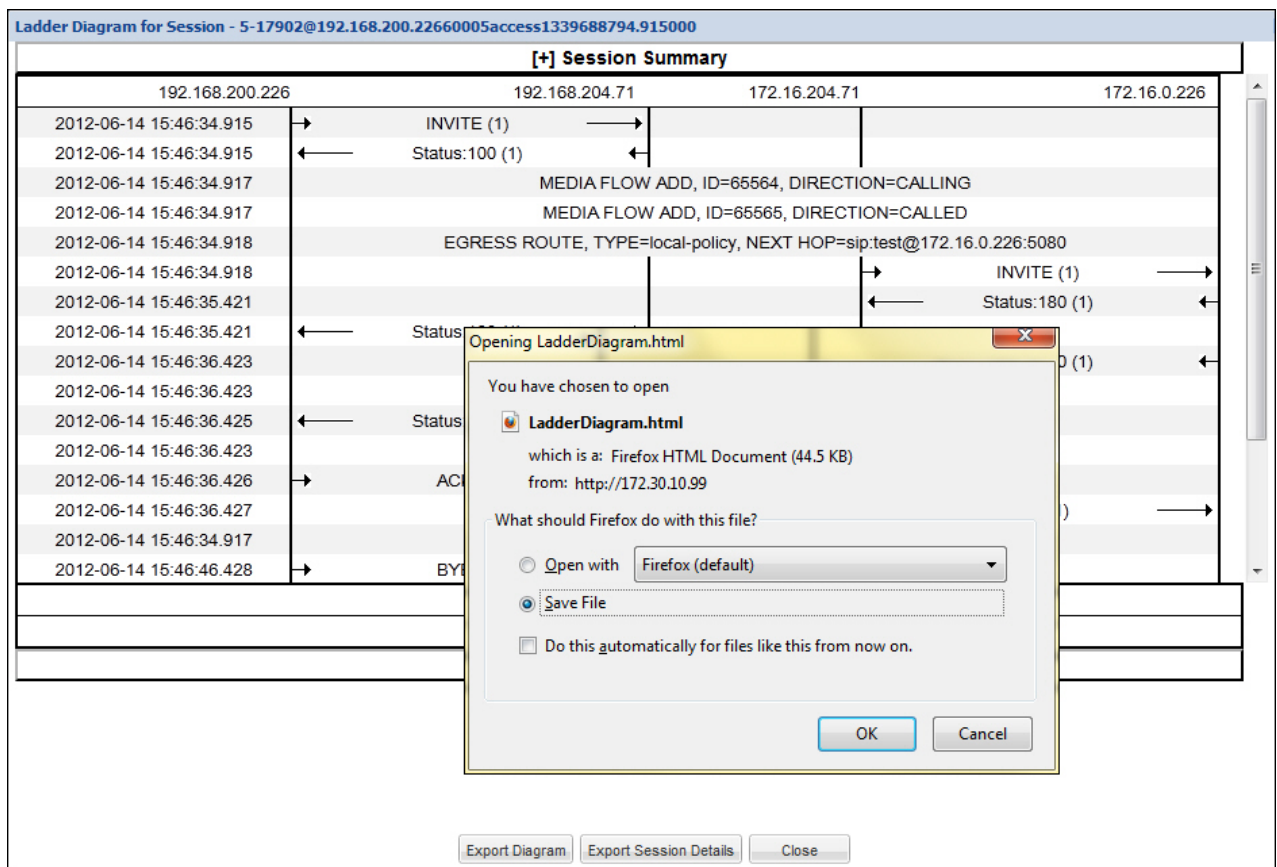
- **Export session details** - Exports the SIP messages and media events associated with the selected session, to a text file.
- **Export summary** - Exports all logged session summary records, to a text file. (Exports ALL call session summary records or the records that matched a search criteria).

From the Ladder Diagram:

- **Export diagram** - Exports all of the information in the Ladder Diagram to an HTML file (Session Summary, SIP Message Details, and QoS statistics).
- **Export session details** - Exports detailed information about the SIP messages and media events in the Ladder Diagram associated with the selected session, to a text file.

The following example shows the export of a Ladder Diagram to a file called LadderDiagram.html.

## Format of Exported Text Files



## Session Summary Exported Text File

The following is an example of a Session Summary exported text file from the Web-based GUI.

### Example

```
-----Session Summary-----
Startup Time: 2011-09-20 12:58:44.375
State: TERMINATED-200
Duration: 5
From URI: sipp <&lt; sip:sipp@172.16.34.16:5060&&tag=1
To URI: sut <&lt; sip:service@172.16.34.225:5060&&tag=13451
Ingress Src Address: 172.16.34.16
Ingress Src Port: 5060
Ingress Dest Address: 172.16.34.225
Ingress Dest Port: 5060
Egress Source Address: 192.168.34.225
Egress Source Port: 5060
Egress Destination Address: 192.168.34.17
Egress Destination Port: 5060
Ingress Realm: access
Egress Realm: backbone
Ingress NetworkIf: access
Egress NetworkIf: backbone
-----Session Summary-----
Startup Time: 2011-09-20 12:58:05.340
State: TERMINATED-200
```

```

Duration: 5
From URI: sipp < sip:sipp@172.16.34.16:5060 >;tag=1
To URI: sut < sip:service@172.16.34.225:5060 >;tag=13450
Ingress Src Address: 172.16.34.16
Igress Src Port: 5060
Igress Dest Address: 172.16.34.225
Igress Dest Port: 5060
Egress Source Address: 192.168.34.225
Egress Source Port: 5060
Egress Destination Address: 192.168.34.17
Egress Destination Port: 5060
Igress Realm: access
Egress Realm: backbone
Igress NetworkIf: access
Egress NetworkIf: backbone

```

## Session Details Exported Text File

The following is an example of the a Session Details exported text file from the Web-based GUI.

### Example

```

Session Details:
-----
Nov 3 08:50:56.852 On [2:0]172.16.34.225:5060 received from 172.16.34.16:5060

INVITE sip:service@172.16.34.225:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.34.16:5060;branch=z9hG4bK-1-0
From: sipp < sip:sipp@172.16.34.16:5060 >;tag=1
To: sut < sip:service@172.16.34.225:5060 >
Call-ID: 1-668@172.16.34.16
CSeq: 1 INVITE
Contact: sip:sipp@172.16.34.16:5060
Max-Forwards: 70
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 135

v=0
o=user1 53655765 2353687637 IN IP4 172.16.34.16
s=-
c=IN IP4 172.16.34.16
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000

-----
Nov 3 08:50:56.855 On [2:0]172.16.34.225:5060 sent to 172.16.34.16:5060

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.16.34.16:5060;branch=z9hG4bK-1-0
From: sipp < sip:sipp@172.16.34.16:5060 >;tag=1
To: sut < sip:service@172.16.34.225:5060 >
Call-ID: 1-668@172.16.34.16
CSeq: 1 INVITE

----MBCD Evt
Nov 3 08:50:56.862 On 127.0.0.1:2945 sent to 127.0.0.1:2944
mbcdEvent=FLOW ADD
FlowCollapsed=enabled

```

## Format of Exported Text Files

---

```
FlowDirection=CALLING
FlowID=65541
MediaFormat=0
MediaReleased=disabled
MediaType=audio/PCMU/
OtherFlowID=0
TOSBits=0
InputSourcev4Addr=
InputSourcePort=0
InputDestv4Addr=172.16.34.225
InputDestPort=10004
OutputSourcev4Addr=192.168.34.225
OutputDestv4Addr=
OutputDestPort=0
InputRealm=access
OutputRealm=backbone
----MBCD Evt
Nov 3 08:50:56.862 On 127.0.0.1:2945 received from 127.0.0.1:2944

mbcdEvent=FLOW ADD
FlowCollapsed=enabled
FlowDirection=CALLED
FlowID=65542
MediaFormat=0
MediaReleased=disabled
MediaType=audio/PCMU/
OtherFlowID=0
TOSBits=0
InputSourcev4Addr=
InputSourcePort=0
InputDestv4Addr=192.168.34.225
InputDestPort=20004
OutputSourcev4Addr=172.16.34.225
OutputDestv4Addr=172.16.34.16
OutputDestPort=6000
InputRealm=backbone
OutputRealm=access

-----
Nov 3 08:50:56.865 On [1:0]192.168.34.225:5060 sent to 192.168.34.17:5060

INVITE sip:service@192.168.34.17:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.34.225:5060;branch=z9hG4bK4od0io20183g8ssv32f1.1
From: sipp <sip:sipp@172.16.34.16:5060>;tag=1
To: sut <sip:service@172.16.34.225:5060>
Call-ID: 1-668@172.16.34.16
CSeq: 1 INVITE
Contact: <sip:sipp@192.168.34.225:5060;transport=udp>
Max-Forwards: 69
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 140

v=0
o=user1 53655765 2353687637 IN IP4 192.168.34.225
s=-
c=IN IP4 192.168.34.225
t=0 0
m=audio 20004 RTP/AVP 0
a=rtpmap:0 PCMU/8000

-----
Nov 3 08:50:56.868 On [1:0]192.168.34.225:5060 received from
192.168.34.17:5060
```

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 192.168.34.225:5060;branch=z9hG4bK4od0io20183g8ssv32f1.1
From: sipp <sip:sipp@172.16.34.16:5060>;tag=1
To: sut <sip:service@172.16.34.225:5060>;tag=2578
Call-ID: 1-668@172.16.34.16
CSeq: 1 INVITE
Contact: <sip:192.168.34.17:5060;transport=UDP>
Content-Length: 0
```

-----  
Nov 3 08:50:56.872 On [2:0]172.16.34.225:5060 sent to 172.16.34.16:5060

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 172.16.34.16:5060;branch=z9hG4bK-1-0
From: sipp <sip:sipp@172.16.34.16:5060>;tag=1
To: sut <sip:service@172.16.34.225:5060>;tag=2578
Call-ID: 1-668@172.16.34.16
CSeq: 1 INVITE
Contact: <sip:service@172.16.34.225:5060;transport=udp>
Content-Length: 0
```

-----  
Nov 3 08:50:56.872 On [1:0]192.168.34.225:5060 received from 192.168.34.17:5060

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.34.225:5060;branch=z9hG4bK4od0io20183g8ssv32f1.1
From: sipp <sip:sipp@172.16.34.16:5060>;tag=1
To: sut <sip:service@172.16.34.225:5060>;tag=2578
Call-ID: 1-668@172.16.34.16
CSeq: 1 INVITE
Contact: <sip:192.168.34.17:5060;transport=UDP>
Content-Type: application/sdp
Content-Length: 137
```

```
v=0
o=user1 53655765 2353687637 IN IP4 192.168.34.17
s=-
c=IN IP4 192.168.34.17
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

----MBCD Evt

Nov 3 08:50:56.878 On 127.0.0.1:2945 sent to 127.0.0.1:2944

```
mbcdEvent=FLOW MODIFY
FlowCollapsed=enabled
FlowDirection=CALLING
FlowID=65541
MediaFormat=0
MediaReleased=disabled
MediaType=audio/PCMU/
OtherFlowID=0
TOSBits=0
InputSourcev4Addr=
InputSourcePort=0
InputDestv4Addr=172.16.34.225
InputDestPort=10004
OutputSourcev4Addr=192.168.34.225
OutputDestv4Addr=192.168.34.17
OutputDestPort=6000
InputRealm=access
OutputRealm=backbone
```

## Format of Exported Text Files

---

-----  
Nov 3 08:50:56.881 On [2:0]172.16.34.225:5060 sent to 172.16.34.16:5060

SIP/2.0 200 OK  
Via: SIP/2.0/UDP 172.16.34.16:5060;branch=z9hG4bK-1-0  
From: sipp <sip:sipp@172.16.34.16:5060>;tag=1  
To: sut <sip:service@172.16.34.225:5060>;tag=2578  
Call-ID: 1-668@172.16.34.16  
CSeq: 1 INVITE  
Contact: <sip:service@172.16.34.225:5060;transport=udp>  
Content-Type: application/sdp  
Content-Length: 138  
v=0  
o=user1 53655765 2353687637 IN IP4 172.16.34.225  
s=-  
c=IN IP4 172.16.34.225  
t=0 0  
m=audio 10004 RTP/AVP 0  
a=rtpmap:0 PCMU/8000

-----  
Nov 3 08:50:56.883 On [2:0]172.16.34.225:5060 received from 172.16.34.16:5060

ACK sip:service@172.16.34.225:5060 SIP/2.0  
Via: SIP/2.0/UDP 172.16.34.16:5060;branch=z9hG4bK-1-5  
From: sipp <sip:sipp@172.16.34.16:5060>;tag=1  
To: sut <sip:service@172.16.34.225:5060>;tag=2578  
Call-ID: 1-668@172.16.34.16  
CSeq: 1 ACK  
Contact: sip:sipp@172.16.34.16:5060  
Max-Forwards: 70  
Subject: Performance Test  
Content-Length: 0

-----  
Nov 3 08:50:56.887 On [1:0]192.168.34.225:5060 sent to 192.168.34.17:5060

ACK sip:192.168.34.17:5060;transport=UDP SIP/2.0  
Via: SIP/2.0/UDP 192.168.34.225:5060;branch=z9hG4bK48k3k1301ot00ssvf1v0.1  
From: sipp <sip:sipp@172.16.34.16:5060>;tag=1  
To: sut <sip:service@172.16.34.225:5060>;tag=2578  
Call-ID: 1-668@172.16.34.16  
CSeq: 1 ACK  
Contact: <sip:sipp@192.168.34.225:5060;transport=udp>  
Max-Forwards: 69  
Subject: Performance Test  
Content-Length: 0

-----  
Nov 3 08:51:01.883 On [2:0]172.16.34.225:5060 received from 172.16.34.16:5060

BYE sip:service@172.16.34.225:5060 SIP/2.0  
Via: SIP/2.0/UDP 172.16.34.16:5060;branch=z9hG4bK-1-7  
From: sipp <sip:sipp@172.16.34.16:5060>;tag=1  
To: sut <sip:service@172.16.34.225:5060>;tag=2578  
Call-ID: 1-668@172.16.34.16  
CSeq: 2 BYE  
Contact: sip:sipp@172.16.34.16:5060  
Max-Forwards: 70  
Subject: Performance Test  
Content-Length: 0

-----  
Nov 3 08:51:01.887 On [1:0]192.168.34.225:5060 sent to 192.168.34.17:5060

```

BYE sip:192.168.34.17:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 192.168.34.225:5060;branch=z9hG4bK5oq46d301gv0dus227f1.1
From: sipp <sip:sipp@172.16.34.16:5060>;tag=1
To: sut <sip:service@172.16.34.225:5060>;tag=2578
Call-ID: 1-668@172.16.34.16
CSeq: 2 BYE
Contact: <sip:sipp@192.168.34.225:5060;transport=udp>
Max-Forwards: 69
Subject: Performance Test
Content-Length: 0

```

```

-----
Nov 3 08:51:01.889 On [1:0]192.168.34.225:5060 received from
192.168.34.17:5060

```

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.34.225:5060;branch=z9hG4bK5oq46d301gv0dus227f1.1
From: sipp <sip:sipp@172.16.34.16:5060>;tag=1
To: sut <sip:service@172.16.34.225:5060>;tag=2578
Call-ID: 1-668@172.16.34.16
CSeq: 2 BYE
Contact: <sip:192.168.34.17:5060;transport=UDP>
Content-Length: 0

```

```

-----
Nov 3 08:51:01.892 On [2:0]172.16.34.225:5060 sent to 172.16.34.16:5060

```

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.16.34.16:5060;branch=z9hG4bK-1-7
From: sipp <sip:sipp@172.16.34.16:5060>;tag=1
To: sut <sip:service@172.16.34.225:5060>;tag=2578
Call-ID: 1-668@172.16.34.16
CSeq: 2 BYE
Contact: <sip:service@172.16.34.225:5060;transport=udp>
Content-Length: 0

```

```

----MBCD Evt

```

```

Nov 3 08:51:01.891 On 127.0.0.1:2945 sent to 127.0.0.1:2944

```

```

mbcdEvent=FLOW DELETE
FlowCollapsed=enabled
FlowDirection=CALLING
FlowID=65541
MediaFormat=0
MediaReleased=disabled
MediaType=audio/PCMU/
OtherFlowID=0
TOSBits=0
InputSourcev4Addr=
InputSourcePort=0
InputDestv4Addr=172.16.34.225
InputDestPort=10004
OutputSourcev4Addr=192.168.34.225
OutputDestv4Addr=192.168.34.17
OutputDestPort=6000
InputRealm=access
OutputRealm=backbone

```

```

----MBCD Evt

```

```

mbcdEvent=FLOW DELETE
FlowCollapsed=enabled
FlowDirection=CALLED
FlowID=65542
MediaFormat=0
MediaReleased=disabled

```

## Format of Exported Text Files

```
MediaType=audio/PCMU/
OtherFlowID=65541
TOSBits=0
InputSourcev4Addr=
InputSourcePort=0
InputDestv4Addr=192.168.34.225
InputDestPort=20004
OutputSourcev4Addr=172.16.34.225
OutputDestv4Addr=172.16.34.16
OutputDestPort=6000
InputRealm=backbone
OutputRealm=access

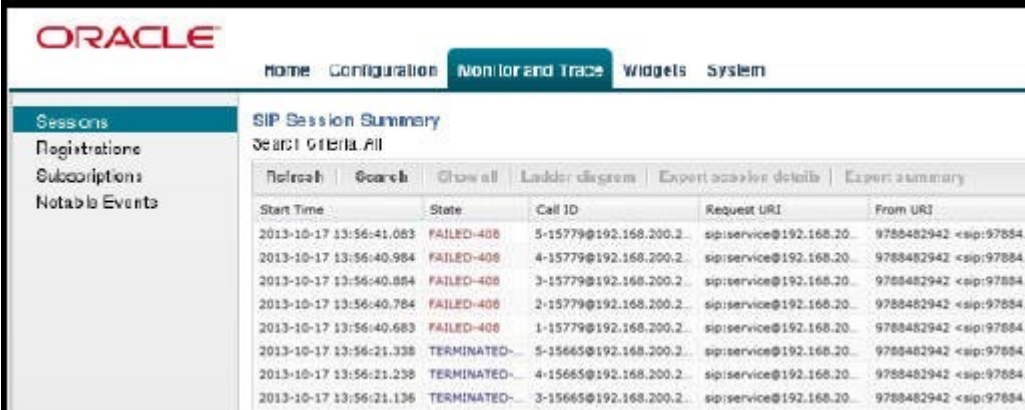
-----Session Summary-----
Startup Time: 2012-01-25 10:28:30.394

State: TERMINATED-200
Duration: 5
From URI: sipp <&lt; sip:sipp@172.16.34.16:5060&&gt;>;tag=1
To URI: sut <&lt; sip:service@172.16.34.225:5060&&gt;>;tag=2578
Ingress Src Address: 172.16.34.16
Igress Src Port: 5060
Igress Dest Address: 172.16.34.225
Igress Dest Port: 5060
Egress Source Address: 192.168.34.225
Egress Source Port: 5060
Egress Destination Address: 192.168.34.17
Egress Destination Port: 5060
Igress Realm: access
Egress Realm: backbone
Igress NetworkIf: access
Egress NetworkIf: backbone
```

## Ladder Diagram Exported HTML File

The following is an example of a Ladder Diagram for a session, exported to an HTML file from the Web-based GUI.

### Example



The screenshot shows the Oracle Communications Session Border Controller GUI. The top navigation bar includes 'Home', 'Configuration', 'Monitor and Trace', 'Widgets', and 'System'. The 'Monitor and Trace' tab is active. On the left, a sidebar menu lists 'Sessions', 'Registrations', 'Subscriptions', and 'Notable Events'. The main content area displays a 'SIP Session Summary' table with columns for Start Time, State, Call ID, Request URI, and From URI. The table contains several rows of session data, including failed and terminated sessions.

Start Time	State	Call ID	Request URI	From URI
2013-10-17 13:56:41.083	FAILED-408	5-15779@192.168.200.2	sip:service@192.168.200.2	9788482942 <sip:97884
2013-10-17 13:56:40.984	FAILED-408	4-15779@192.168.200.2	sip:service@192.168.200.2	9788482942 <sip:97884
2013-10-17 13:56:40.884	FAILED-408	3-15779@192.168.200.2	sip:service@192.168.200.2	9788482942 <sip:97884
2013-10-17 13:56:40.784	FAILED-408	2-15779@192.168.200.2	sip:service@192.168.200.2	9788482942 <sip:97884
2013-10-17 13:56:40.683	FAILED-408	1-15779@192.168.200.2	sip:service@192.168.200.2	9788482942 <sip:97884
2013-10-17 13:56:21.338	TERMINATED-	5-15665@192.168.200.2	sip:service@192.168.200.2	9788482942 <sip:97884
2013-10-17 13:56:21.238	TERMINATED-	4-15665@192.168.200.2	sip:service@192.168.200.2	9788482942 <sip:97884
2013-10-17 13:56:21.138	TERMINATED-	3-15665@192.168.200.2	sip:service@192.168.200.2	9788482942 <sip:97884



[-] Session Summary																																																																																																	
State	TERMINATED-200			Duration	10																																																																																												
From URI	"+2273636" <tel:781-414-2345>;tag=60005			To URI	sut <sip:kam@192.168.204.71:5060>;tag=50004																																																																																												
Ingress Src IP:Port	192.168.200.226:5070			Egress Src IP:Port	172.16.204.71:5060																																																																																												
Ingress Dest IP:Port	192.168.204.71:5060			Egress Dest IP:Port	172.16.0.226:5070																																																																																												
Ingress Realm	access			Egress Realm	core																																																																																												
Ingress Network Intf	M00			Egress Network Intf	M10																																																																																												
Ingress Transport	UDP			Egress Transport	UDP																																																																																												
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