Configure Options Ping Between CUCM and CUBE

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Introduction

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

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Prerequisites

Requirements

Cisco recommends that you have knowledge of these topics:

- Cisco Call Manager Administration
- Cisco Unified Border Element or Gateway Administration
- Session Initiation Protocol *(*SIP)

Components Used

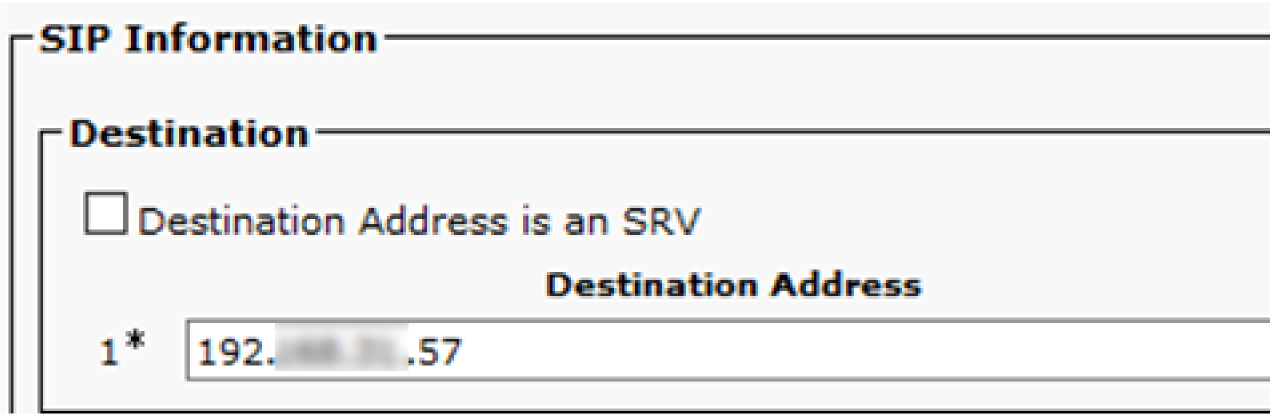
- Cisco Integrated Services Router (ISR4351/K9)
- Cisco Unified Communications Manager 12.0
- Cisco Unified IP Phone

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, ensure that you understand the potential impact of any command.

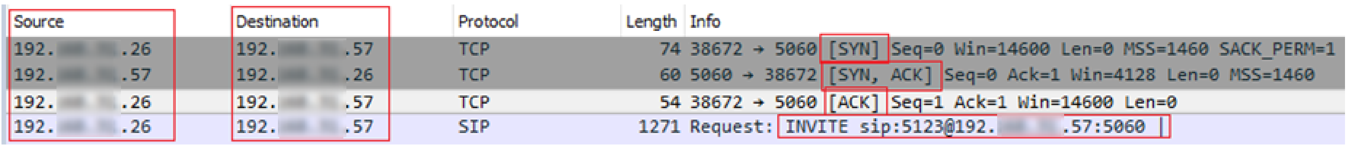
Background Information

It is important to review how CUCM extends a call out of a SIP Trunk as shown below:

For CUCM to extend a call out of a SIP trunk, it proceeds to establish a Transmission Control Protocol (TCP) 3-way handshake with the IP address specified in the Trunk Configuration page as shown in the image:



TCP 3-way handshake in wireshark looks as shown in the image :



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

This is done on a per-call, per node basis; so CUCM is forced to wait for a timeout on the Synchronize (SYN) message or an error from the SIP service before it tries an alternate trunk or GW (Gateway).

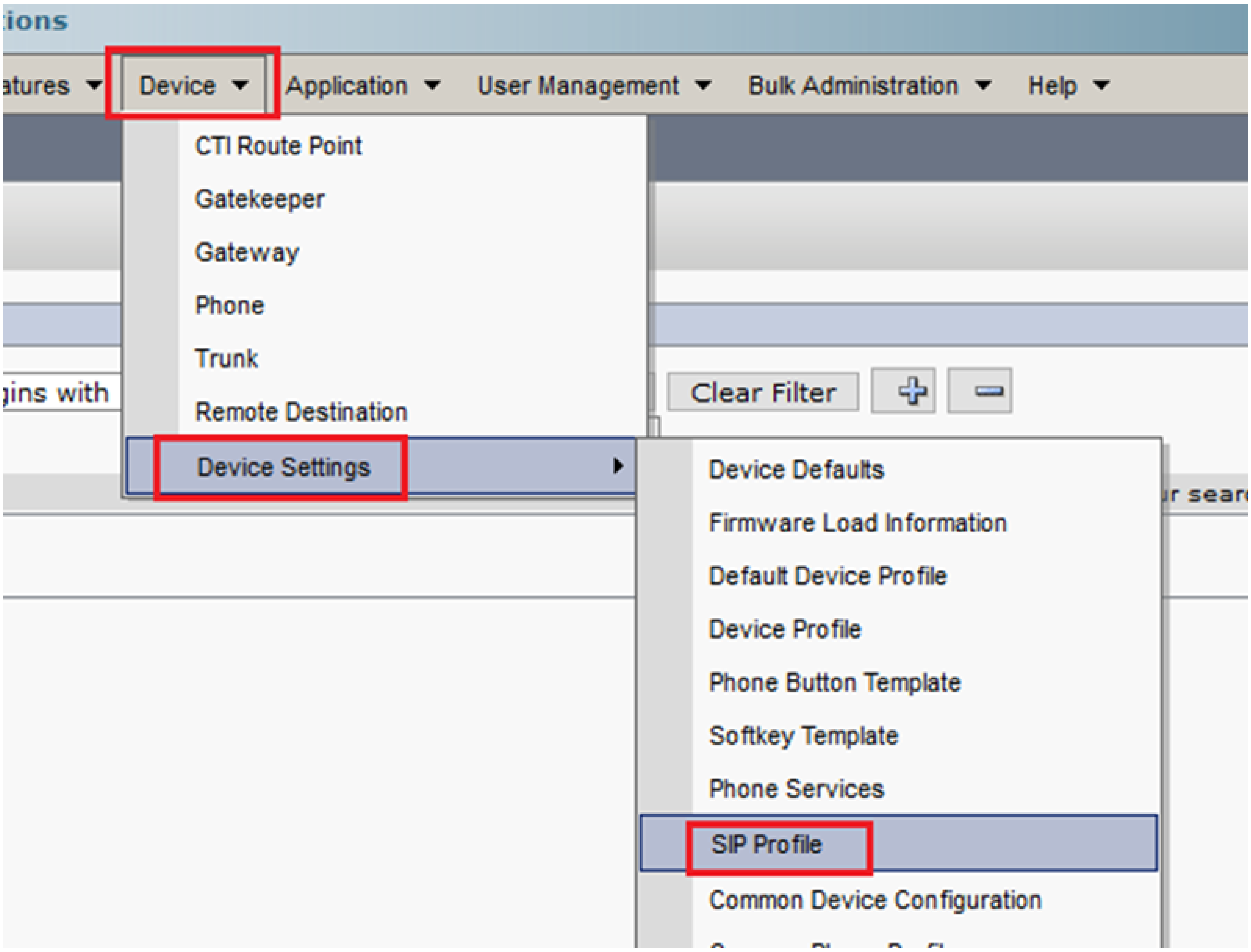
In order to solve this issue, you enable Options Ping and proactively check the status of your SIP trunks.

When you enable Options Ping on your SIP trunk, you also add SIP Trunk Status and uptime statistics where it is possible to monitor the state of each SIP trunk and troubleshoot the moment a trunk goes down. These statistics are seen on the SIP trunk Configuration page.

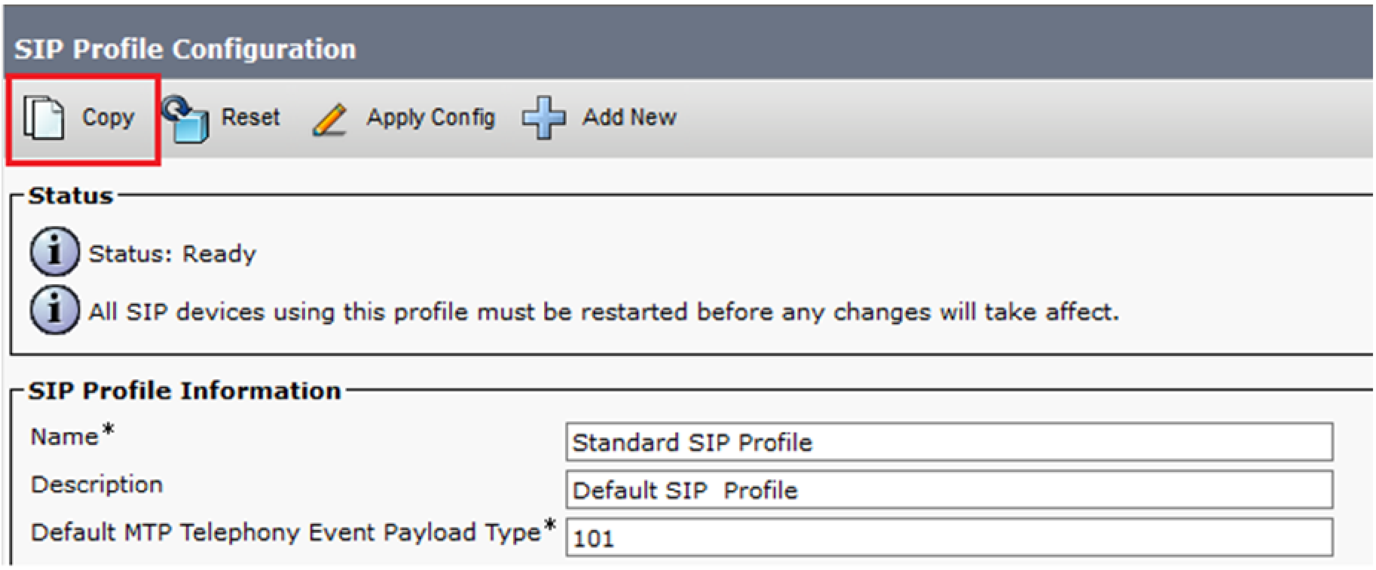
Configure

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

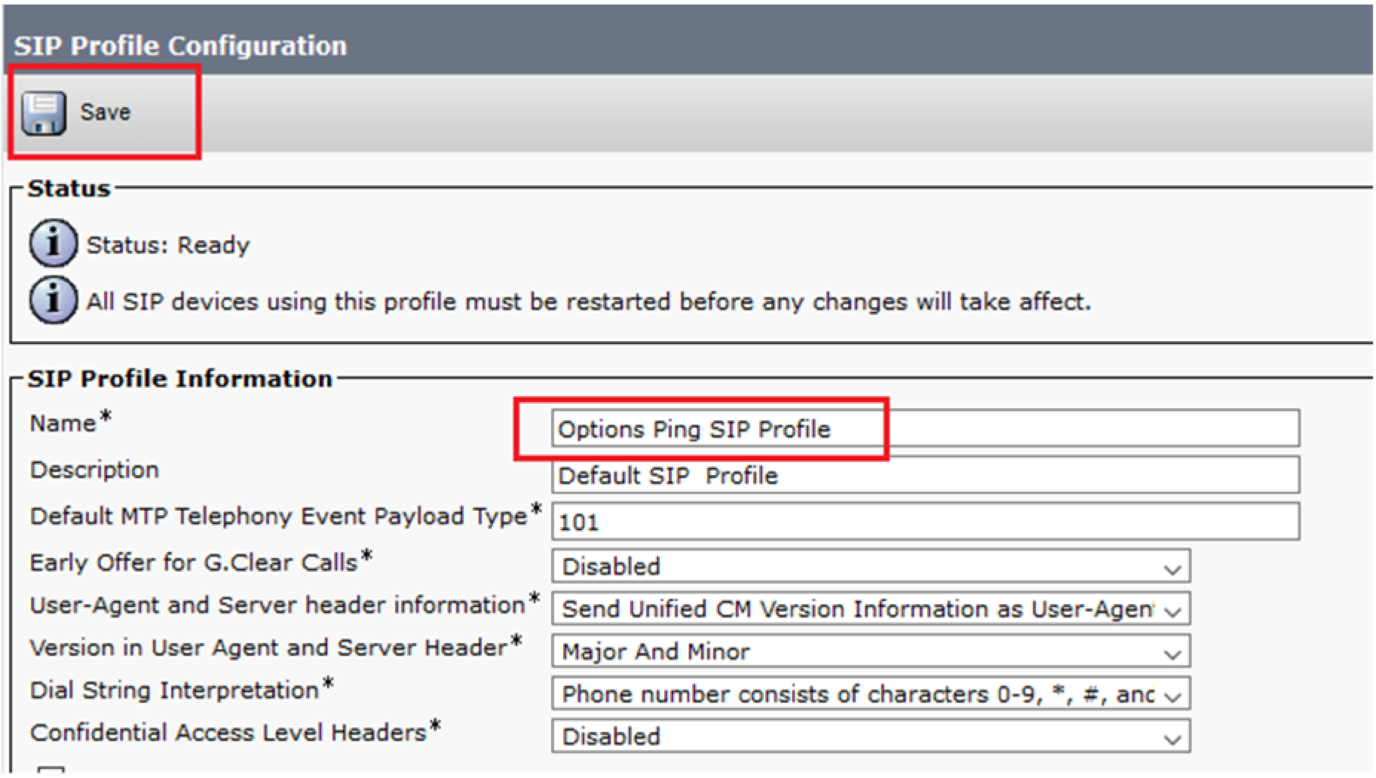
- **Navigate to Cisco Unified CM Administration >> Device >> Device Settings >> SIP Profile** as shown in the image:

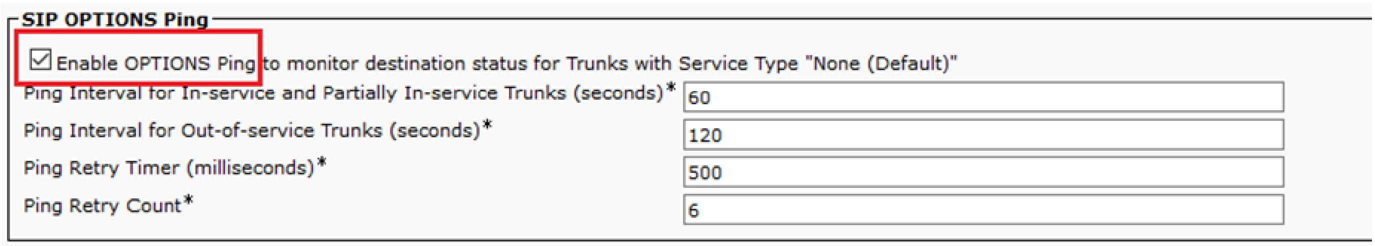


- **Click find** and decide if you want to create a new **SIP Profile**, edit a **SIP Profile** that already exists or make a copy of a SIP Profile. For this example, create a copy of the **Standard SIP Profile** as shown in the images:



- Rename the new SIP Profile and **enable Options Ping** as shown in the image:

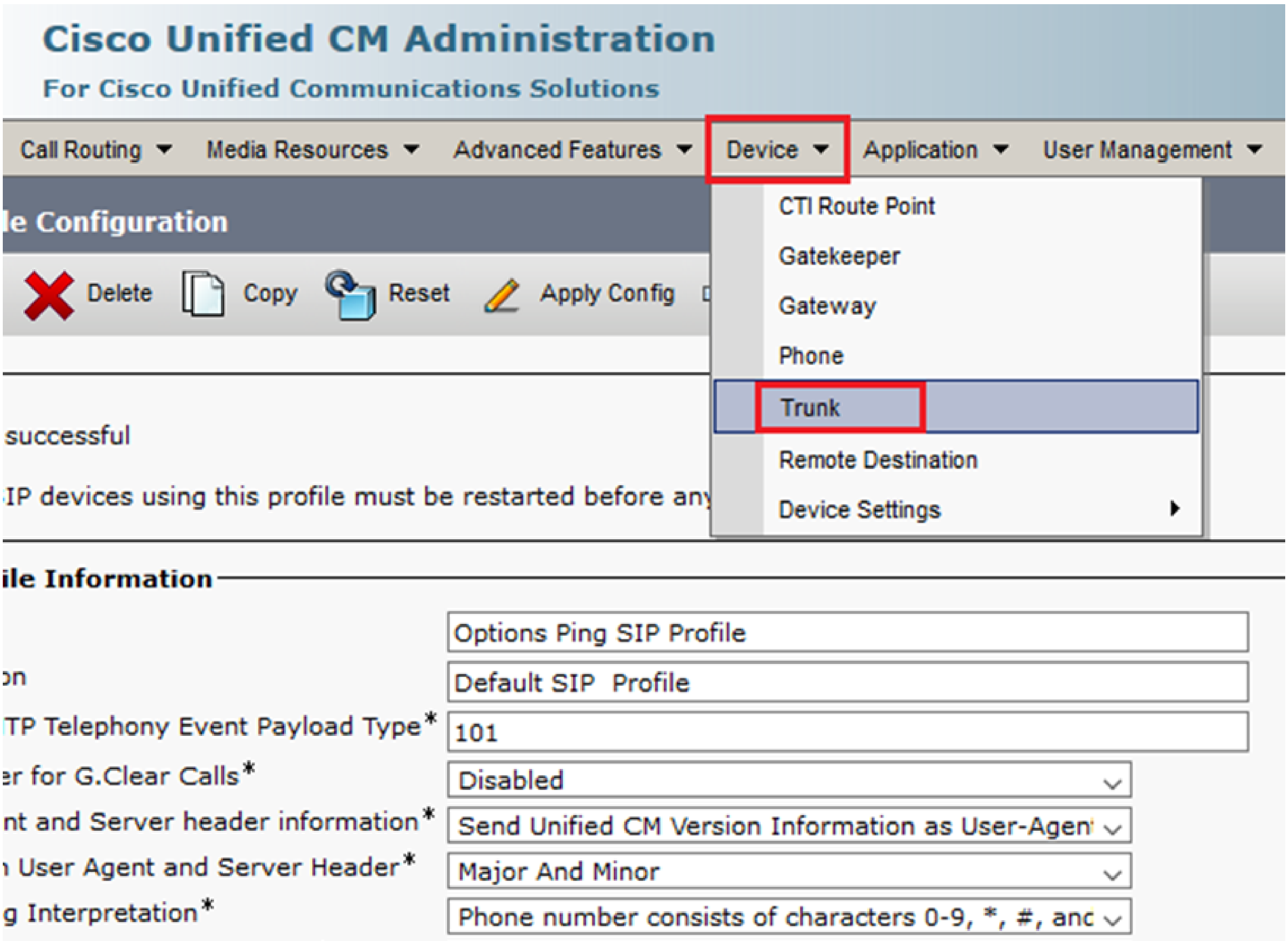


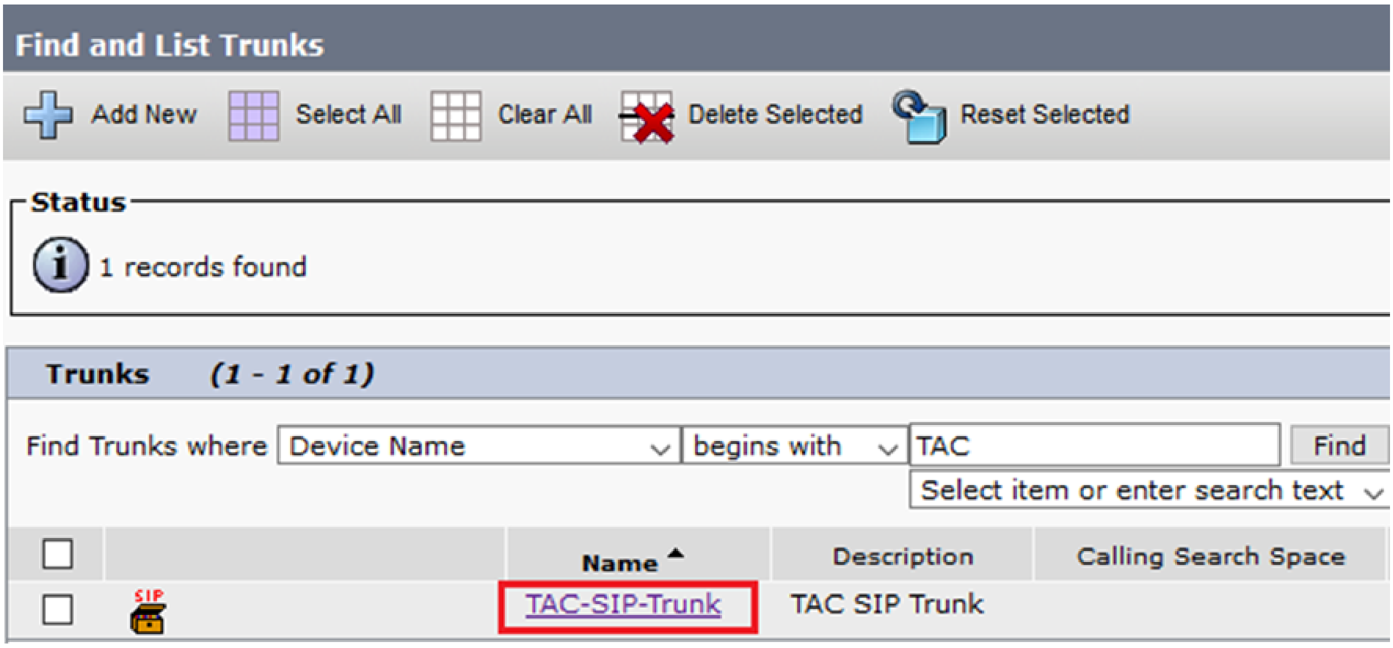


Step 2. **Add the SIP Profile to the SIP trunk** in question and **click Save**:

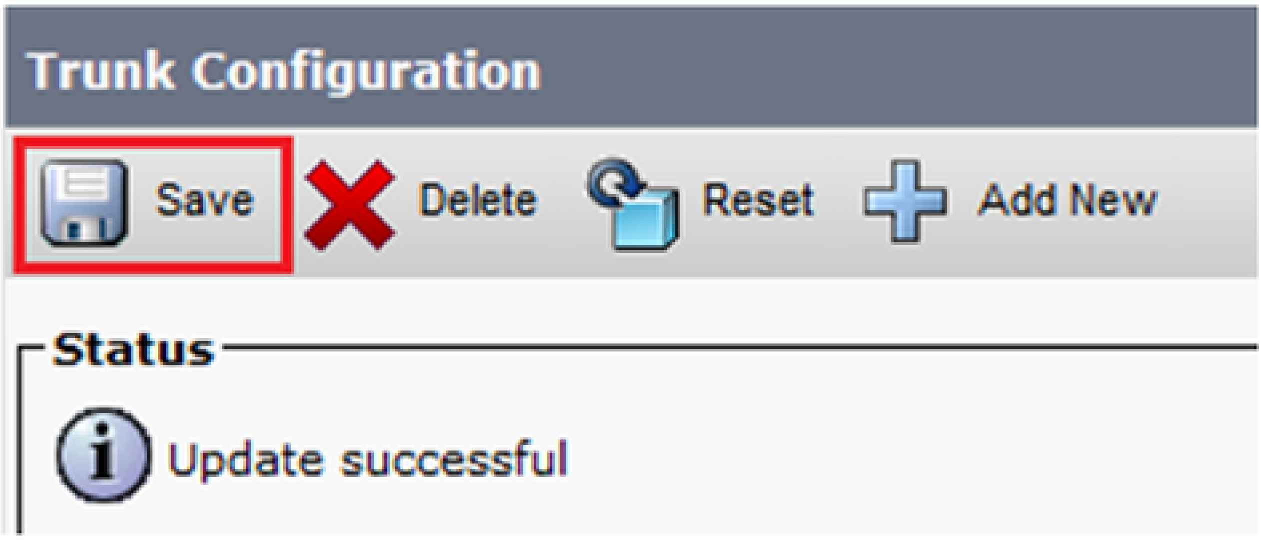
> **Note**: Keep in mind that this trunk must have been previously configured. If you need guidance on how to configure a SIP trunk, visit the link: [System Configuration Guide]

- **Navigate to Device >> Trunk** and choose the trunk you want to edit as shown in the image:

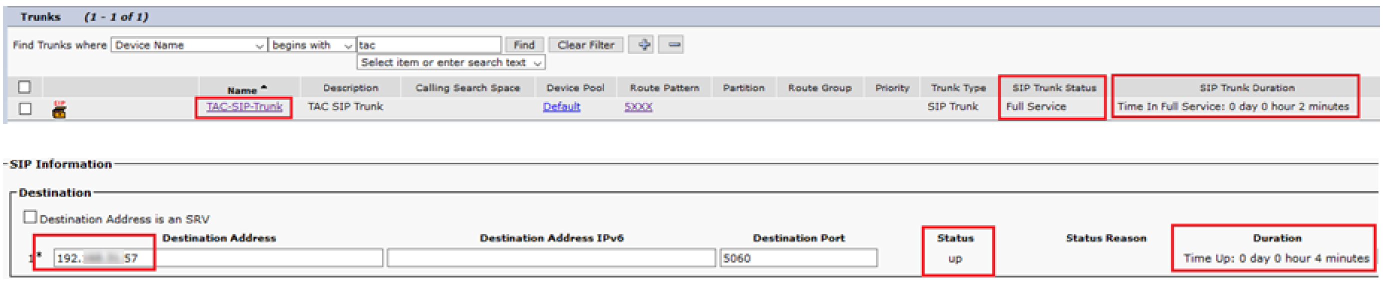




- Notice that the Status, Status Reason, and Duration are set to N/A.
- **Choose the correct SIP Profile, and click Save**



- At this point CUCM must be able to monitor the status of the **SIP trunk** as shown in the image:



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

- Navigate to the ISR Cisco Unified Border Element or Gateway and confirm what dial-peer you want to add the Options Ping to as shown in the image:

```
LESQUIVE-4351-A(config)#do show run | sec dial-peer voice 100
dial-peer voice 100 voip
 description CUCM dial-peer
 session protocol sipv2
 session target ipv4:192.[redacted].26
 dtmf-relay rtp-nte sip-kpml
 codec g711ulaw
```

- Add Options Ping with the command: **voice-class sip options-keepalive** as shown in the image:

```
LESQUIVE-4351-A(config)#do show run | sec dial-peer voice 100
dial-peer voice 100 voip
 description CUCM dial-peer
 session protocol sipv2
 session target ipv4:192.[redacted].26
 dtmf-relay rtp-nte sip-kpml
 codec g711ulaw
LESQUIVE-4351-A(config)#dial-peer voice 100
LESQUIVE-4351-A(config-dial-peer)#voice-class sip options-keepalive
```

Verify

Use this section in order to confirm that Options messages are exchanged correctly.

> **Note**: If you need to understand how to run a packet capture on CUCM eth0 port, follow the instructions in this link: [Packet Capture on CUCM Appliance Model]

- Notice that the TCP 3-way handshake is only done once, when the trunk is restarted and afterwards we only have OPTIONS messages sent from CUCM to ISR where a 200 OK is expected as a response. These messages are exchanged every 60 seconds by default.

Source	Destination	Protocol	Length	Info
192.[redacted].26	192.[redacted].57	TCP	74	46535 → 5060 [SYN] Seq=0 Win=14600 Len=0 MSS=1460
192.[redacted].57	192.[redacted].26	TCP	60	5060 → 46535 [SYN, ACK] Seq=0 Ack=1 Win=4128 Len=0
192.[redacted].26	192.[redacted].57	TCP	54	46535 → 5060 [ACK] Seq=1 Ack=1 Win=14600 Len=0
192.[redacted].26	192.[redacted].57	SIP	451	Request: OPTIONS sip:192.[redacted].57:5060
192.[redacted].57	192.[redacted].26	TCP	60	5060 → 46535 [ACK] Seq=1 Ack=398 Win=3731 Len=0
192.[redacted].57	192.[redacted].26	SIP/SDP	1014	Status: 200 OK |

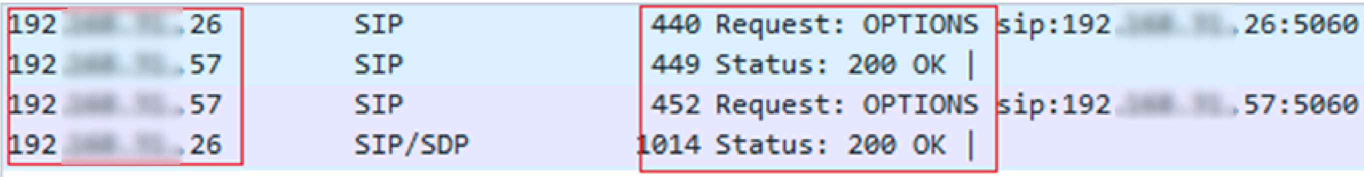
- Notice that Options messages are only sent from 192.X.X.26 (CUCM) to 192.X.X.57 (ISR) because only CUCM is configured to monitor the trunk status:

Time	Source	Destination	Protocol	Length	Info
13:37:46.029581	192.[redacted].26	192.[redacted].57	SIP	451	Request: OPTIONS sip:192.[redacted].57:5060
13:37:46.031672	192.[redacted].57	192.[redacted].26	SIP/SDP	1014	Status: 200 OK |
13:38:47.552245	192.[redacted].26	192.[redacted].57	SIP	451	Request: OPTIONS sip:192.[redacted].57:5060
13:38:47.554691	192.[redacted].57	192.[redacted].26	SIP/SDP	513	Status: 200 OK |
13:39:48.895232	192.[redacted].26	192.[redacted].57	SIP	452	Request: OPTIONS sip:192.[redacted].57:5060
13:39:48.897399	192.[redacted].57	192.[redacted].26	SIP/SDP	1014	Status: 200 OK |
13:40:50.418479	192.[redacted].26	192.[redacted].57	SIP	451	Request: OPTIONS sip:192.[redacted].57:5060
13:40:50.420957	192.[redacted].57	192.[redacted].26	SIP/SDP	1014	Status: 200 OK |
13:41:51.014881	192.[redacted].26	192.[redacted].57	SIP	451	Request: OPTIONS sip:192.[redacted].57:5060
13:41:51.017117	192.[redacted].57	192.[redacted].26	SIP/SDP	1013	Status: 200 OK |
13:42:52.389610	192.[redacted].26	192.[redacted].57	SIP	451	Request: OPTIONS sip:192.[redacted].57:5060

- Now when a call is made, CUCM already knows the trunk is in an operational status and sends an Invite right away:

192.[redacted].57	192.[redacted].26	SIP/SDP	1013	Status: 200 OK |
192.[redacted].26	192.[redacted].57	SIP	451	Request: OPTIONS sip:192.[redacted].57:5060 |
192.[redacted].57	192.[redacted].26	SIP/SDP	1013	Status: 200 OK |
192.[redacted].26	192.[redacted].57	SIP	1271	Request: INVITE sip:5123@192.[redacted].57:5060 |

- If you did step 3 (Optional configuration on CUBE) you see Options messages sent both ways:



Troubleshoot

- In order to troubleshoot Options Ping in CUCM, you need:

- The best option to start is with a Packet Captures from CUCM Eth0 port, more details: Packet Capture on CUCM Appliance Model

  Open the capture with 3party free software Wireshark, and filter with SIP

- You can also check detailed Cisco Callmanager traces, download them with RTMT, find 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)

- In order to troubleshoot Options Ping in ISR4351, you need:

- Debug ccsip messages
- Debug ccapi inout
- Packet Captures from interface that points towards CUCM