SIP Trunk Monitoring

Overview

This feature groups the monitoring of SIP dial-peer, endpoints and servers by consolidating dial-peers with the same SIP Out-of-Dialog OPTIONS ping setup.

The SIP Out-Of-Dialog OPTIONS Ping Group feature is an existing mechanism that is used by Cisco Unified Border Element (CUBE) to monitor the status of a single SIP dial-peer destination (keepalive). A generic heartbeat mechanism allows you to monitor the status of SIP servers or endpoints and provide the option of marking a dial peer as inactive (busyout) upon total heartbeat failure.

You can also consolidate the sending of OPTIONS ping packets by grouping dial peers with the same destination. You must create a profile to send one set of OPTIONS ping for a group of dial-peers. If that ping fails, then all of the associated dial-peers are busied out (inactive) by CUBE.

Note Configuring the same Options profile on two or more dial-peers with different bind interfaces configured is not supported. This leads to a scenario wherein the OPTIONS SIP message is not sent from all bind interfaces except the first configured one. But the dial-peer is always marked as ACTIVE. Similarly, it is also not supported in multi VRF setup.

You can use the **shutdown** command to suspend monitoring of all dial peers associated with a keepalive profile.

The command **voice-class sip options-keepalive profile tag** is used to monitor a group of SIP servers or endpoints and the existing **voice-class sip options-keepalive** command is used to monitor a single SIP endpoint or server.

You can configure a server group to be a part of a OPTIONS ping group. A SIP dial peer is updated to BUSY state only if all targets of its server group does not response to the OPTIONS ping. Members of a server group are tested in turn, not in parallel. That is, if the first server group member becomes unavailable, then the second member is tested, and so on. Only when all of the group members are exhausted, is the dial-peer busied out.

Technical Assistance

Description	Link
The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies. To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.	http://www.cisco.com/support

Feature Information

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 1: Feature Information for SIP Out-of-dialog OPTIONS Ping Group

Feature Name	Releases	Feature Information
SIP Out-of-dialog OPTIONS Ping Group	Baseline Functionality	This feature groups the monitoring of SIP dial peers endpoints and servers by consolidating SIP Out-Of-Dialog Options of dial peers with the similar OPTIONS ping setup.

OPTIONS Ping for DNS SRV Hosts

From Cisco IOS XE Cupertino 17.9.1a, you can monitor all the SRV hosts that are part of the DNS destination using the OPTIONS Keepalive mechanism. It is therefore possible to load balance calls across all active destinations.

This feature may be used by configuring a dial-peer target with a fully qualified domain name (FQDN) that resolves to a set of DNS SRV records.

A Domain Name System Service Record (DNS SRV) record comprises of multiple resources, each with its own weight, priority and host name. CUBE uses DNS again to resolve the IP address for each of these hostnames.CUBE then triggers an Out-of-Dialog OPTIONS ping to each of these addresses to monitor the status of the hosts.

Once the DNS A query is successful, CUBE triggers an Out-of-Dialog OPTIONS Ping. The Out-of-Dialog OPTIONS Ping mechanism is used by CUBE to monitor the status of the single SIP dial-peer destination (keepalive).

If the DNS lookup returns multiple addresses, then Out-of-Dialog keepalive sessions are established with each of the hosts. For the same destination hostname and the same **voice-class sip options-keepalive profile** *tag* , only a single Out-of-Dialog OPTIONS SRV entry will be added in the keepalive session table. This SRV entry in the keepalive session table can maintain the session details for each of the hosts.

CUBE compares the least value of Time to Live (TTL) recorded for both SRV resolution and Type A/AAAA resolution. CUBE maintains the least value that is obtained, against the DNS SRV entry in the Out-of-Dialog session table. CUBE starts a periodic timer based on the least TTL value. When the TTL timer expires, the Out-of-Dialog session status is removed for all the existing host entries. Thereafter, CUBE performs a new SRV or Type-A or AAAA lookup to update the DNS SRV entry list.

A generic heartbeat mechanism allows you to monitor the status of SIP servers or endpoints. It provides the option of marking a dial-peer as **active**, inactive (**busyout**) for a total heartbeat failure, and partially active (**partial**). A dial-peer is marked as partially active if at least one of the destinations is active out of a group, and the rest are inactive (busyout). A dial-peer is marked as busyout, only if all the destinations in the dial-peer have a heartbeat failure and fail to respond.

Once OPTIONS Ping is successful, this destination is considered for the routing of call handling. CUBE monitors all the destinations irrespective of the response (503, 200 OK, and so on) that it receives. Based on the response, CUBE identifies the destination that it can communicate with. If CUBE receives a 503 response or no response for the INVITE, CUBE then marks that destination as busyout and attempts call on the next destination that is marked as active. The call is rejected if all possible destinations are busied out.

Note

- If Outbound Proxy is configured on dial-peer or a tenant that is associated with the dial-peer, CUBE maintains keepalive session with the Outbound Proxy address.
- You need to configure the same transport type for the dial-peers with same SRV destination.

You must configure **voice-class sip options-keepalive profile <tag>** under the specific **dial-peer** to support the DNS SRV lookup using the OPTIONS keepalive mechanism. If you configure the **voice-class sip options-keepalive** command under the dial-peer, Load Balancing using DNS SRV is not supported.

Note We recommended that you configure the same **voice-class sip options-keepalive profile <tag>** under all dial-peers that have the same DNS session target. This helps to reduce the OPTIONS Ping traffic.

To display the status of the destination when options-keepalive is configured under dial-peer, use the CLI command **show dial-peer voip keepalive status <dp-tag> | tenant <tenant-id> | <cr>**. The options keepalive status is maintained by CUBE for individual session targets and server groups in this command. The keepalive status is displayed for IPv4, IPv6, and DNS format destinations.

The CLI command **show dial-peer voice summary** is enhanced to display the overall keepalive status for the DNS SRV at the dial-peer level.

To display the status of session target DNS with the list of servers resolved against the DNS SRV records, use the CLI command **show voice class sip-options-keepalive <profile-tag>**.

Load Balancing for DNS SRV Hosts

The usage of DNS SRV as the target for CUBE helps in load balancing of the outbound SIP call traffic across the trunk. Based on the priority, weight, and status of the DNS SRV records, the multiple hosts associated with DNS SRV are used. CUBE distributes calls across the SRVs based on the priority and status of the DNS SRV records.

During call routing, CUBE reads through the DNS SRV records that it has collected. Based on the information, CUBE identifies the trunk dial-peer destinations that are still available. Based on this, CUBE can distribute the traffic of outbound SIP calls in a more efficient way.

If configured, CUBE uses the Out-of-Dialog OPTIONS Ping mechanism to monitor the status of the hosts defined by the dial-peer destination SRV record. For more information on OPTIONS Ping for DNS SRV hosts, see OPTIONS Ping for DNS SRV Hosts, on page 2.

Configure SIP Out-of-Dialog OPTIONS Ping Group

Before you begin

Configure SIP profiles and server groups.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class sip-options-keepalive** *keepalive-group-profile-id*
4. **description** *text*
5. **transport {tcp [tls] | udp | system}**
6. **sip-profiles** *profile-number*
7. **down-interval** *down-interval*
8. **up-interval** *up-interval*
9. **retry** *retry-interval*
10. **exit**
11. **dial-peer voice** *dial-peer-id* **voip**
12. **session protocol sipv2**
13. **voice-class sip options-keepalive profile** *keepalive-group-profile-id*
14. **session server-group** *server-group-id*
15. **end**
16. **show voice class sip-options-keepalive** *keepalive-group-profile-id*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enters privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**voice class sip-options-keepalive** *keepalive-group-profile-id* **Example:** `Device(config)# voice class sip-options-keepalive 171`	Configures a keepalive profile and enters voice class configuration mode. • You can use the **shutdown** command to suspend keepalive activity for all dial peers associated with the keepalive profile.
Step 4	**description** *text* **Example:** `Device(config-class)# description Target Boston`	Configures a textual description for the keepalive heartbeat connection.
Step 5	**transport {tcp [tls] | udp | system}** **Example:** `Device(config-class)# transport tcp`	Defines the transport protocol that is used for the keepalive heartbeat connection. • The default value is system.
Step 6	**sip-profiles** *profile-number* **Example:** `Device(config-class)# sip-profiles 100`	Specifies the SIP profile that is to be used to send this message. • To configure a SIP profile, refer to “Configuring SIP Parameter Modification”.
Step 7	**down-interval** *down-interval* **Example:** `Device(config-class)# down-interval 35`	Configures the time (in seconds) at which an OPTIONS ping is sent to the dial-peer endpoint when the heartbeat connection to the endpoint is in Down status. • The default value is 30.
Step 8	**up-interval** *up-interval* **Example:** `Device(config-class)# up-interval 65`	Configures the time (in seconds) at which an OPTIONS ping is sent to the dial-peer endpoint when the heartbeat connection to the endpoint is in Up status. • The default value is 60.
Step 9	**retry** *retry-interval* **Example:** `Device(config-class)# retry 30`	Configures the maximum number of OPTIONS ping retries that are permitted for a dial-peer destination. After receiving failed responses for the configured number of OPTIONS pings, the heartbeat connection status should be switched from Up to Down. • The default value is 5. • If a successful response is received for an OPTIONS ping, the retry counter is set to zero.

	Command or Action	Purpose
Step 10	**exit** **Example:** `Device(config-class)# exit`	Exits voice class configuration mode and enters global configuration mode.
Step 11	**dial-peer voice** *dial-peer-id* **voip** **Example:** `Device(config)# dial-peer voice 123 voip`	Defines a local dial peer and enters dial peer configuration mode.
Step 12	**session protocol sipv2** **Example:** `Device(config-dial-peer)# session protocol sipv2`	Specifies SIP version 2 as the session protocol for calls between local and remote routers using the packet network.
Step 13	**voice-class sip options-keepalive profile** *keepalive-group-profile-id* **Example:** `Device(config-dial-peer)# voice-class sip options-keepalive profile 171`	Associates the dial peer with the specified keepalive group profile. The dial peer is monitored by CUBE according to the parameters defined by this profile.
Step 14	**session server-group** *server-group-id* **Example:** `Device(config-dial-peer)# session server-group 151`	Associates the dial peer with the specified keepalive group profile. The dial peer is monitored by the device according to the parameters defined by this profile.
Step 15	**end** **Example:** `Device(config-dial-peer)# end`	Exits dial peer configuration mode and enters privileged EXEC mode.
Step 16	**show voice class sip-options-keepalive** *keepalive-group-profile-id* **Example:** `Device# show voice class sip-options-keepalive 171`	Displays information about voice class server group.

Configuration Examples For SIP Out-of-Dialog OPTIONS Ping Group

Example: SIP Out-of-Dialog OPTIONS Ping for Group of SIP Endpoints

```
!Configuring the SIP profile
Device(config)# voice class sip-profiles 100
Device(config-class)# request OPTIONS sip-header SIP-Req-URI modify "; SIP/2.0" ";user=phone
 SIP/2.0"
```

```
!Configuring the SIP Keepalive Group
Device(config)# voice class sip-options-keepalive 171
Device(config-class)# transport tcp
Device(config-class)# sip-profile 100
Device(config-class)# down-interval 30
Device(config-class)# up-interval 60
Device(config-class)# retry 5
Device(config-class)# description Target New York
Device(config-class)# exit

!Configuring an outbound SIP Dial Peer
Device(config)# dial-peer voice 123 voip
Device(config-dial-peer)# session protocol sipv2
!Associating the Dial Peer with a keepalive profile group
Device(config-dial-peer)# session target dns:example.com
Device(config-dial-peer)# voice-class sip options-keepalive profile 171
Device(config-dial-peer)# end

!Verifying the Keepalive group configurations
Device# show voice class sip-options-keepalive 171

Voice class sip-options-keepalive: 171           AdminStat: Up
 Description: Target New York
 Transport: system              Sip Profiles: 100
 Interval(seconds) Up: 60               Down: 30
 Retry: 5

  Peer Tag      Server Group    OOD SessID      OOD Stat        IfIndex
  --------      ------------    ----------      --------        -------
  123                                                           100
```

Example: SIP Out-of-dialog OPTIONS Ping for Group of SIP Servers

```
!Configuring the Server Group
Device(config)# voice class server-group 151
Device(config-class)# ipv4 10.1.1.1 preference 1
Device(config-class)# ipv4 10.1.1.2 preference 2
Device(config-class)# ipv4 10.1.1.3 preference 3
Device(config-class)# hunt-scheme round-robin
Device(config-class)# description It has 3 entries
Device(config-class)# exit

!Configuring an E164 pattern map class
Device(config)# voice class e164-pattern-map 3000
Device(config-class)# e164 300

!Configuring an outbound SIP dial peer.
Device(config)# dial-peer voice 181 voip
!Associate a destination pattern map
Device(config-dial-peer)# destination e164-pattern-map 3000
Device(config-dial-peer)# session protocol sipv2
!Associate a server group with the dial peer
Device(config-dial-peer)# session server-group 151
!Associate the dial peer with a keepalive profile group
Device(config-dial-peer)# voice-class sip options-keepalive profile 171
Device(config-dial-peer)# end

!Verifying the Keepalive group configurations
Device# show voice class sip-options-keepalive 171
```

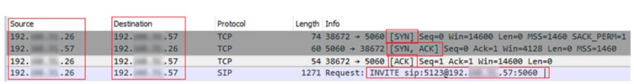
```
Voice class sip-options-keepalive: 171           AdminStat: Up
 Description: Target New York
 Transport: system              Sip Profiles: 100
 Interval(seconds) Up: 60               Down: 30
 Retry: 5

  Peer Tag      Server Group    OOD SessID      OOD Stat        IfIndex
  --------      ------------    ----------      --------        -------
  123                                                           100
  181           151                             Busy            106

  Server Group: 151             OOD Stat: Busy
   OOD SessID   OOD Stat
   ----------   --------
   1            Busy
   2            Busy
   3            Busy

 OOD SessID: 1                  OOD Stat: Busy
  Target: ipv4:10.1.1.1
  Transport: system             Sip Profiles: 100

 OOD SessID: 2                  OOD Stat: Busy
  Target: ipv4:10.1.1.2
  Transport: system             Sip Profiles: 100

 OOD SessID: 3                  OOD Stat: Busy
  Target: ipv4:10.5.0.1
  Transport: system             Sip Profiles: 100

------------------------------------------------------
```

Configure OPTIONS Ping Between CUCM and CUBE

This section describes how to enable Options Ping between Cisco Unified Communications Manager (CUCM) and Cisco Unified Border Element (CUBE).

The following figure describes how a CUCM extends a call out of a SIP Trunk:

The following figure shows the TCP three-way handshake in Wireshark:



Source	Destination	Protocol	Length	Info
192. .26	192. .57	TCP	74	38672 → 5060 [SYN] Seq=0 Win=14600 Len=0 MSS=1460 SACK_PERM=1
192. .57	192. .26	TCP	60	5060 → 38672 [SYN, ACK] Seq=0 Ack=1 Win=4128 Len=0 MSS=1460
192. .26	192. .57	TCP	54	38672 → 5060 [ACK] Seq=1 Ack=1 Win=14600 Len=0
192. .26	192. .57	SIP	1271	Request: INVITE sip:5123@192. .57:5060

Perform the following steps to configure OPTIONS ping between CUBE and CUCM:

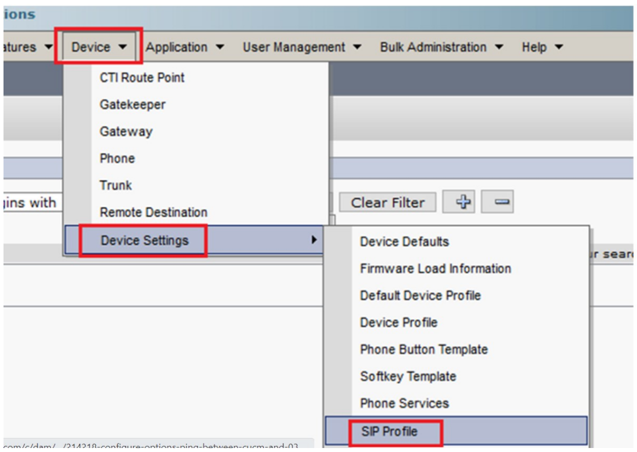
SUMMARY STEPS

1. Enable SIP **Options Ping** in the **SIP Profile Configuration:**
2. Add the SIP profile to the SIP trunk and click **Save**:
3. Enable SIP Options Ping on the far end of the SIP Trunk. In this case, 192.X.X.57 (ISR 4351).

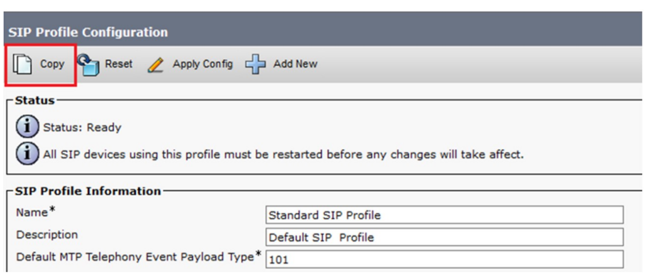
DETAILED STEPS

Step 1 Enable SIP **Options Ping** in the **SIP Profile Configuration:**

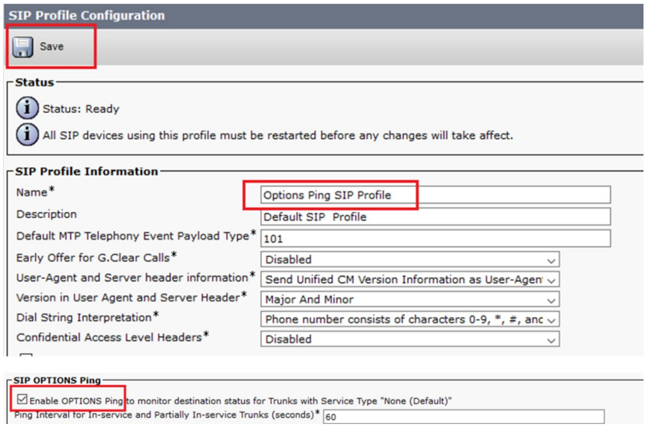
a) Navigate to **Cisco Unified CM Administration** >> **Device** >> **Device Settings** >> **SIP Profile** as shown in the following figure:



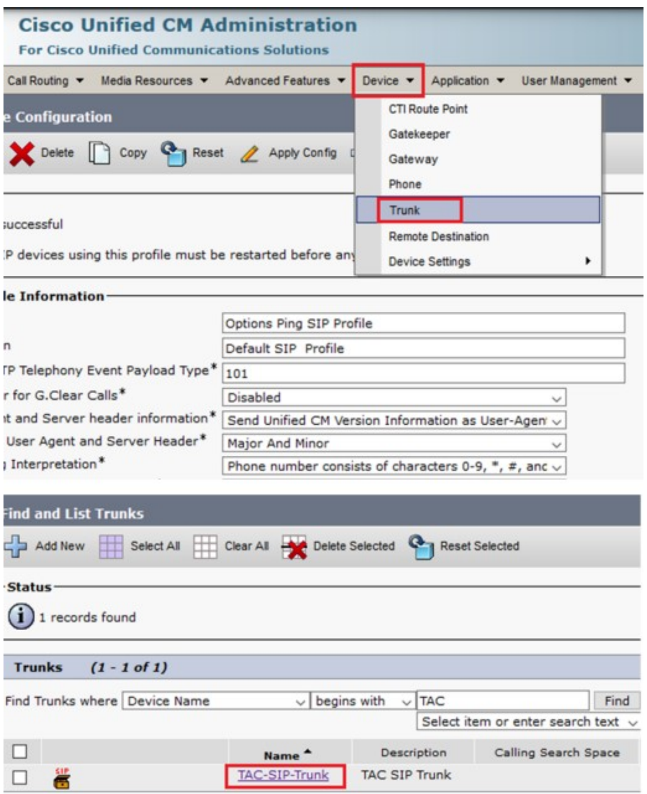
b) **Click find** and decide if you want to create a new **SIP Profile**, edit a **SIP Profile** that exists or make a copy of a SIP Profile. For this example, create a copy of the **Standard SIP Profile** as shown in the following figures:



c) Rename the new SIP Profile and enable the OPTIONS Ping option as shown in the following figure:



Step 2 Add the SIP profile to the SIP trunk and click **Save**:



Note

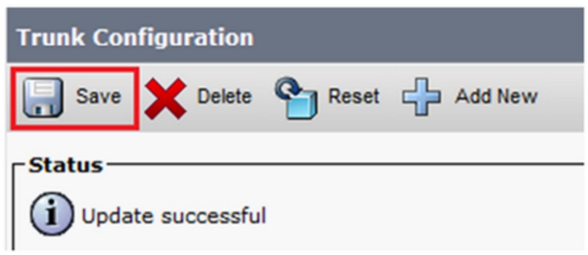
- You must have previously configured this trunk. See System Configuration Guide.
- Set **Status**, **Status Reason**, and **Duration** to N/A.

a) Navigate to **Device** >> **Trunk**.

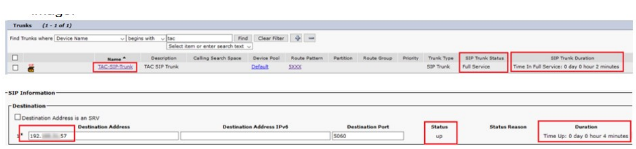
b) Choose the correct SIP profile and click **Save**.

Note

- You must reset the trunk after saving for the changes to take effect.
- The reset disconnects active calls and does not allow any incoming calls for a short time.



c) Monitor the status of the **SIP Trunk** in Cisco Unified Communications Manager.



Step 3 Enable SIP Options Ping on the far end of the SIP Trunk. In this case, 192.X.X.57 (ISR 4351).

a) Navigate to the ISR CUBE or Gateway and confirm what dial-peer you want to add the Options Ping to as shown in the following figure:

```
ISR4351#show running-config | s voice 100
dial-peer voice 100 voip
description CUCM Dial-Peer
session protocol sipv2
session target ipv4:192.x.x.26
dtmf-relay rtp-nte
codec g711ulaw
no vad
```

b) Add Options Ping with the command: voice-class sip options-keepalive using a profile.

```
dial-peer voice 100 voip
description CUCM Dial-Peer
session protocol sipv2
session target ipv4:192.x.x.26
voice-class sip options-keepalive profile 1
dtmf-relay rtp-nte
codec g711ulaw
no vad
```

What to do next

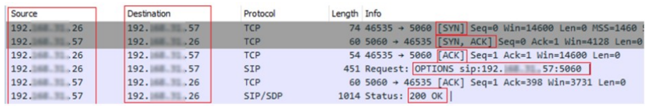
Verify

Confirm that Options messages are exchanged correctly in this section.

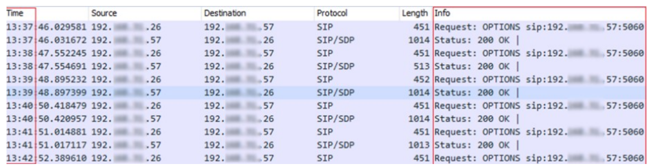
Note

To understand how to run a packet capture on CUCM eth0 port, follow the instructions in this link: Packet Capture on CUCM Appliance Model.

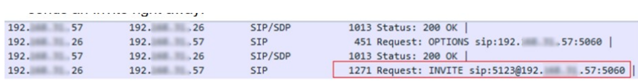
- The TCP three-way handshake is only done once, when you restart the trunk. Afterwards, you only have OPTIONS messages that are sent from CUCM to ISR where you expect a 200 OK as a response. These messages are exchanged every 60 seconds by default.



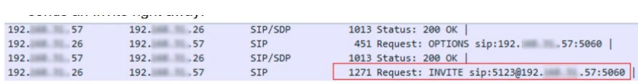
- Options messages are only sent from 192.X.X.26 (CUCM) to 192.X.X.57 (ISR) because only CUCM is configured to monitor the trunk status:



- When you call, CUCM already knows that the trunk is in an operational status and sends an invite:



- If you have completed Step 3 configuration on CUBE, you see that Options messages sent both ways:



Troubleshoot

To troubleshoot Options Ping in CUCM, you need:

- Start with Packet Captures from CUCM Eth0 port. For more details, Packet Capture on CUCM Appliance Model.
- Check detailed Cisco Call Manager traces. Download them with RTMT. See the steps here: How to Collect Traces for CUCM 9.x or Later
- Verify the SIPTrunkOOS Reason codes in this link: System Error Message.
  - Local=1 (request timeout)
  - Local=2 (local SIP stack is not able to create a socket connection with the remote peer)
  - Local=3 (DNS query failed)

To troubleshoot Options Ping in CUBE, you need the following:

- **debug ccsip messages**
- **debug ccsip non-call**

- **debug voip ccapi inout**
- Packet captures from an interface that point toward CUCM.

Additional References

Related Documents

Related Topic	Document Title
Voice commands	Cisco IOS Voice Command Reference
Cisco IOS Commands	Cisco IOS Command List, All Releases
SIP Configuration Guide	
Configuring SIP profiles	
Configuring server groups	