Configuration Guide for Google CES Agent Handoff Using Avaya SBC V10.2.1.1-104-25336

TekVizion

Table of Contents

1 Audience

This document is intended for the SIP Trunk customer's technical staff and Value-Added Reseller (VAR) having installation and operational responsibilities.

1.1 Introduction

This configuration guide describes configuration steps for **Google CES Agent Handoff** using **Avaya SBC V10.2.1.1-104-25336**.

1.1.1 TekVizion Labs

TekVizion Labs™ is an independent testing and verification facility offered by TekVizion, Inc. TekVizion Labs offers several types of testing services including:

- Remote Testing – provides secure, remote access to certain products in TekVizion Labs for pre-Verification and ad hoc testing.
- Verification Testing – Verification of interoperability performed on-site at TekVizion Labs between two products or in a multi-vendor configuration.
- Product Assessment – independent assessment and verification of product functionality, interface usability, assessment of differentiating features as well as suggestions for added functionality, stress, and performance testing, etc.

TekVizion is a systems integrator specifically dedicated to the telecommunications industry. Our core services include consulting/solution design, interoperability/Verification testing, integration, custom software development and solution support services. Our services help service providers achieve a smooth transition to packet-voice networks, speeding delivery of integrated services. While we have expertise covering a wide range of technologies, we have extensive experience surrounding our practice areas which include SIP Trunking, Packet Voice, Service Delivery, and Integrated Services.

The TekVizion team brings together experience from the leading service providers and vendors in telecom. Our unique expertise includes legacy switching services and platforms, and unparalleled product knowledge, interoperability, and integration experience on a vast array of VoIP and other next-generation products. We rely on this combined experience to do what we do best: help our clients advance the rollout of services that excite customers and result in new revenues for the bottom line. TekVizion leverages this real-world, multi-vendor integration and test experience and proven processes to offer services to vendors, network operators, enhanced service providers, large enterprises and other professional services firms. TekVizion's headquarters, along with a state-of-the-art test lab and Executive Briefing Center, is located in Plano, Texas.

For more information on TekVizion and its practice areas, please visit TekVizion Labs website.

2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of Google CES Agent Handoff with Avaya SBC V10.2.1.1-104-25336 configuration.

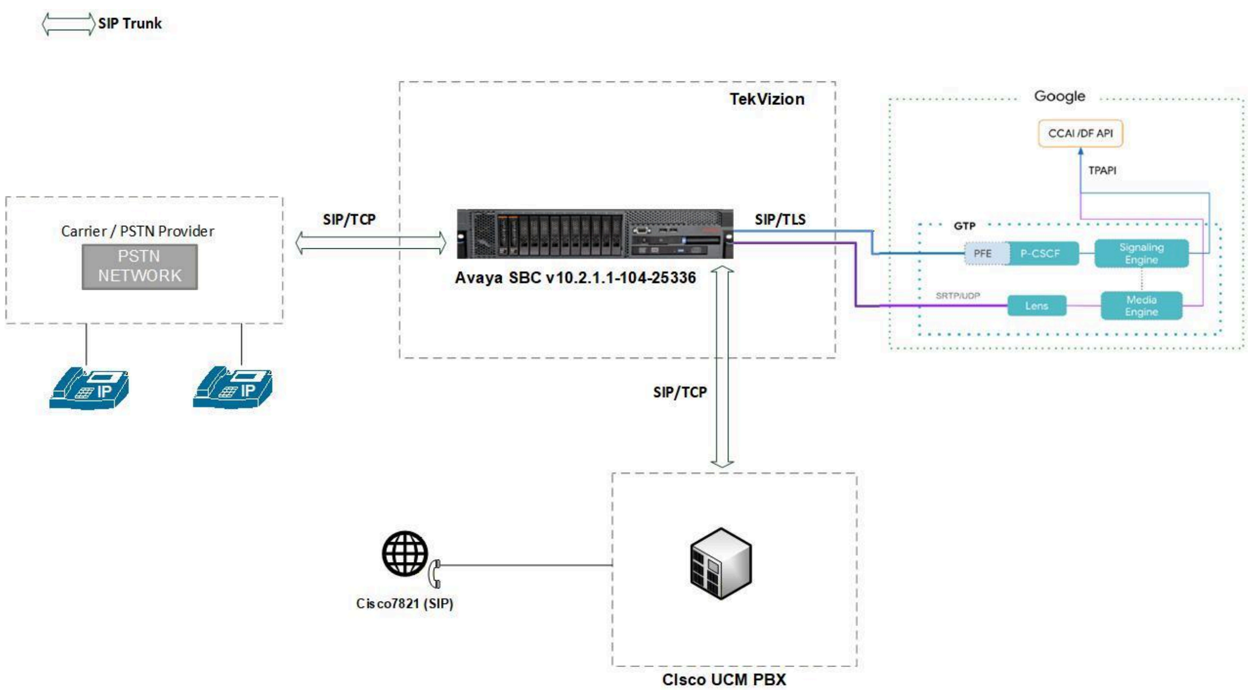

Figure 1: SIP Trunk Lab Reference Network

The lab network consists of the following components:

- Google CES Cloud Environment
- Avaya SBC V10.2.1.1-104-25336
- OnPrem PBX (Cisco UCM)
- PSTN Gateway

3 Hardware Components

- Running on ESXi- 7.0.3: Avaya SBC V10.2.1.1-104-25336

4 Software Requirements

- Avaya SBC software version: 10.2.1.1-104-25336
- Cisco Unified Communications Manager V15.0.1.11901-2

5 Google CES Certified Avaya SBC Version

Table 1 – Google CES Certified Avaya SBC Version

Google CES Certified Avaya SBC Version	
Avaya SBC	10.2.1.1-104-25336

6 Features

6.1 Caveats and Limitations

DTLS	DTLS towards Google CES is not tested

6.2 Failed Testcase

- None

7 Configuration

7.1 Configuration Checklist

Below are the steps that are required to configure Avaya SBC.

Table 2 – Avaya SBC Configuration Steps

Step	Description	Reference
Step 1	Avaya SBC Login	Section 7.4.1
Step 2	Server Interworking	Section 7.4.2
Step 3	SIP Servers	Section 7.4.3
Step 4	Topology Hiding	Section 7.4.4
Step 5	Routing	Section 7.4.5
Step 6	Signaling Manipulation	Section 7.4.6
Step 7	Media Rules	Section 7.4.7
Step 8	End Point Policy Groups	Section 7.4.8
Step 9	Media Interface	Section 7.4.9
Step 10	Network Management	Section 7.4.10
Step 11	Signaling Interface	Section 7.4.11
Step 12	End Point Flow	Section 7.4.12
Step 13	TLS Configuration	Section 7.4.13

7.2 IP Address Worksheet

The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document are for **illustrative purposes only**.

Table 3 – IP Address Worksheet

Component	IP Address
Google CES	
Signaling	us.telephony.goog:5672
Media	74.125.X.X
OnPrem PBX	
LAN IP Address	10.80.X.X
Avaya SBC	
LAN IP Address	10.80.X.X
WAN IP Address	192.65.X.X

7.3 Google CES API Configuration

Below link can be referred to configuring Google CES API configuration for Agent Handoff.

https://docs.cloud.google.com/contact-center/insights/docs/troubleshooting

7.4 Avaya SBC Configuration

The following configuration is implemented on the Avaya SBC for Google CES Agent Handoff.

7.4.1 Avaya SBC Login

- Log into Avaya SBC web interface by typing "https://X.X.X.X/sbc".
- Enter the Username and Password
- Click Log In

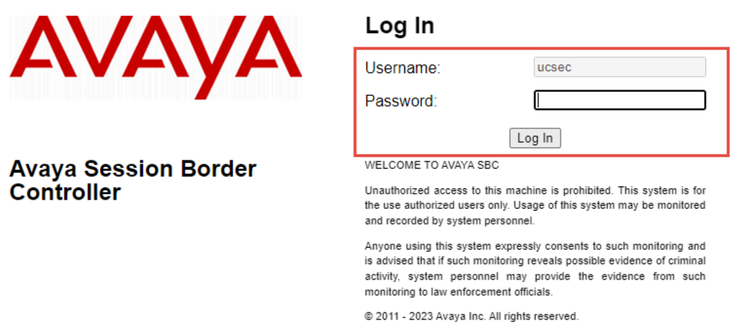

Figure 2: Avaya SBC Login

- Navigate to Device and select (SA) from drop down to expand the configuration for Avaya SBC.
- Device Management displays the system version and current operational status.

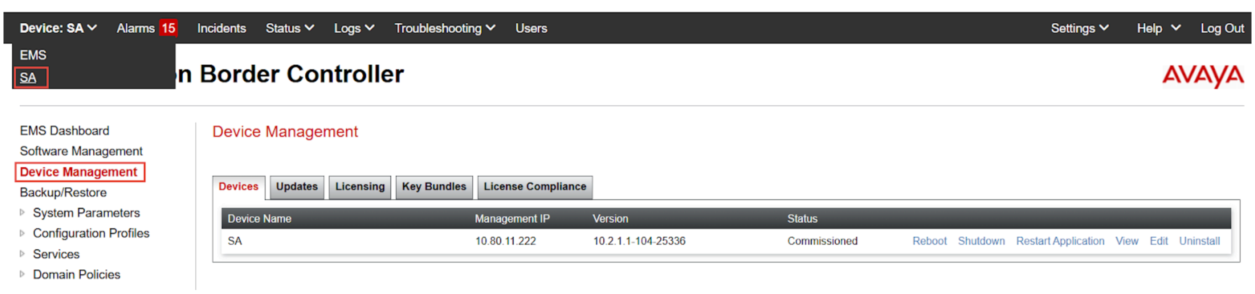

Figure 3: Selection of Avaya SBC Device

7.4.2 Server Interworking

Server Interworking for OnPrem PBX

- Navigate: **Configuration Profiles** ▯ **Server Interworking**
- Select the default Interworking Profile avaya-ru, click **Clone**

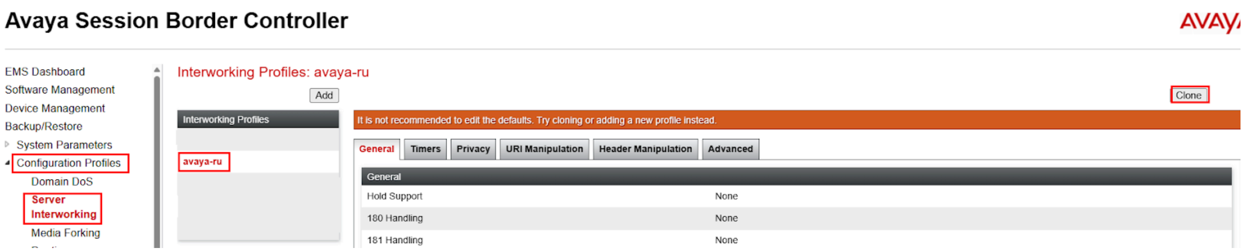

Figure 4: Server Interworking Profile for OnPrem PBX

- Set Clone Name: **CUCM**
- Click **Finish**

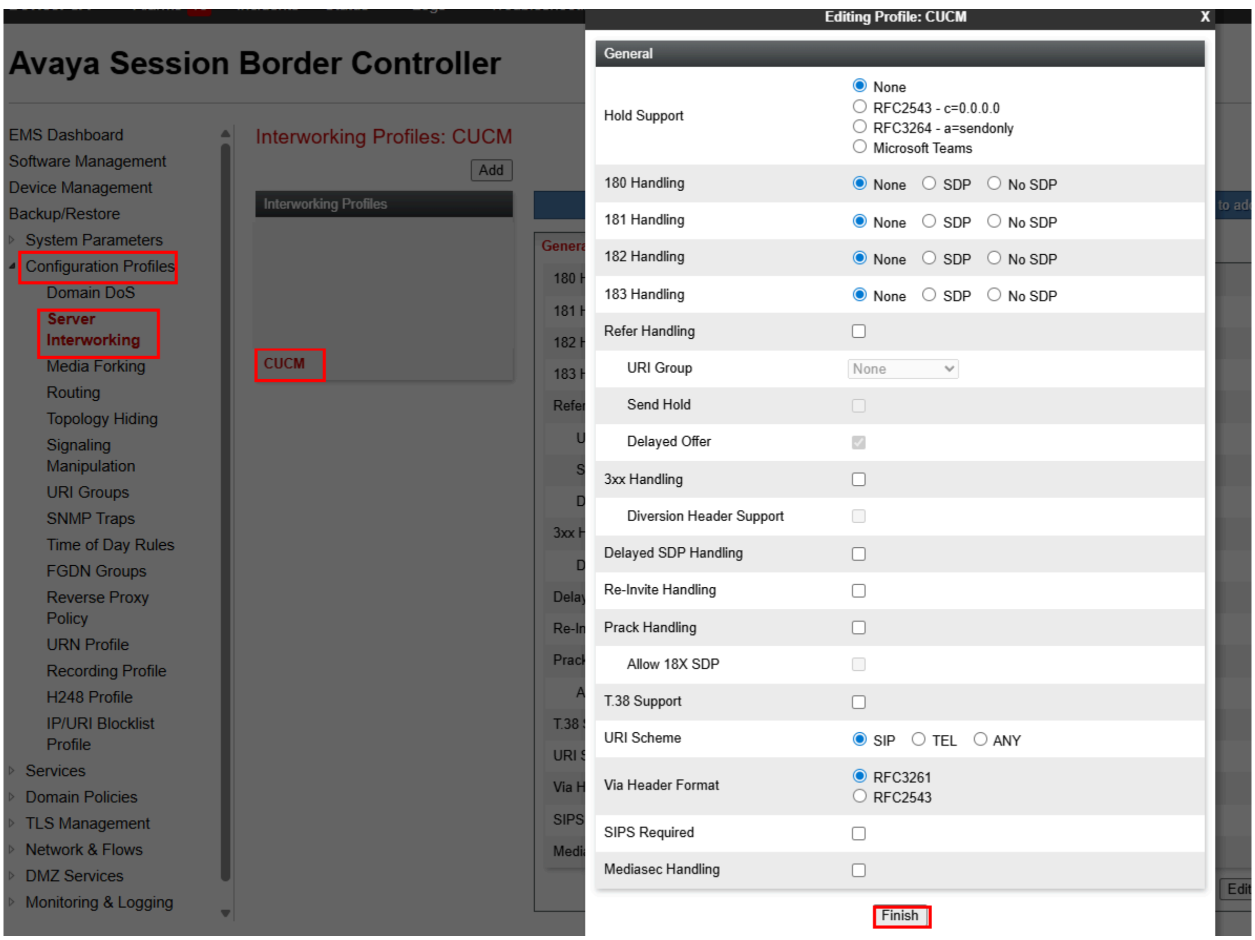

Figure 5: Server Interworking Profile for OnPrem PBX (Cont.)

Server Interworking for **Google CES**

- Navigate: **Configuration Profiles** □ **Server Interworking**
- Set Name: **Google**
- SIPS Required: **Unchecked**
- Click **Finish**

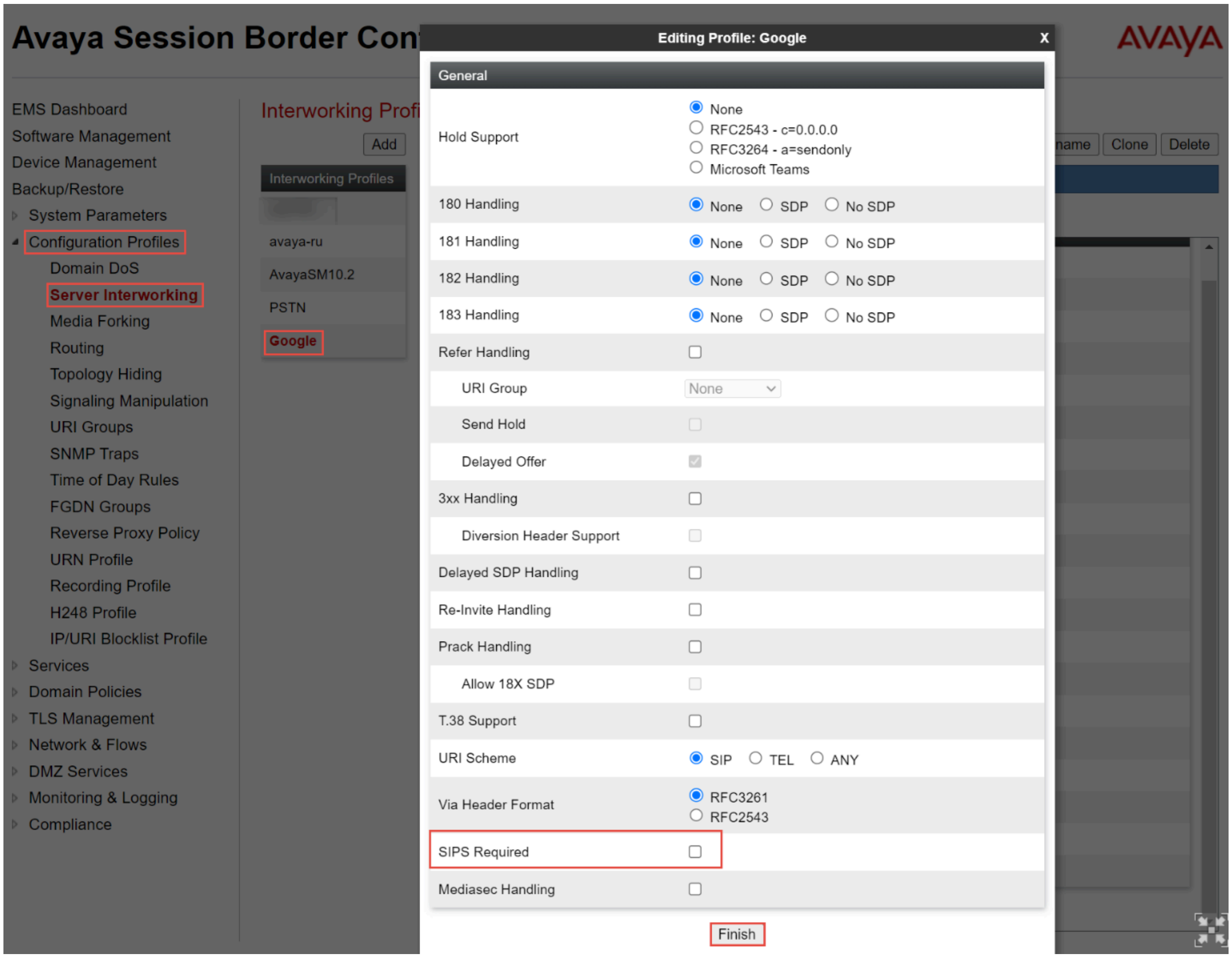

Figure 6: Server Interworking Profile for Google CES

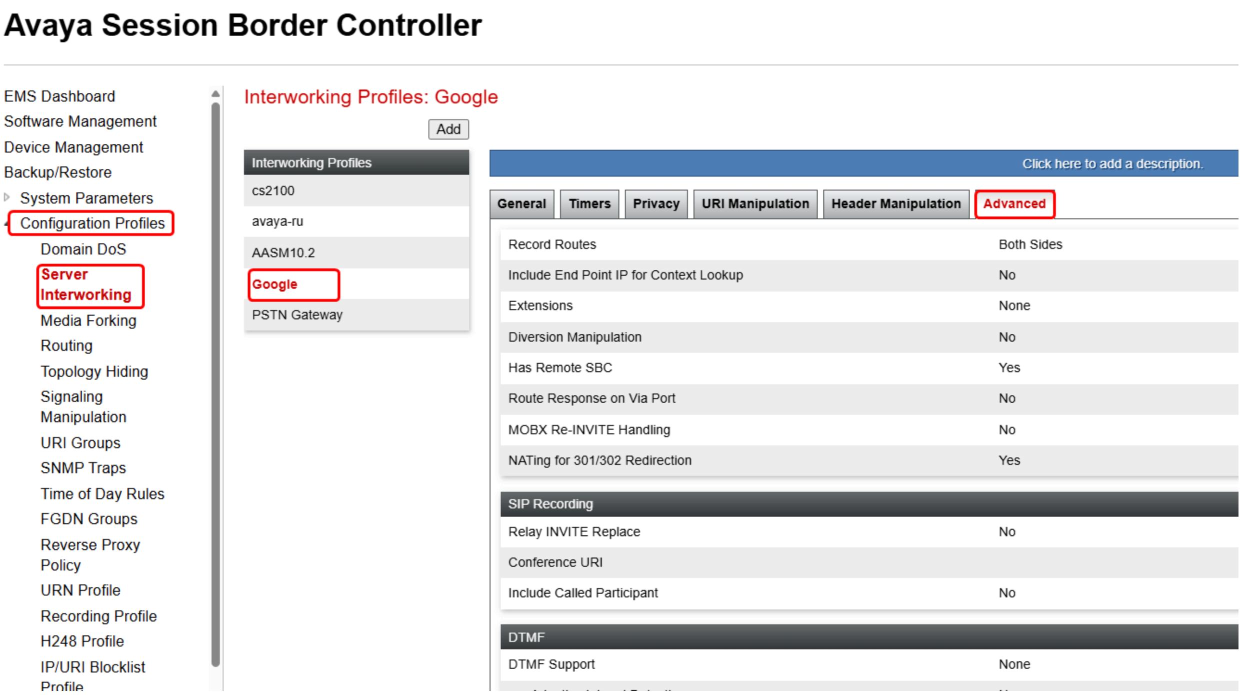

Figure 7: Server Interworking Profile for Google CES (Cont.)

Server Interworking for **PSTN Gateway**

- Repeat the same procedure to create the Interworking Profile towards **PSTN Gateway**
- Set Refer Handling: **Checked**
- Click **Finish**

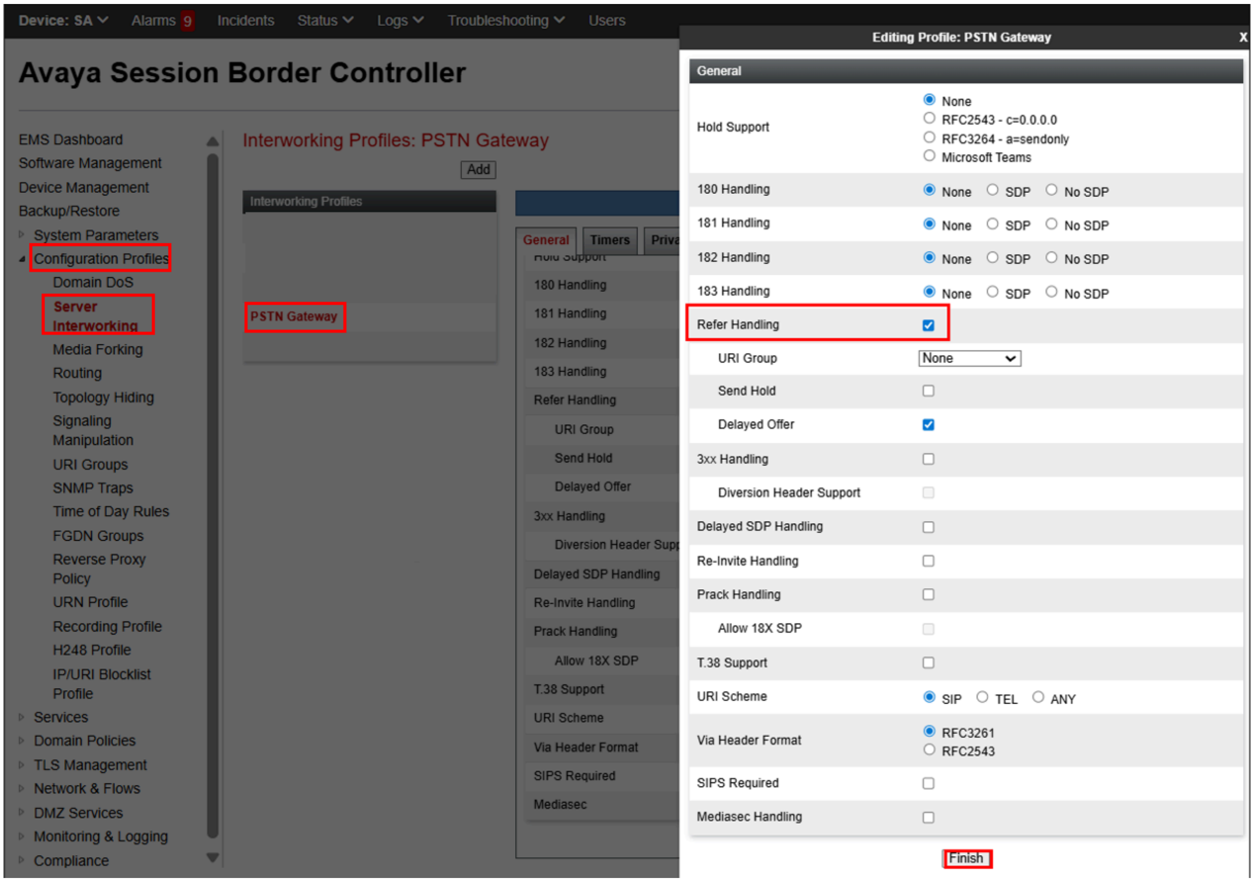

Figure 8: Server Interworking Profile for PSTN Gateway

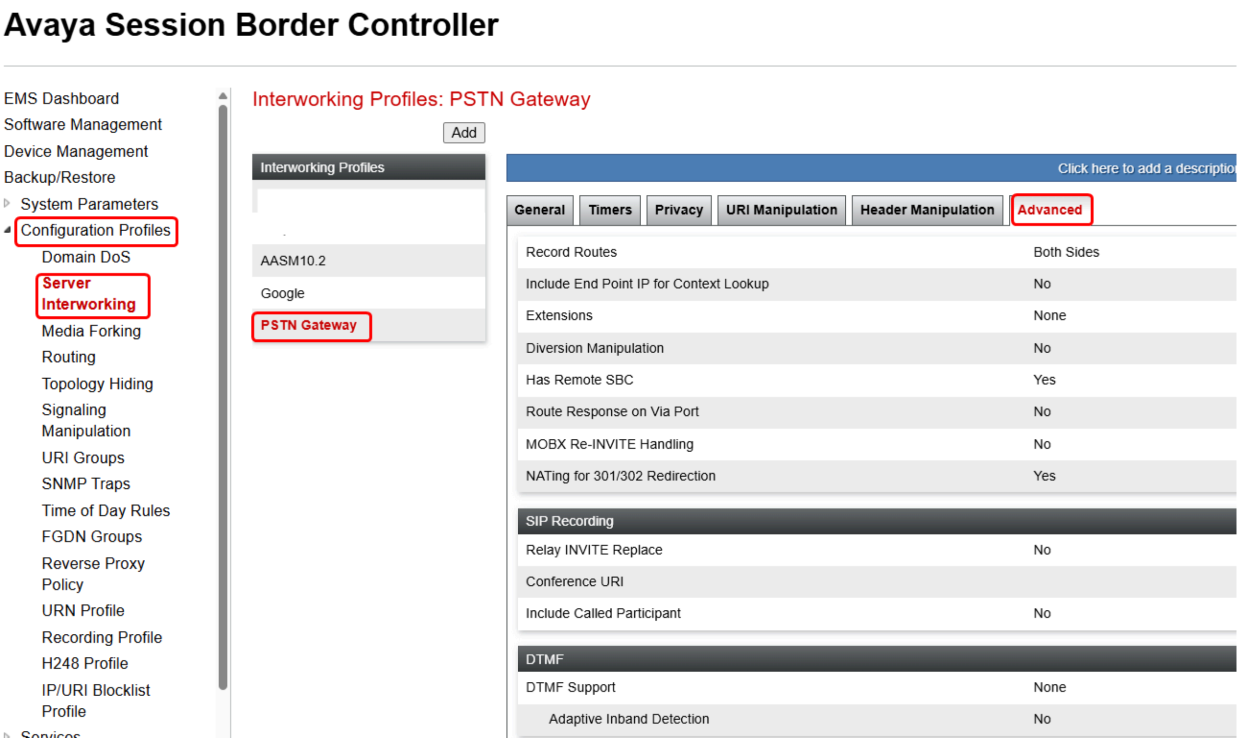

Figure 9: Server Interworking Profile for PSTN Gateway (Cont.)

7.4.3 SIP Servers

SIP Server for **OnPrem PBX**

- Navigate: **Services** □ **SIP Servers**
- Click **Add**
- Set Profile Name: **CUCM**
- Click **Next**

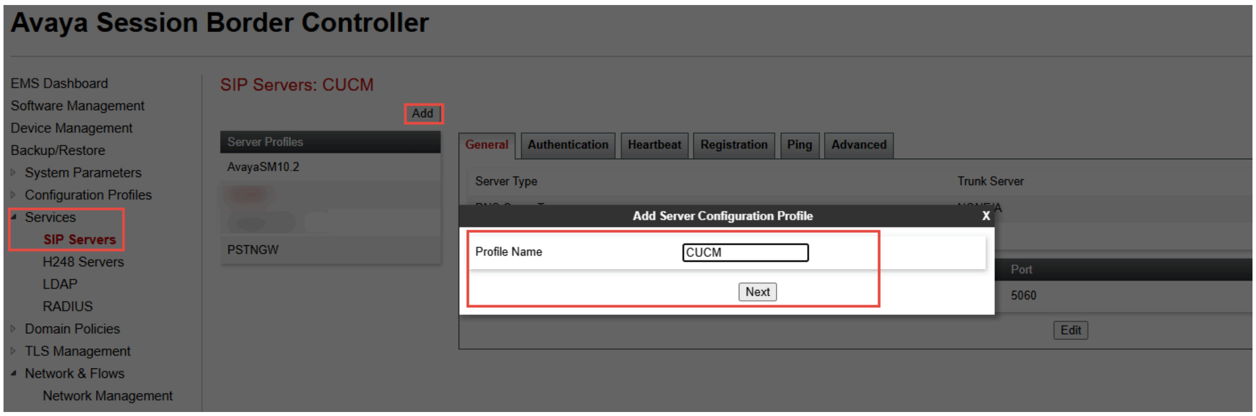

Figure 10: SIP Server for OnPrem PBX

- Set Server Type: Select **Trunk Server** from the drop down
- Set IP Address/FQDN/CIDR Range: **10.80.X.X**
- Set Port: **5060**
- Set Transport: **TCP**
- Click **Finish**

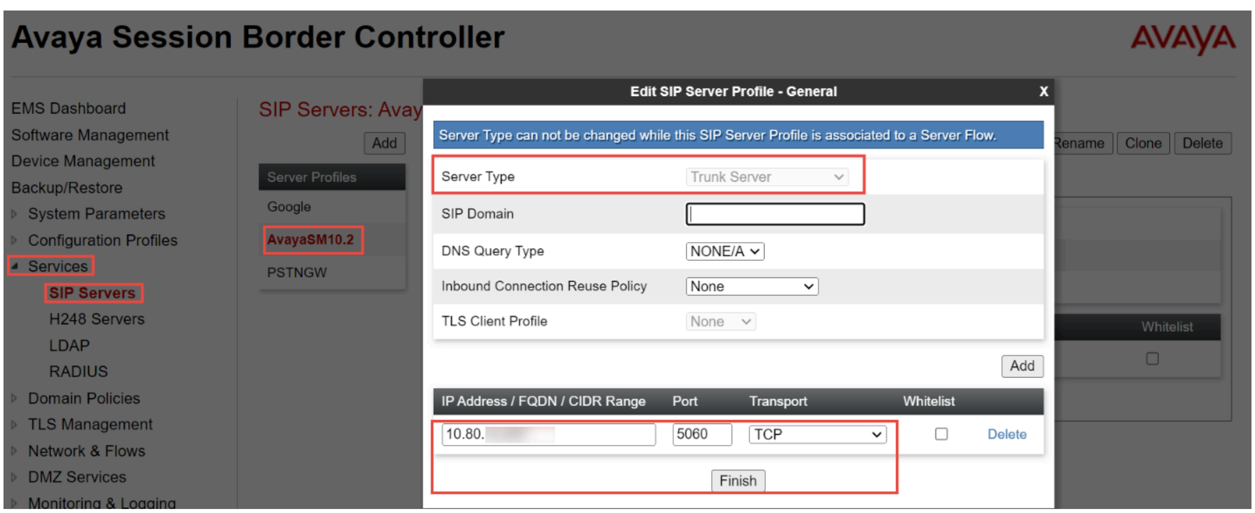

Figure 11: SIP Server for OnPrem PBX (Cont.)

- Navigate: **Heartbeat** tab
- Set Enable Heartbeat: **Checked**
- Set Method: **OPTIONS**
- Set Retry Timeout on Connection Failure: **30 seconds**
- Set Frequency: **60 seconds**
- Set From URI: **ping@10.80.X.X**
- Set To URI: **ping@10.80.X.X**
- Click **Finish**

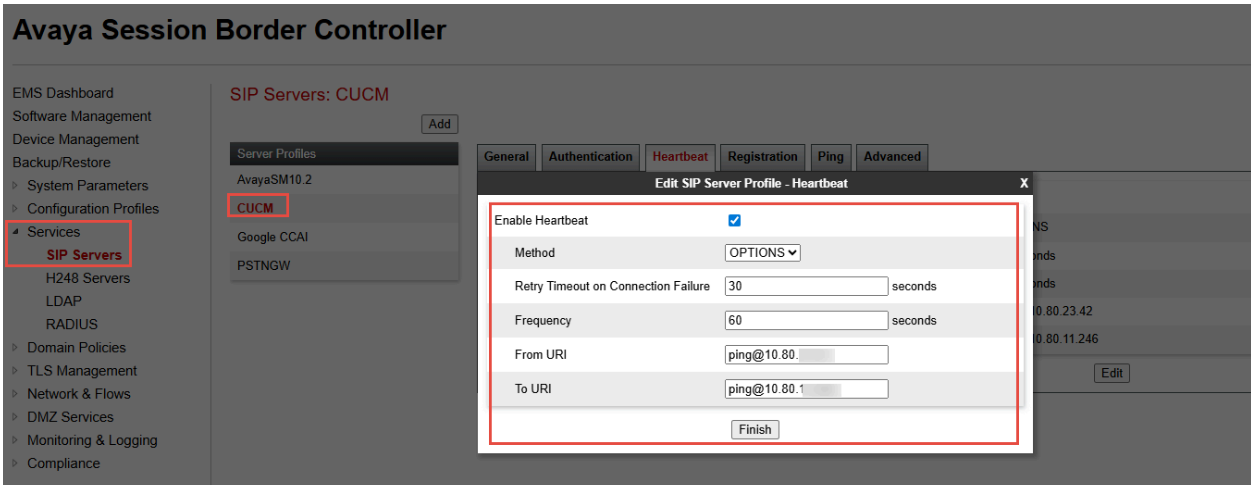

Figure 12: SIP Server for OnPrem PBX (Cont.)

- Navigate: **Ping** tab
- Set Enable Ping: **Checked**
- Set Ping interval: **60 seconds**
- Set Response Timeout: **30 seconds**
- Click **Finish**

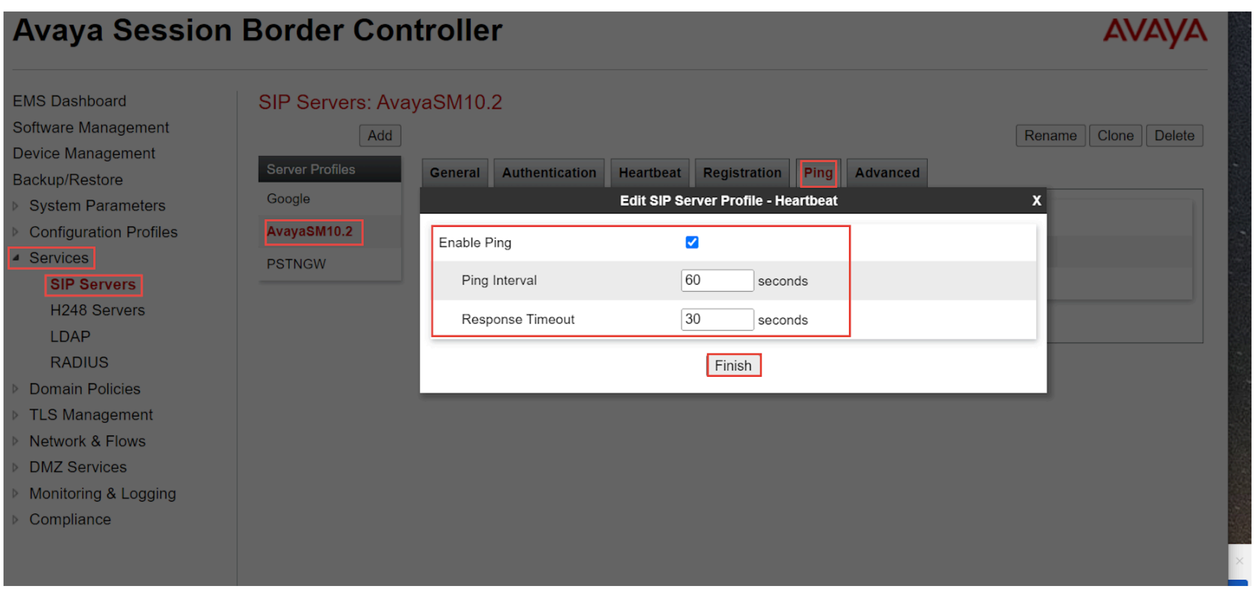

Figure 13: SIP Server for OnPrem PBX (Cont.)

- Navigate: **Advanced** tab
- Set Enable Grooming: **Checked**
- Set Interworking Profile: Select **CUCM.** Refer Section 7.4.2
- Click **Finish**

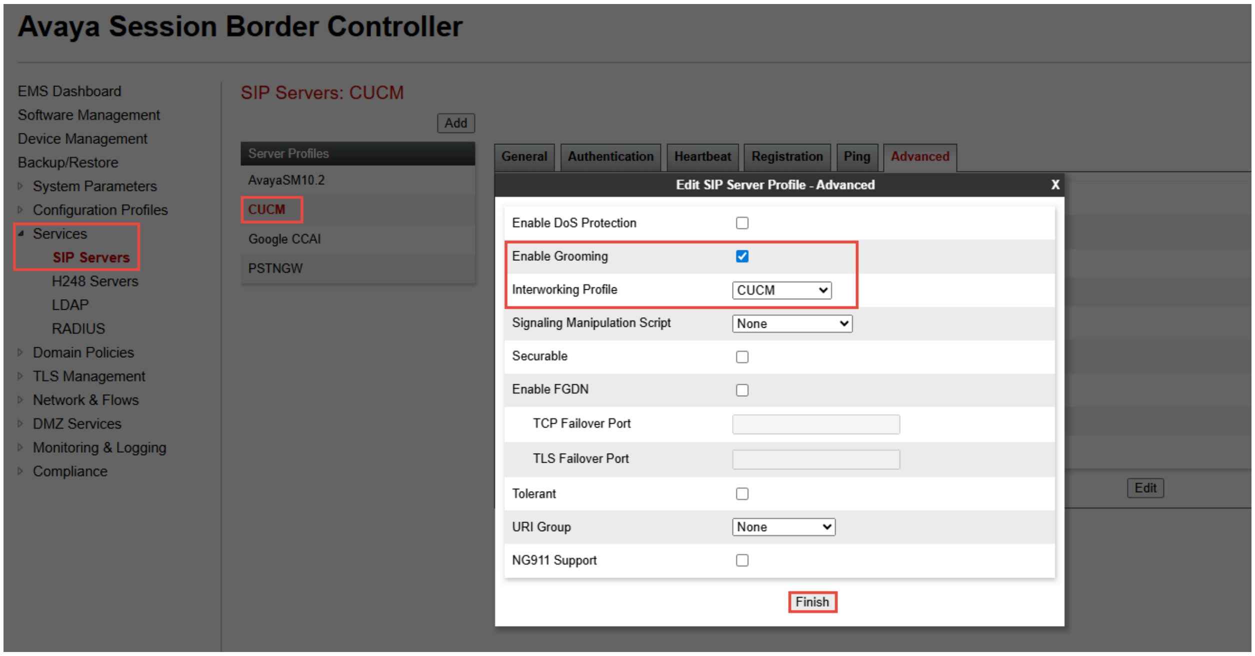

Figure 14: SIP Server for OnPrem PBX (Cont.)

SIP Server for Google CES

- Navigate: **Services** ⎕ **SIP Servers**
- Click **Add**
- Set Profile Name: **Google**
- Click **Next**

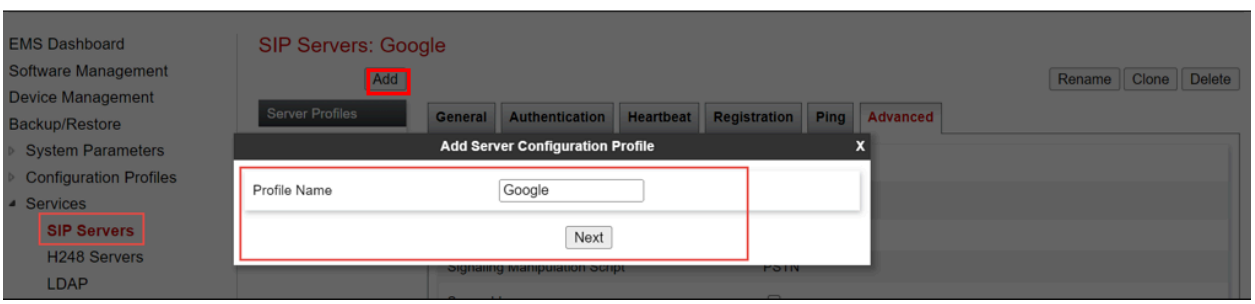

Figure 15: SIP Server for Google CES

- Set Server Type: Select **Trunk Server** from the drop down
- Set TLS Client Profile: **Google.** Refer Section 7.4.13
- Set IP Address/FQDN/CIDR Range: **us.telephony.goog**
- Set Port: **5672**
- Set Transport: **TLS**
- Click **Finish**

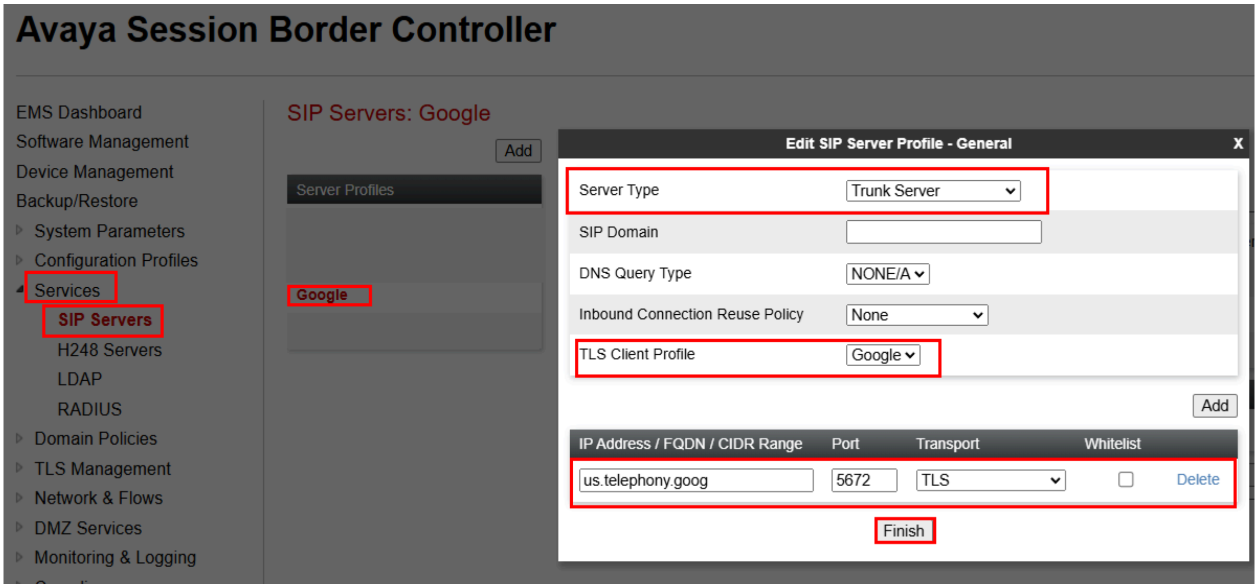

Figure 16: SIP Server for Google CES (Cont.)

- Navigate: **Heartbeat** tab
- Set Enable Heartbeat: **Checked**
- Set Method: **OPTIONS**
- Set Retry Timeout on Connection Failure: **30 seconds**
- Set Frequency: **30 seconds**
- Set From URI: **ping@192.65.X.X**
- Set To URI: **ping@us.telephony.goog**
- Click **Finish**

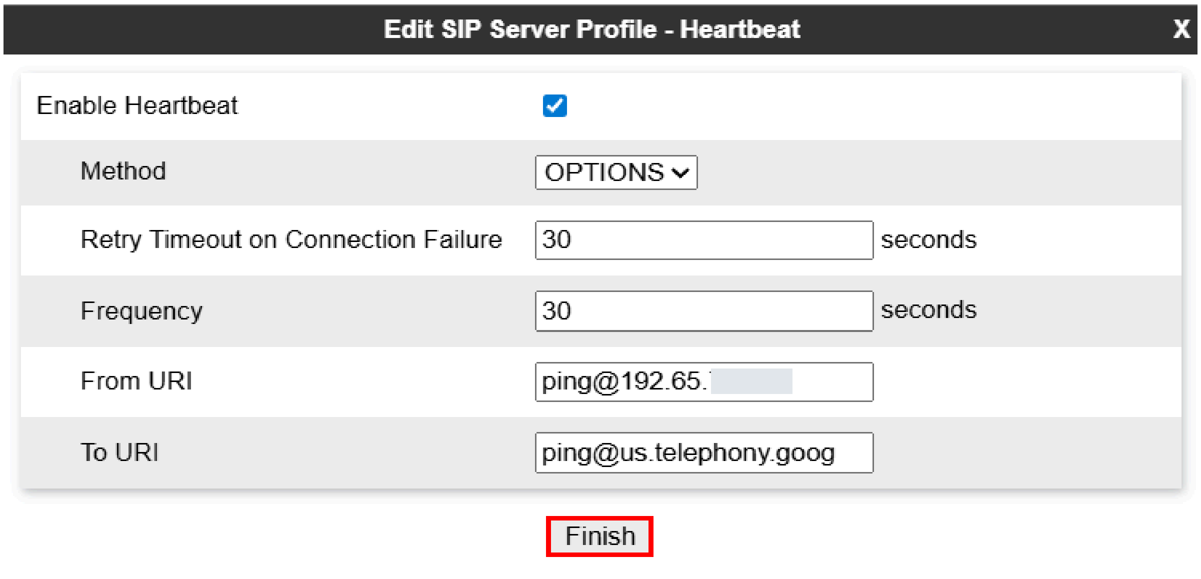

Figure 17: SIP Server for Google CES (Cont.)

- Navigate to Ping tab
- Set Enable Ping: **Checked**
- Set Ping interval: **60 seconds**
- Set Response Timeout: **30 seconds**
- Click **Finish**

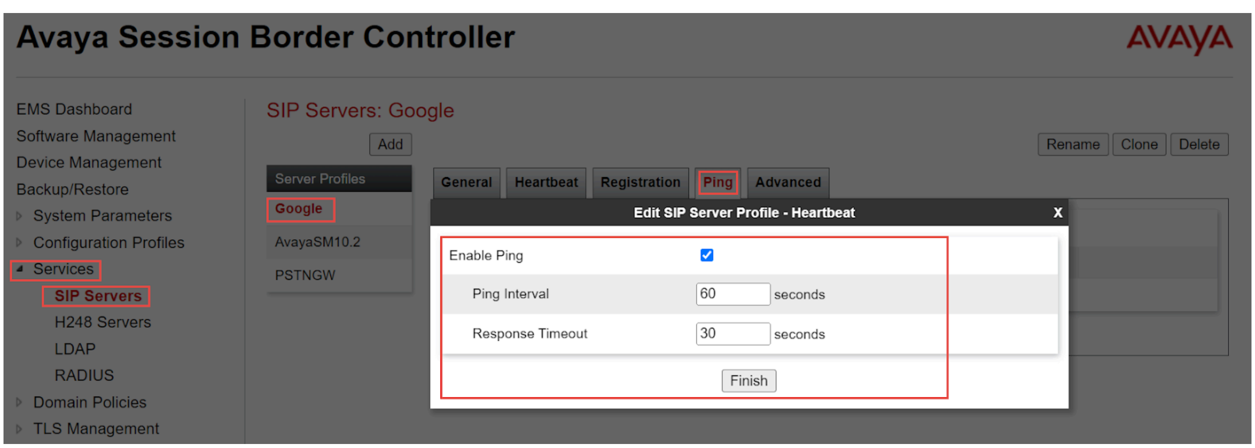

Figure 18: SIP Server for Google CES (Cont.)

- Navigate: **Advanced** tab
- Set Enable Grooming: **Checked**
- Set Interworking Profile: Select **Google.** Refer [Section 7.4.2]
- Click **Finish**

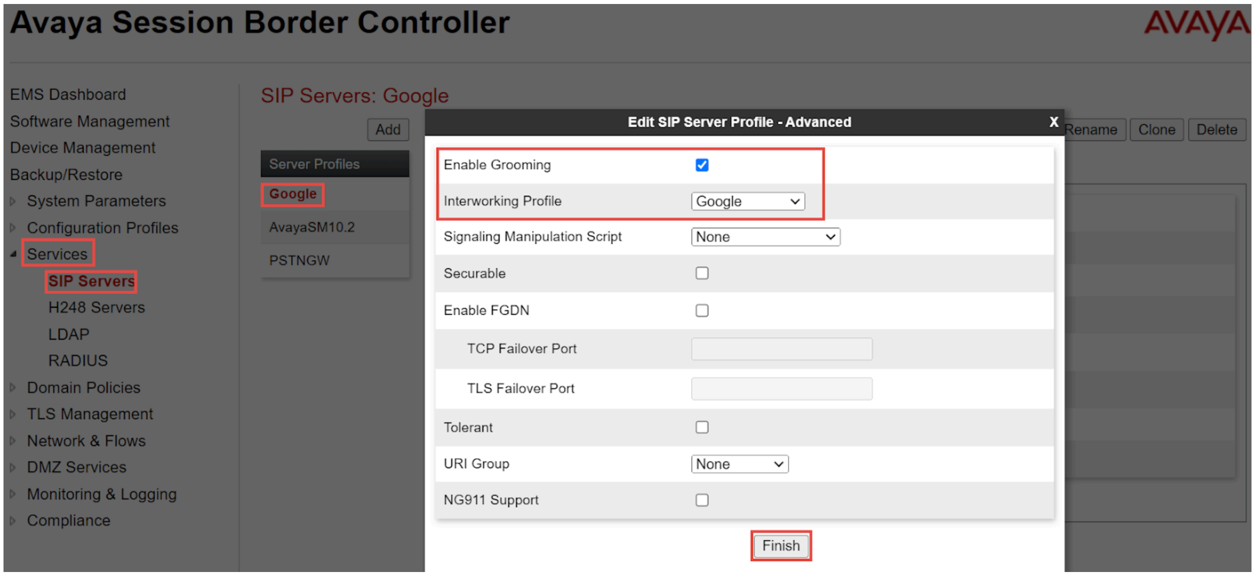

Figure 19: SIP Server for Google CES (Cont.)

SIP Server for **PSTN Gateway**

- Navigate: **Services** □ **SIP Servers**
- Click **Add**
- Set Profile Name: **PSTNGW**
- Click **Next**

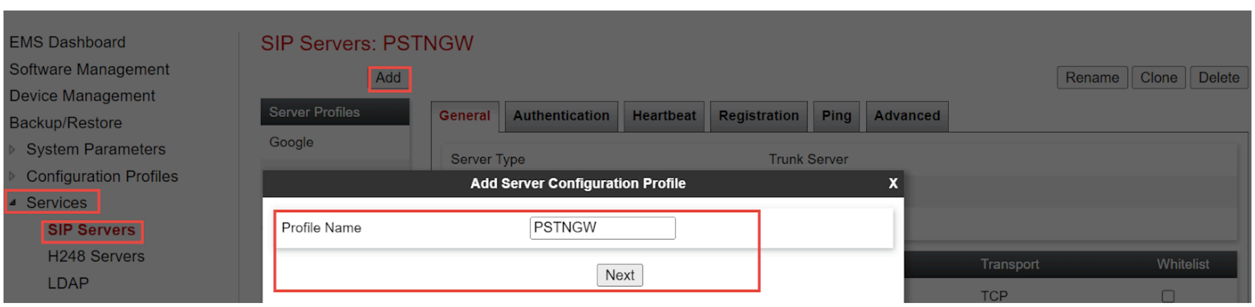

Figure 20: SIP Server for PSTN Gateway

- Set Server Type: Select **Trunk Server** from the drop down
- Set IP Address/FQDN/CIDR Range: **10.64.X.X**
- Set Port: **5060**
- Set Transport: **TCP**
- Click **Finish**

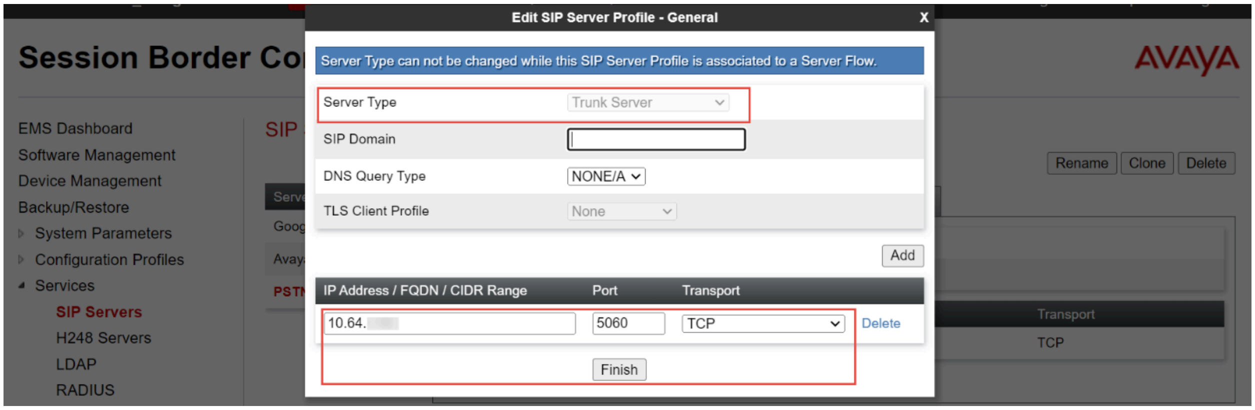

Figure 21: SIP Server for PSTN Gateway (Cont.)

- Navigate: **Heartbeat** tab
- Set Enable Heartbeat: **Checked**
- Set Method: **OPTIONS**
- Set Retry Timeout on Connection Failure: **30 seconds**
- Set Frequency: **30 seconds**
- Set From URI: **ping@10.80.X.X**
- Set To URI: **ping@10.64.X.X**
- Click **Finish**

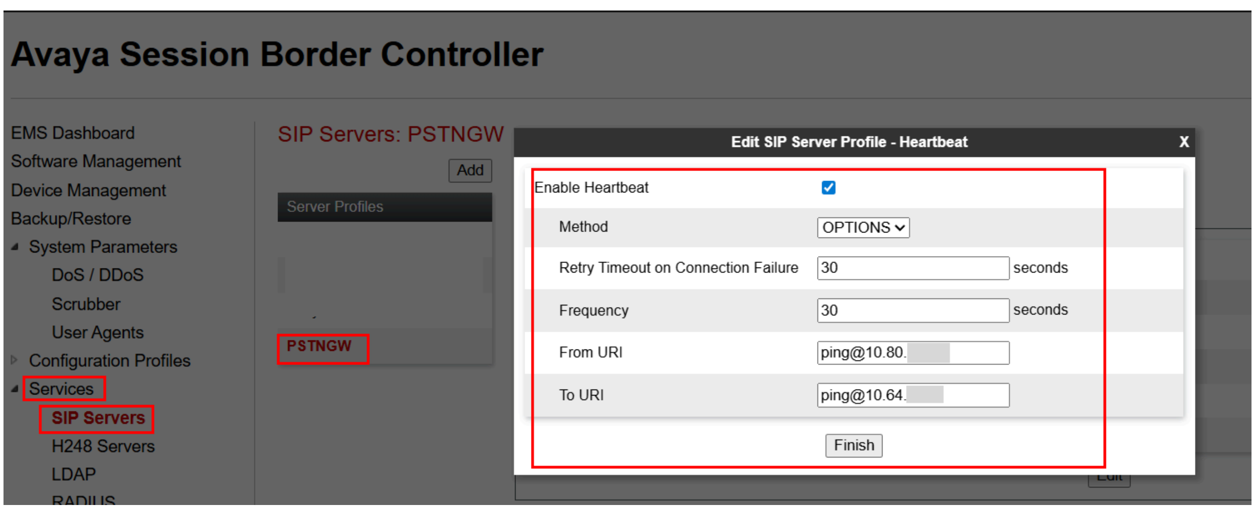

Figure 22: SIP Server for PSTN Gateway (Cont.)

- Navigate: **Ping** tab
- Set Enable Ping: **Checked**
- Set Ping interval: **60 seconds**
- Set Response Timeout: **30 seconds**
- Click **Finish**

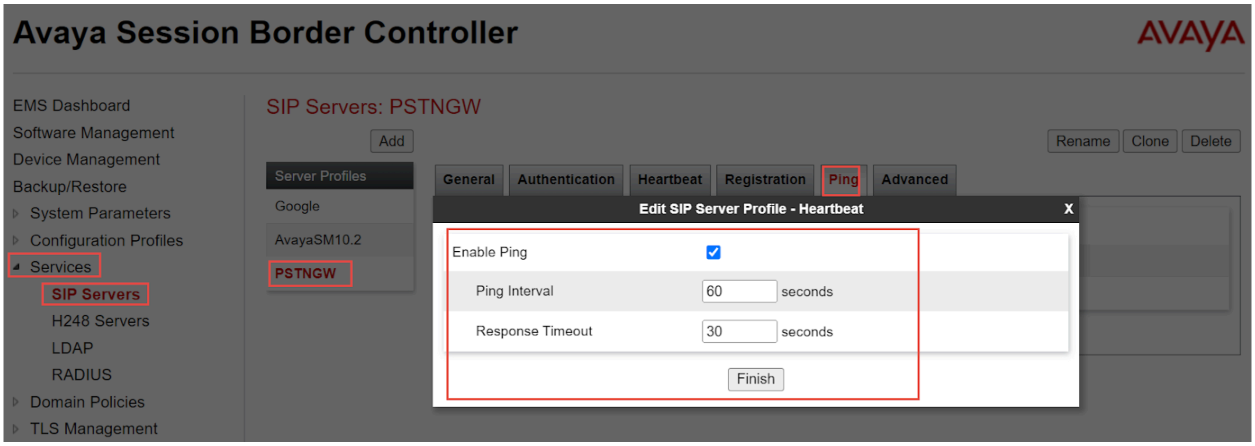

Figure 23: SIP Server for PSTN Gateway (Cont.)

- Navigate: **Advanced** tab
- Set Enable Grooming: **Checked**
- Set Interworking Profile: Select **PSTN.** Refer [Section 7.4.2]
- Click **Finish**

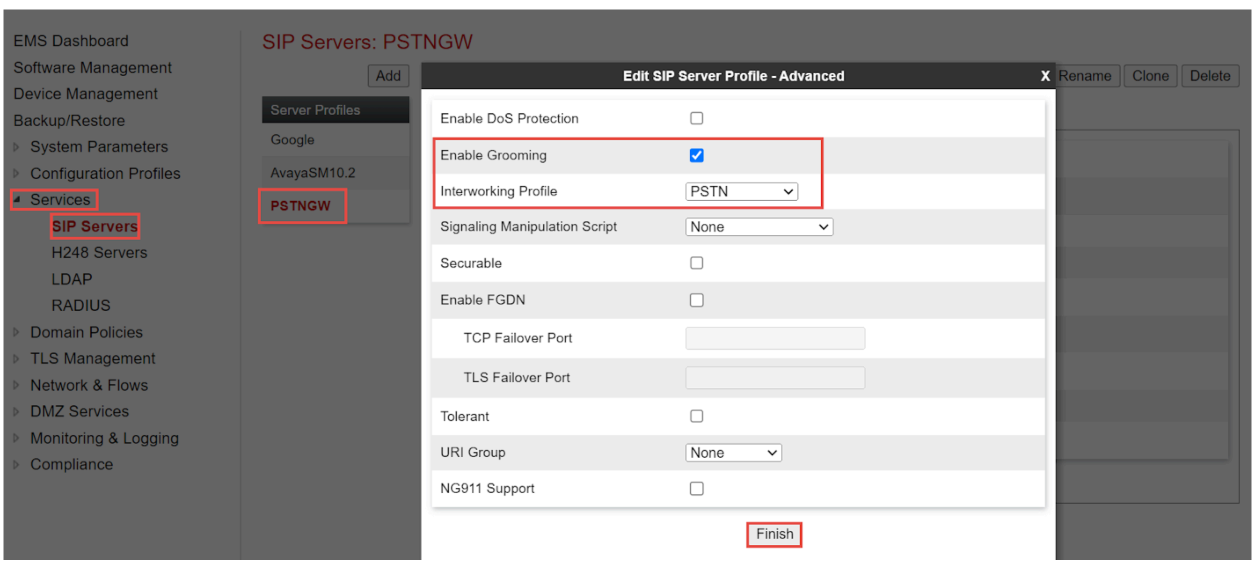

Figure 24: SIP Server for PSTN Gateway (Cont.)

7.4.4 Topology Hiding

Topology Hiding profile for **Google**

- Topology Hiding profiles are added for Google CES to overwrite and hide certain headers
- Navigate: **Configuration Profiles** □ **Topology Hiding**
- Click **Add**
- Set Profile Name: **Google CCAI**
- Click **Finish**

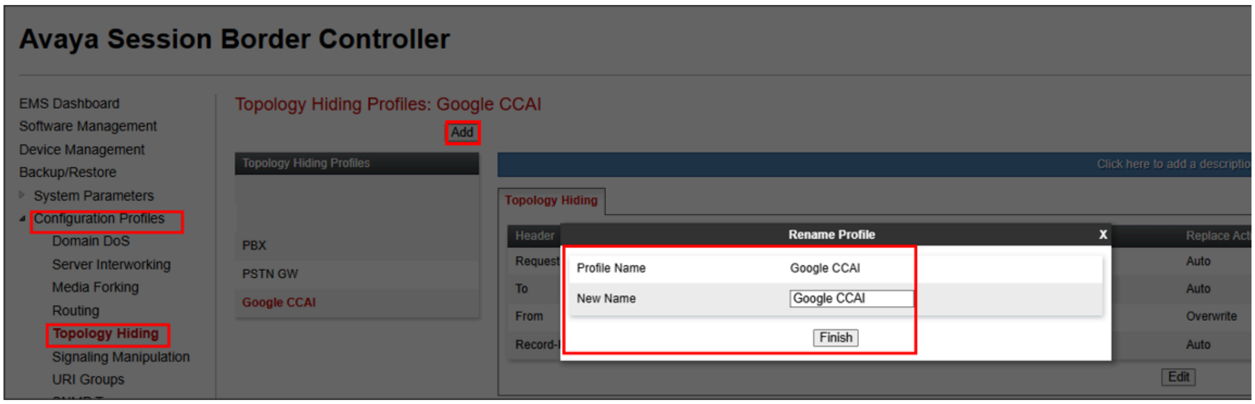

Figure 25: Topology Hiding for Google CES

- Select the newly created profile **Google CCAI** and Click **Edit**
- Overwrite Value: Replace the **From Header** with IP address **192.65.X.X**
- Click **Finish**

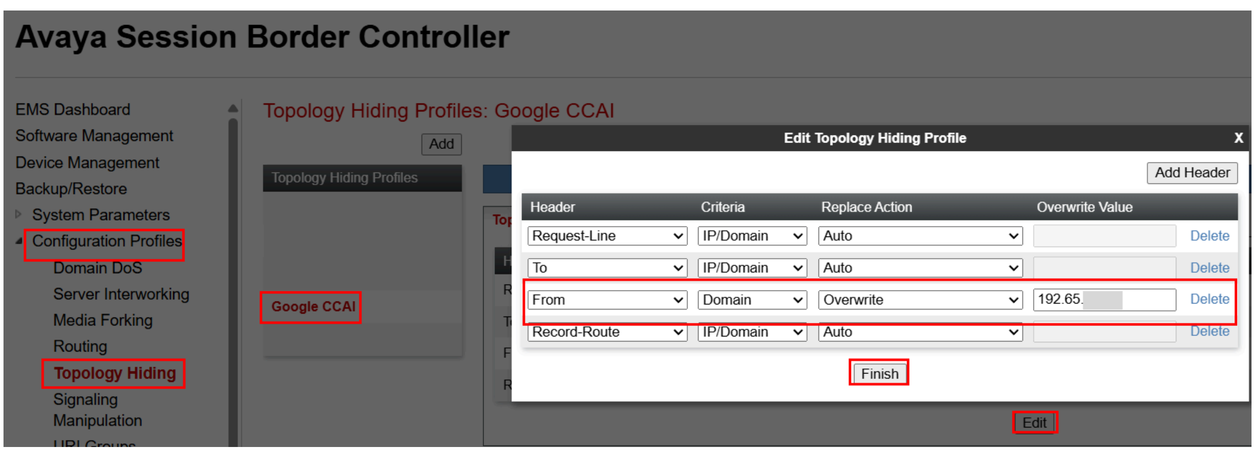

Figure 26: Topology Hiding for Google CES (Cont.)

Topology Hiding profile for **OnPrem PBX**

- Select the newly created profile **CUCM** and Click **Edit**
- Overwrite Value: Replace the **From Header** with **us.telephony.goog**
- Click **Finish**

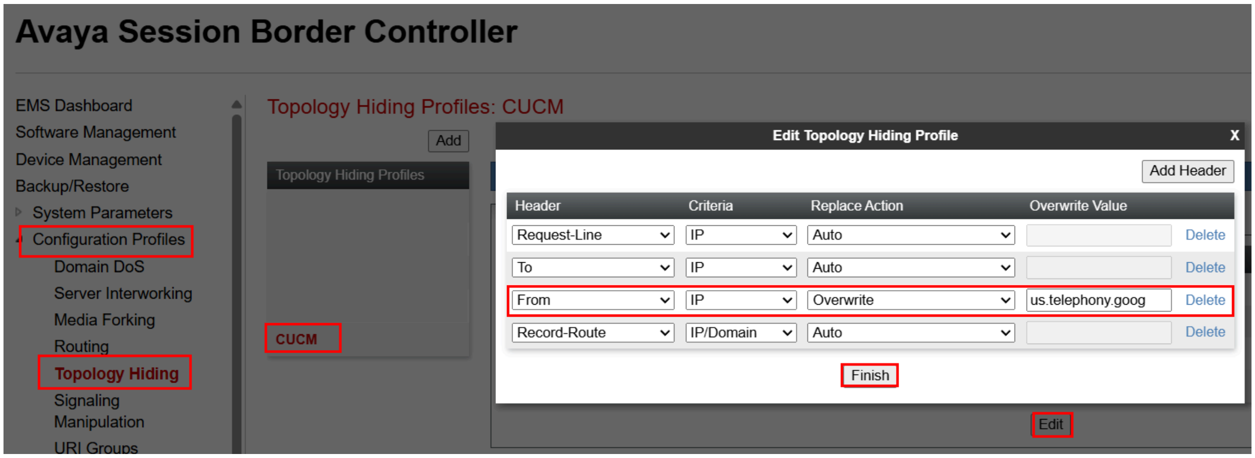

Figure 27: Topology Hiding for OnPrem PBX

Topology Hiding profile for **PSTN Gateway**

- Select the newly created profile **PSTN GW** and Click **Edit**
- Overwrite Value: Replace the **Request-Line Header** with **10.64.X.X**
- Overwrite Value: Replace the **To Header** with **10.64.X.X**
- Overwrite Value: Replace the **From Header** with **10.80.X.X**
- Click **Finish**

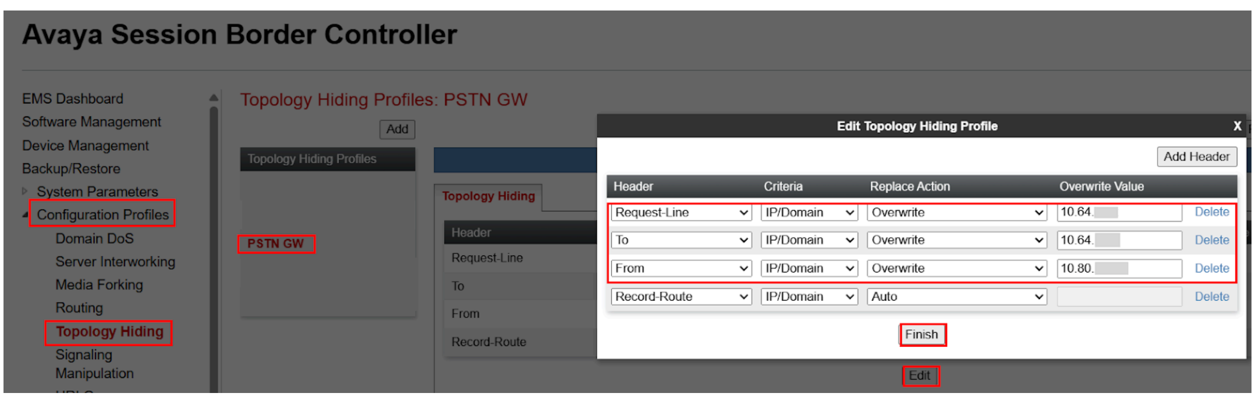

Figure 28: Topology Hiding for PSTN Gateway

7.4.5 Routing

Routing for **OnPrem PBX**

- Navigate: **Configuration Profiles** □ **Routing**
- Click **Add**
- Set Profile Name: **CUCM**
- Click **Next**

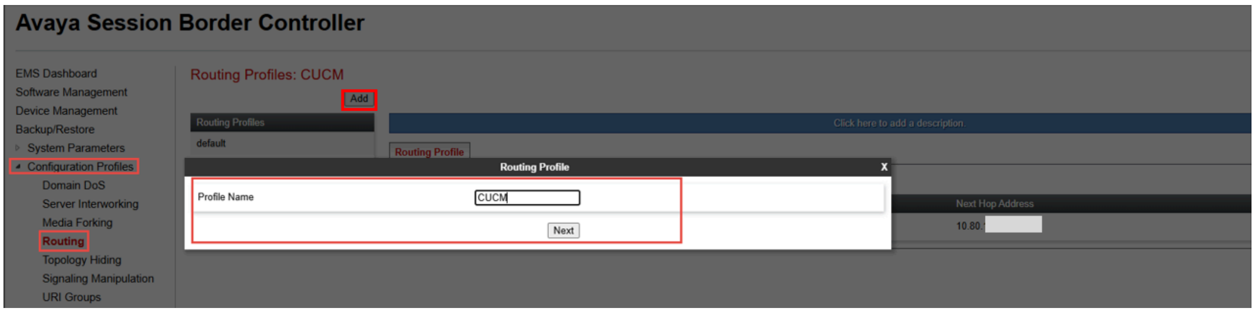

Figure 29: Routing for OnPrem PBX

- Select URI Group: *
- At Routing Profile Window, Click **Add**
- Set Priority/Weight: **1**
- Select SIP Server Profile: **CUCM.** Refer [Section 7.4.3](#)
- Next Hop Address: **10.80.X.X** from the drop-down menu
- Click **Finish**

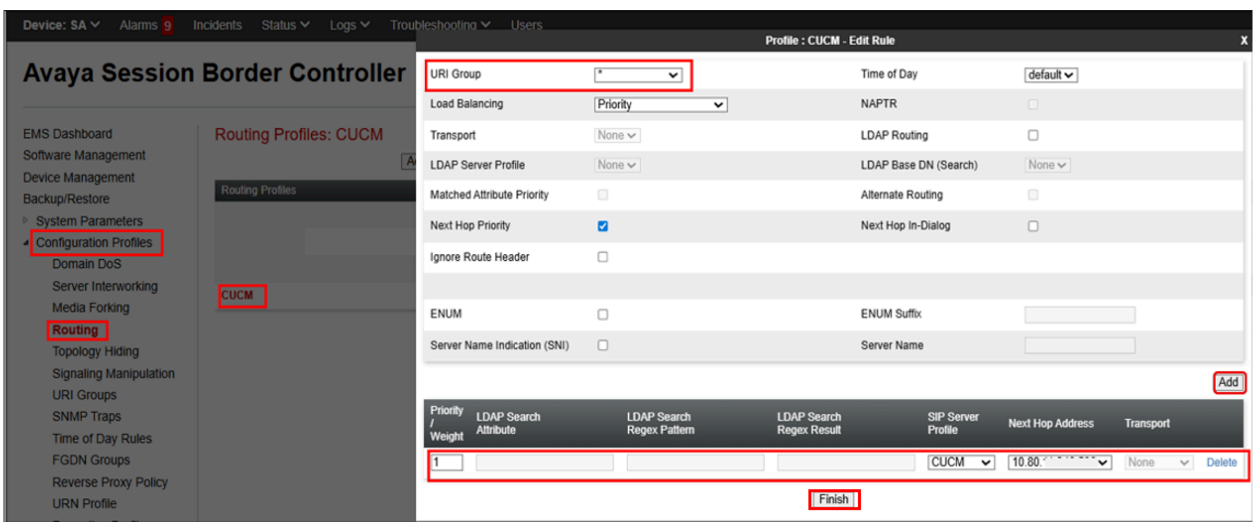

Figure 30: Routing for OnPrem PBX (Cont.)

Routing for **PSTN Gateway**

- Navigate: **Configuration Profiles** □ **Routing**
- Click **Add**
- Set Profile Name: **PSTNGW**
- Click **Next**

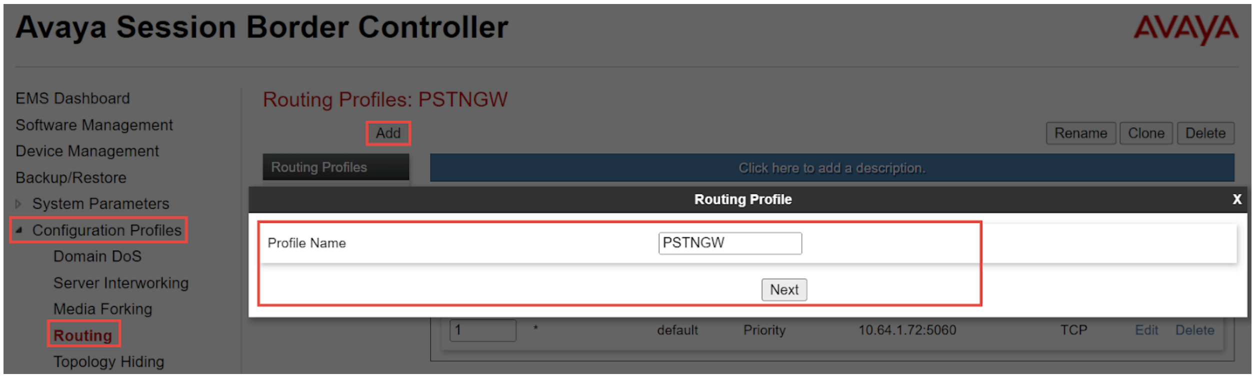

Figure 31: Routing for PSTN Gateway

- Select URI Group: *
- At Routing Profile Window, Click **Add**
- Set Priority/Weight: **1**
- Select SIP Server Profile: **PSTNGW.** Refer [Section 7.4.3]()
- Select Next Hop Address: **10.64.X.X**
- Click **Finish**

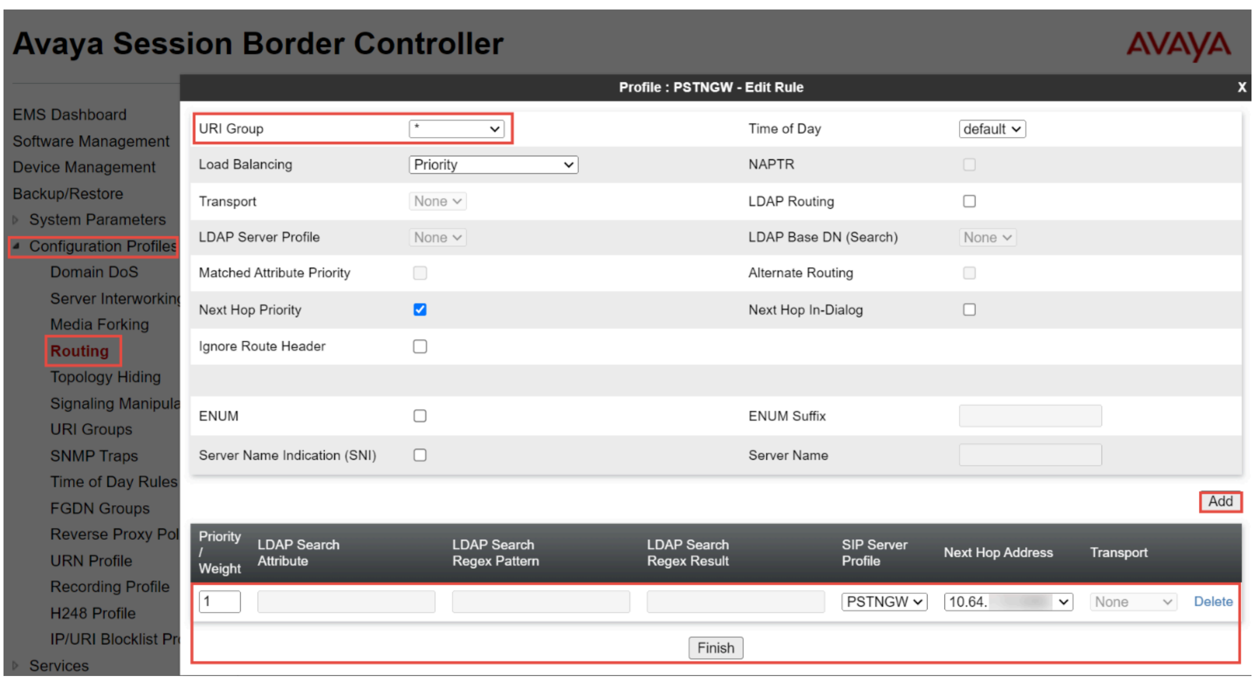

Figure 32: Routing for PSTN Gateway (Cont.)

Routing for **Google CES**

- Navigate: **Configuration Profiles** □ **Routing**
- Click **Add**
- Set Profile Name: **Google**
- Click **Next**

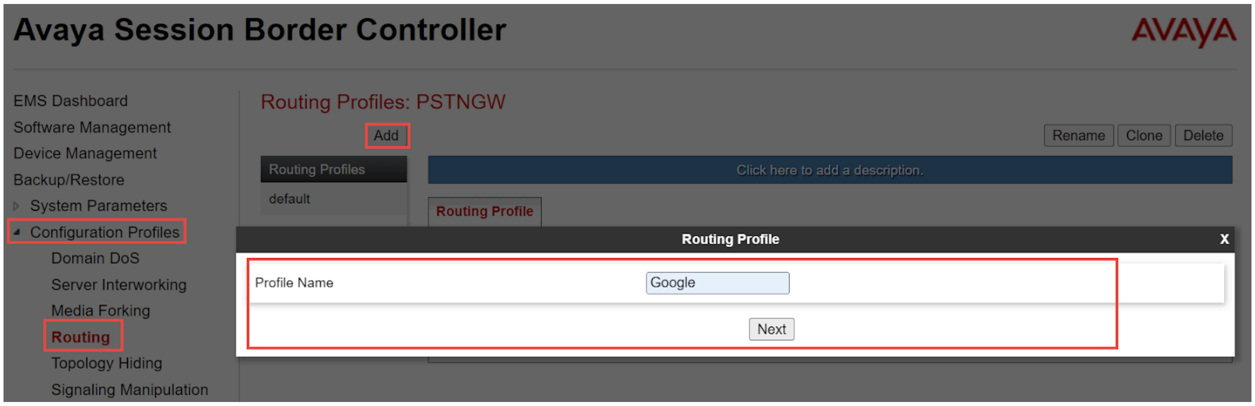

Figure 33: Routing for Google CES

- At Routing Profile Window, Click **Add**
- Set Priority/Weight: **1**
- Select SIP Server Profile: **Google.** Refer [Section 7.4.3]()
- Next Hop Address: **us.telephony.goog**
- Click **Finish**

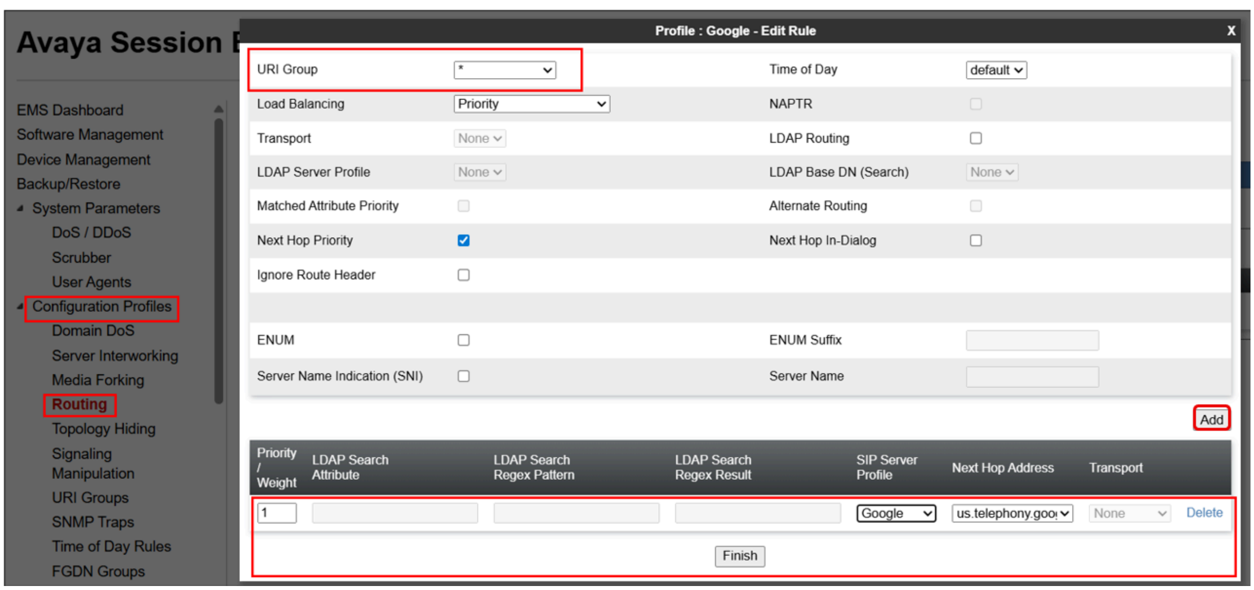

Figure 34: Routing for Google CES (Cont.)

7.4.6 Signaling Manipulation

Signaling Manipulation for Google CES

- Navigate: **Configuration Profiles** □ **Signaling Manipulation**
- Click **Add**

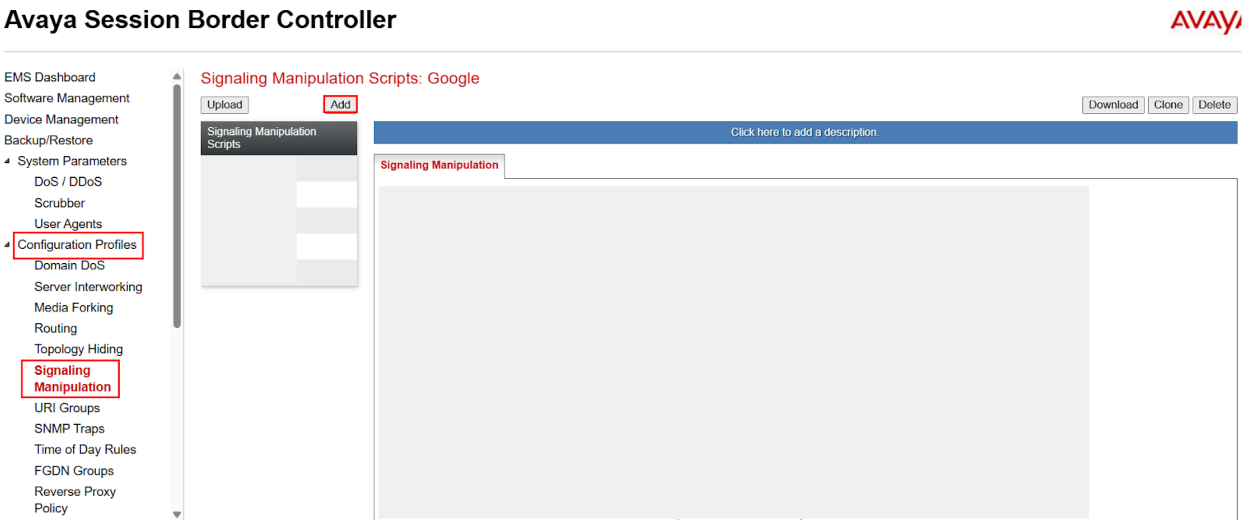

Figure 35: Signaling Manipulation

- Title: **Google CCAI_SR**
- Below sigma script is created to add **Call-Info** header towards Google CES with the Dialog Flow API request along with the Conversation ID.
- Regex **"&slash"** is appended to the Regex **%baseURI** as shown below. Subsequently, the "&slash" regex is replaced with the "/" symbol through string manipulation.
- Regex **%baseUri** value provided below is a reference value. Project name(**"ccai-38XXXXX/conversations"**) present in the Call-Info header will vary according to the project created by user. **Sr_** is an unique identifier and it matches the regex pattern requirement of call info header.
- Click **Save**

Figure 36: Signaling Manipulation Editor - Google CES

Sigma Script:

```
        within session "all"
        {
           act on request where %DIRECTION="OUTBOUND" and  %ENTRY_POINT="POST_ROUTING"
        and %METHOD="INVITE"
           {
              %aor = %HEADERS["Call-ID"][1];
              %baseUri =
        "<http:&slash/dialogflow.googleapis.com/v2beta1/projects/ccai-3898XX/conversations/Sr_";
              append( %baseUri, %aor);
               %newUri1 = ">;purpose=Goog-ContactCenter-Conversation";
        append( %baseUri, %newUri1);
              %HEADERS["Call-Info"][1] = %baseUri;
              %HEADERS["Call-Info"][1].URI.regex_replace("&slash","/");
              %HEADERS["Request_Line"][1].URI.USER.regex_replace("^.*", "+1314944XXXX");
              %HEADERS["TO"][1].URI.USER.regex_replace("^.*", "+1314944XXXX");
              %HEADERS["FROM"][1].URI.USER.regex_replace("^..........", "+1214550XXXX");
        %HEADERS["Allow"][1].regex_replace(", UPDATE,", "");
           }

}
```

Signaling Manipulation for OnPrem PBX

- Navigate: **Configuration Profiles Signaling Manipulation**
- Click **Add**

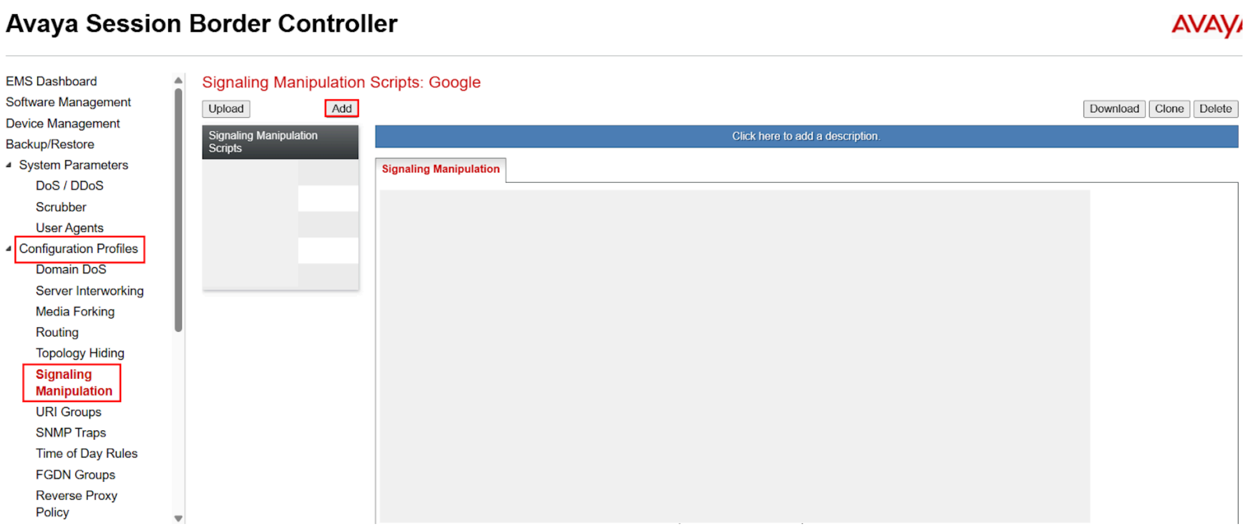

Figure 37: Signaling Manipulation for CUCM

- Title: **CUCM**
- Below script is created to manipulate the Request line, To and from headers received from Google CES during Agent handoff call.
- Click **Save**

Figure 38: Signaling Manipulation Editor for OnPrem PBX

Sigma Script:

```
within session "all"
{
   act on request where %DIRECTION="OUTBOUND" and  %ENTRY_POINT="POST_ROUTING" 
and %METHOD="INVITE"
   {

      %HEADERS["Request_Line"][1].URI.USER.regex_replace("^..........", "+1972852XXXX");
      %HEADERS["TO"][1].URI.USER.regex_replace("^..........", "+1972852XXXX");
      %HEADERS["FROM"][1].URI.USER.regex_replace("^..........", "+1214550XXXX");

   }

      }
```

7.4.7 Media Rules

- Configure Navigate: **Domain Policies** □ **Media Rules**
- Click **Add**
- Set Rule Name: **Google**
- Click **Next**

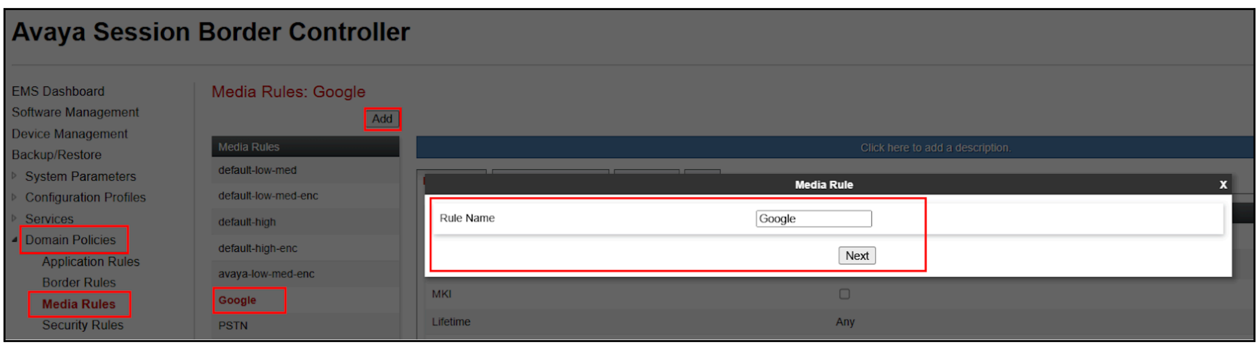

Figure 39: Media Rules

- Set Preferred Format #1: **SRTP_AES_CM_128_HMAC_SHA1_80**
- Click **Finish**

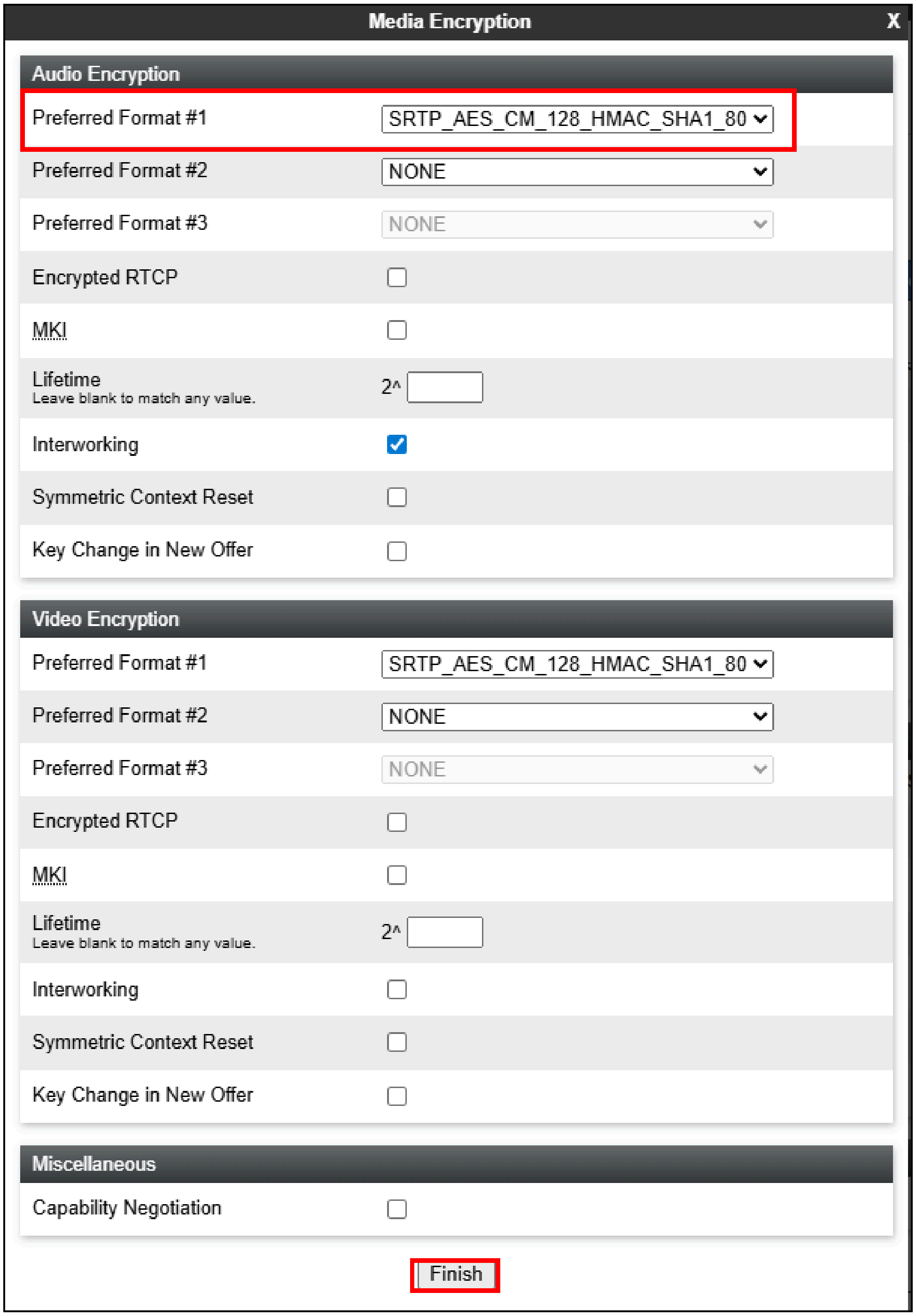

Figure 40: Media Rules (Cont.)

7.4.8 End Point Policy Groups

End Point Policy Group for **Google CES**

- Navigate: **Domain Policies** □ **End Point Policy Groups**
- Select **default-low** under Policy Groups
- Click **Clone**
- Set Group Name: **Google CCAI**
- Click **Finish**

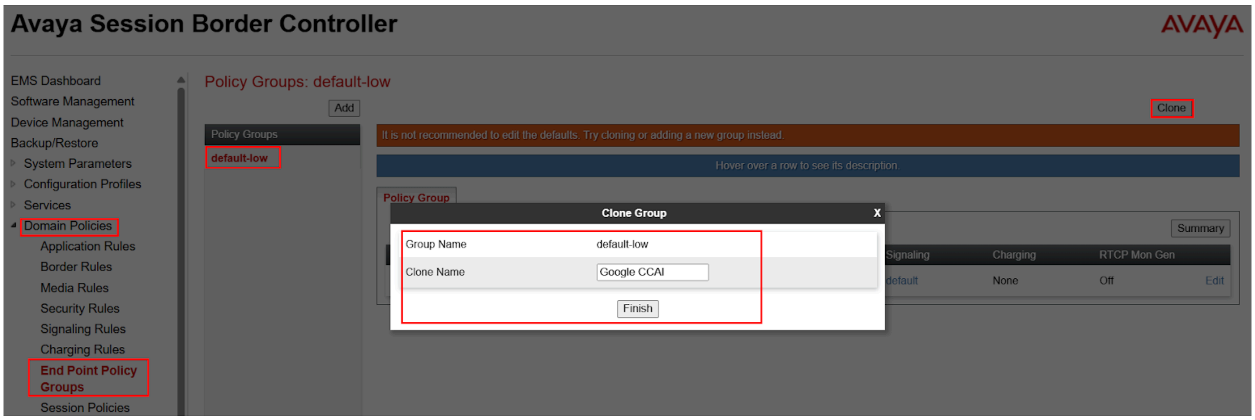

Figure 42: End Point Policy Group for Google CES

- Select Media Rule: **Google.** Refer [Section 7.4.7](#)
- Click **Finish**

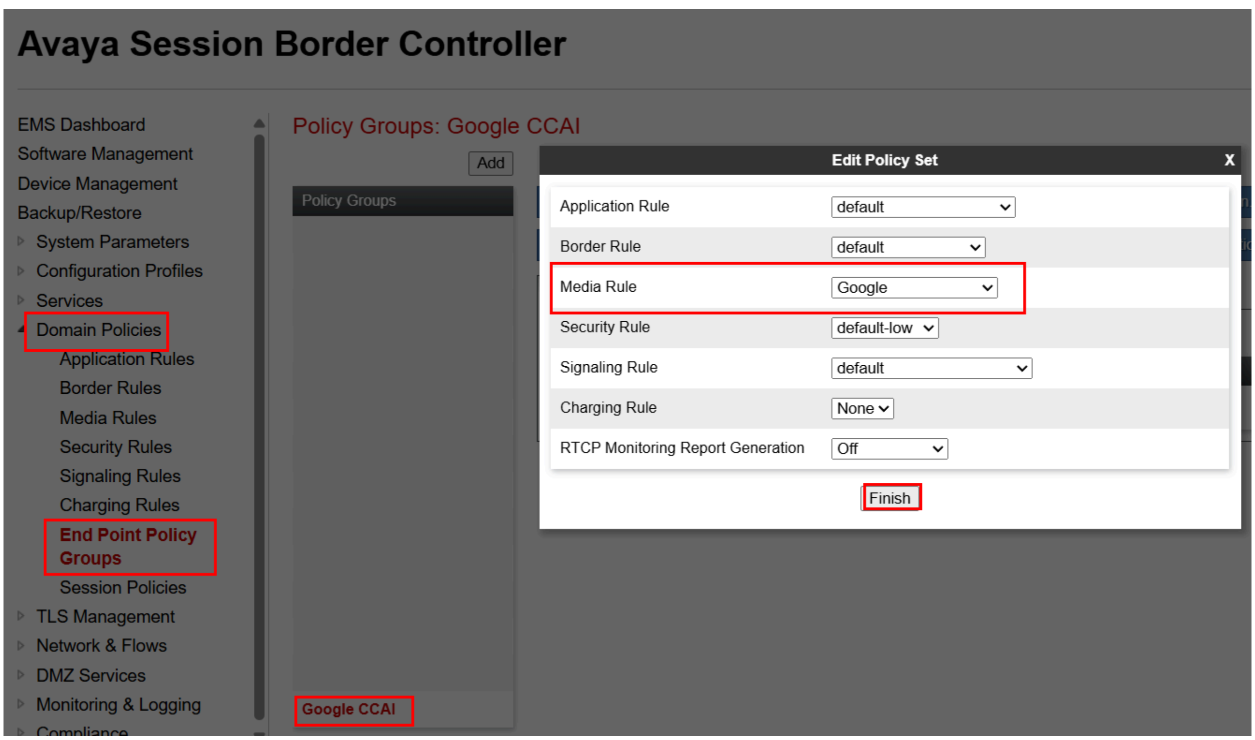

Figure 43: End Point Policy Group for Google CES

End Point Policy Group for **PSTN Gateway and OnPrem PBX**

- Navigate: **Domain Policies □ End Point Policy Groups**
- Select **default-low** under Policy Groups
- Click **Clone**
- Set Group Name: **PSTN_PBX**
- Click **Finish**

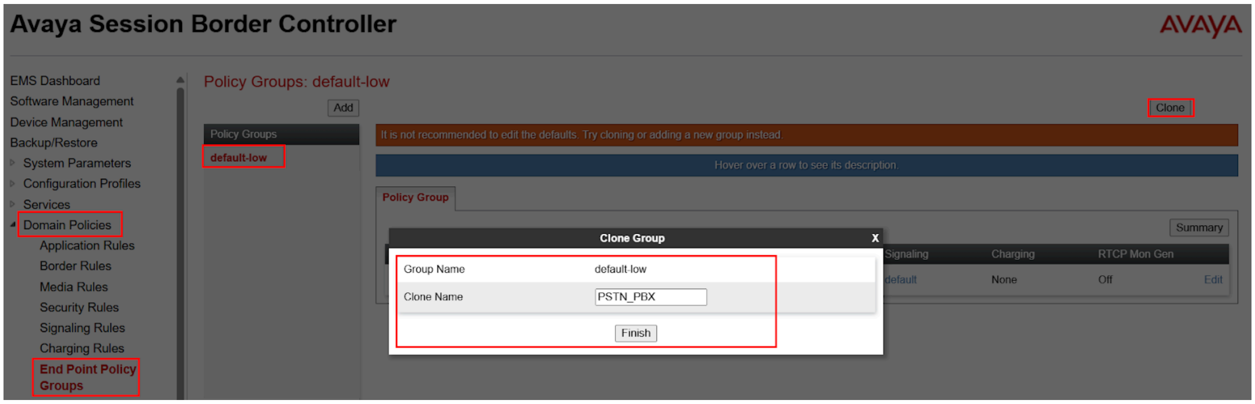

Figure 44: End Point Policy Group for PSTN Gateway and OnPrem PBX

- Click **Finish**

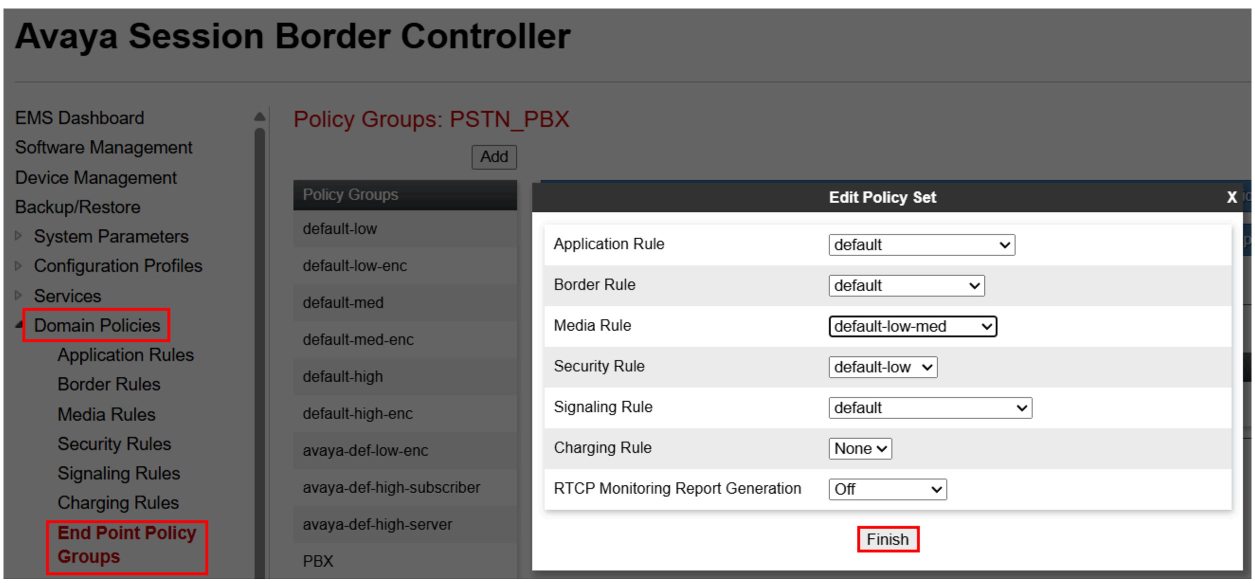

Figure 45: End Point Policy Group for PSTN Gateway and OnPrem PBX (Cont.)

7.4.9 Media Interface

- Navigate: **Network & Flows** □ **Media Interface**. Click **Add**
- Set Name: **Google_MI**
- Set IP Address: **Google (B1, VLAN 0)** from the drop down and the IP address populates automatically. The IP address for Interface facing Google CES is **192.65.X.X**
- Set Port Range: **35000-40000**
- Click **Finish**

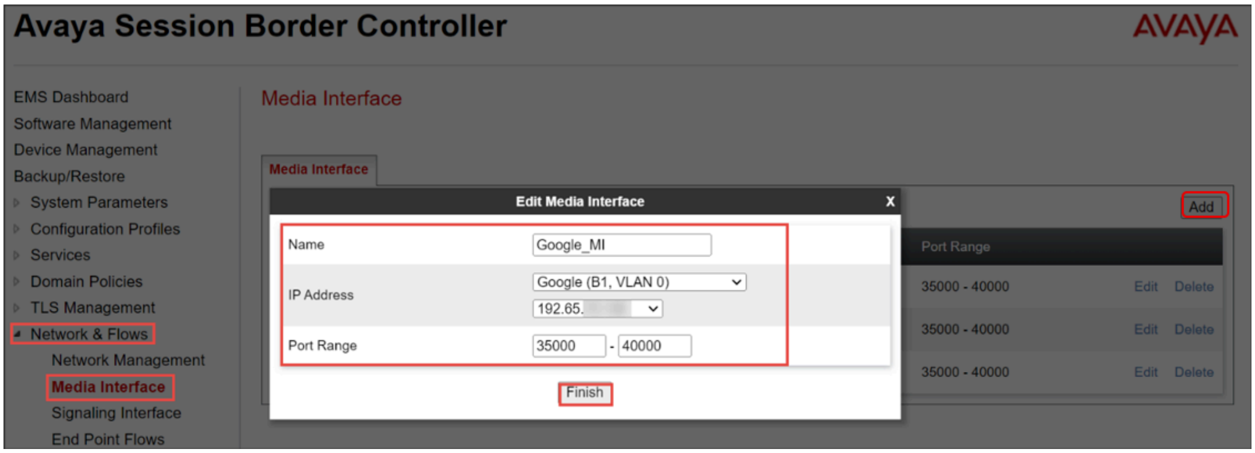

Figure 46: Media Interface Facing Google CES

- Set Name: **PSTN_PBX**
- Set IP Address: **PSTN PBX (B1, VLAN 0)**. from the drop down and the IP address populates automatically.
- The IP address for Interface facing PSTN Gateway and OnPrem PBX is **10.80.X.X**
- Set Port Range: **35000-40000**
- Click **Finish**

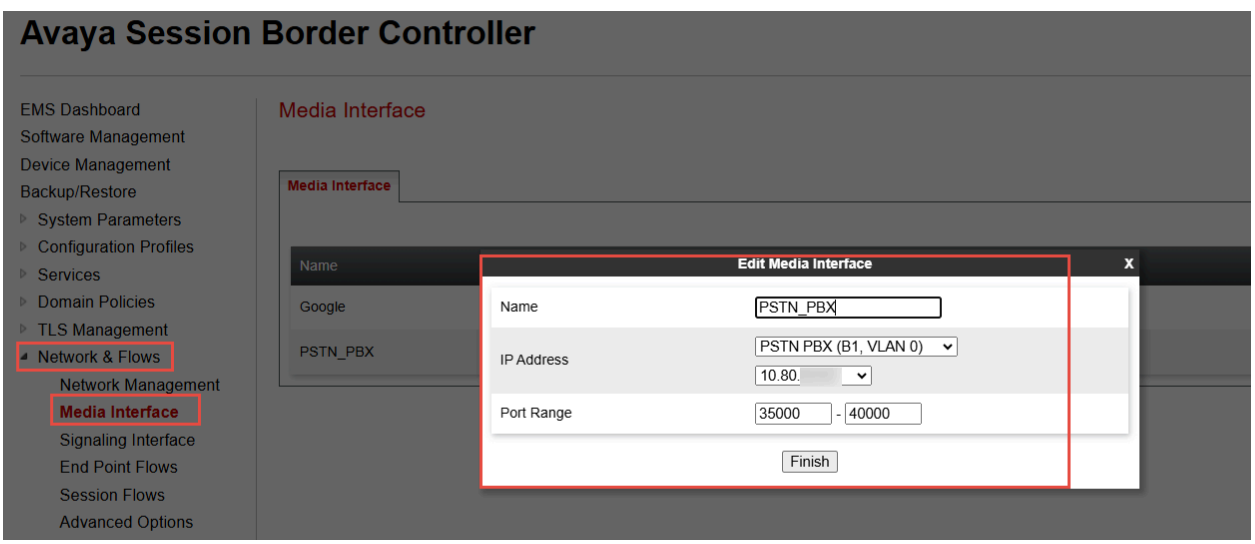

Figure 47: Media Interface Facing PSTN Gateway and OnPrem PBX

7.4.10 Network Management

Network Management for **Google CES**

- Navigate: **Network & Flows** **Network Management**. Click Add, new Add Network Interface window appears
- Set Name: **Google** is given for the network facing **Google**
- Set Default Gateway IP Address: **192.65.X.X**
- Set Network Prefix or Subnet Mask**: 255.255.X.X**
- Set Interface: **B1**
- Set IP Address: **192.65.X.X**
- Click **Finish**

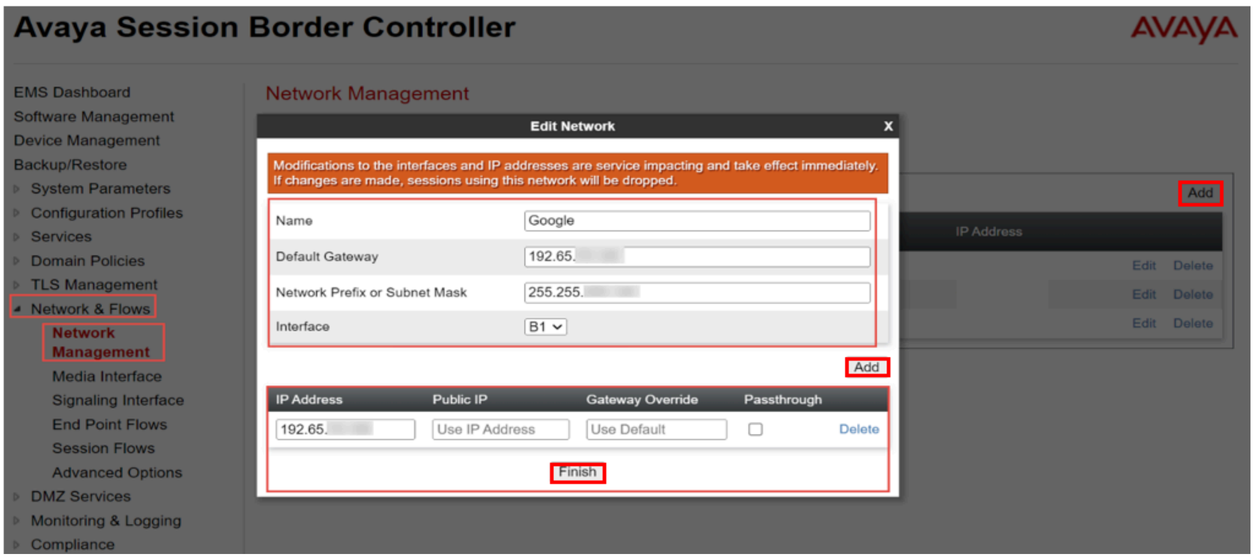

Figure 48: Network Management Facing Google CES

Network Interface for **PSTN Gateway and OnPrem PBX**

- Set Name: **Google** is given for the network facing **PSTN PBX**
- Set Default Gateway IP Address: **10.80.X.X**
- Set Network Prefix or Subnet Mask: **255.255.255.0**
- Set Interface: **B1**
- Set IP Address: **10.80.X.X**
- Click **Finish**

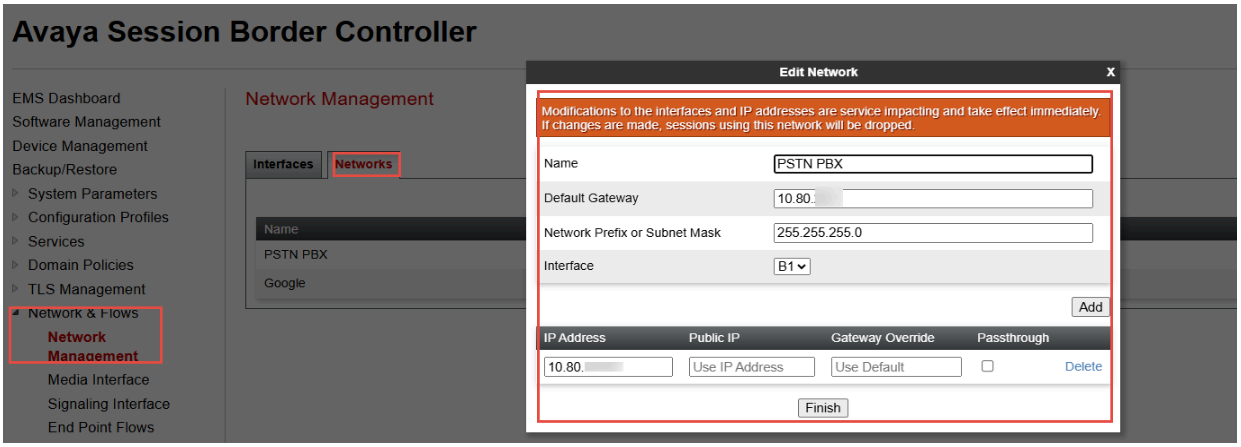

Figure 49: Network Management Facing PSTN Gateway and OnPrem PBX

7.4.11 Signaling Interface

Signaling Interface for **Google**

- Navigate to: **Network & Flows** □ **Signaling Interface**. Click **Add**, new Add Signaling Interface window appears
- Set Name: **Google_SI** is given for the interface facing **Google**
- Set IP Address: **Google (B1, VLAN 0), 192.65.X.X**
- Set TLS Port: **5061**
- Select TLS Profile: **Google.** Refer Section 7.4.13
- Click **Finish**

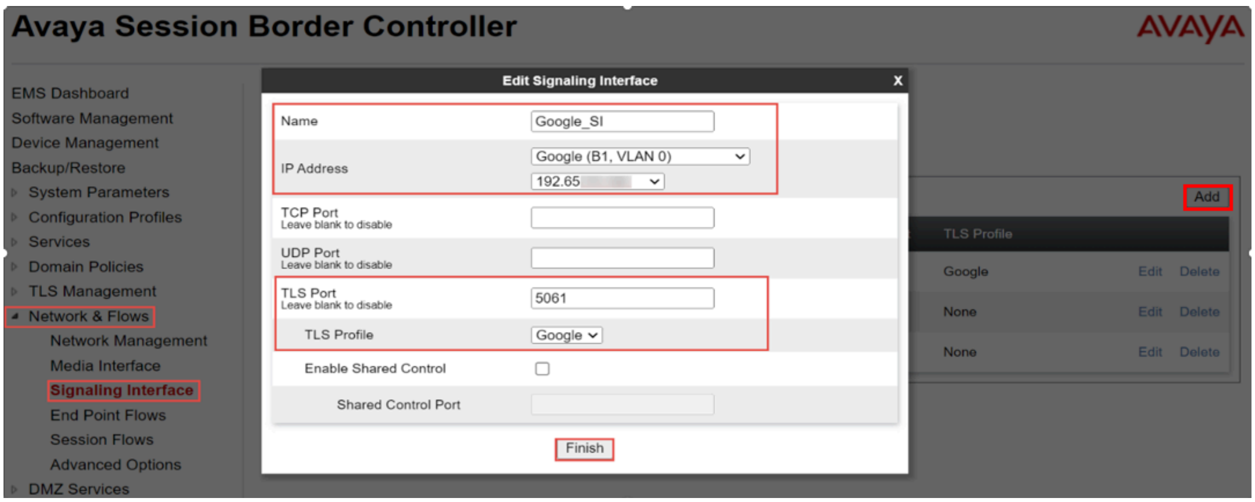

Figure 50: Signaling Interface Facing Google CES

Signaling Interface for **PSTN Gateway and OnPrem PBX**

- Set Name: **PSTN PBX**
- Set IP Address: **PSTN PBX (B1, VLAN 0), 10.80.X.X**
- Set TCP Port: **5060**
- Click **Finish**

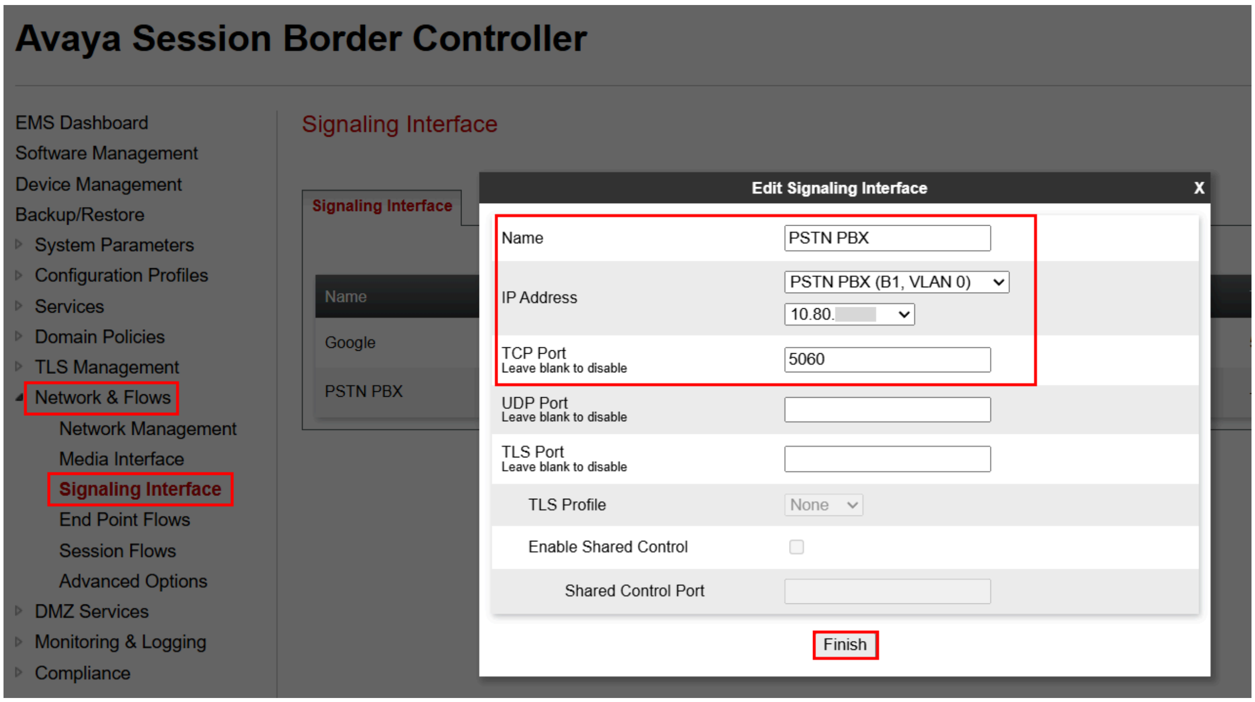

Figure 51: Signaling Interface Facing PSTN Gateway and OnPrem PBX

7.4.12 End Point Flow

End Point Flow for **OnPrem PBX, Google CES and PSTN Gateway**

- Navigate: **Network & Flows** □ **End Point Flows**□ **Server Flows** Click **Add**

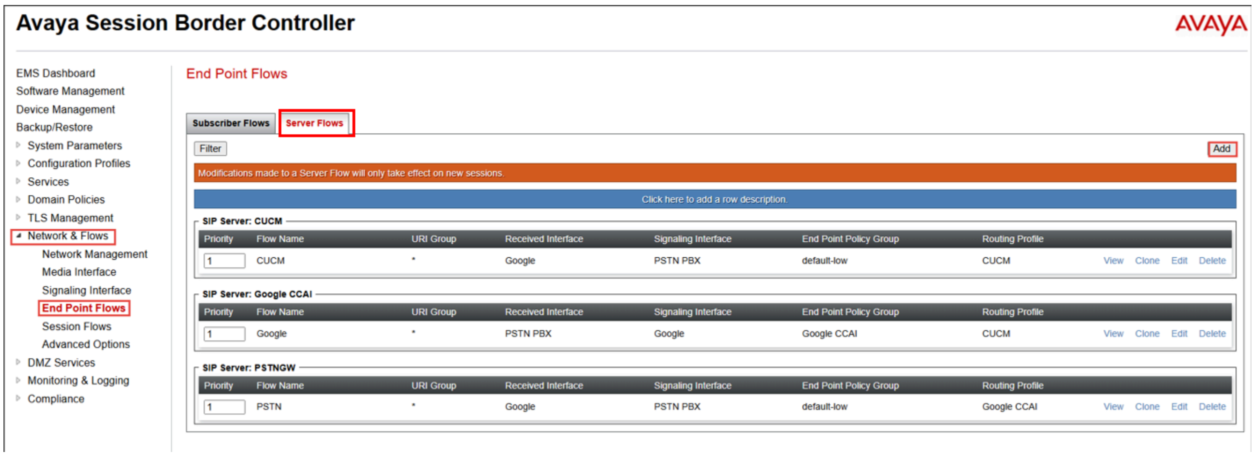

Figure 52: Server Flows

- Set SIP Server: **PSTNGW**
- Set Flow Name: **PSTN**
- Select SIP Server profile: **PSTNGW**
- Select URI Group: *
- Select Received Interface: **Google**
- Select Signaling Interface: **PSTN PBX**
- Select Media Interface: **PSTN_PBX**
- Select End Policy Group: **default-low**
- Select Routing Profile: **Google CCAI**
- Select Topology Hiding Profile: **PSTN GW**
- Click **Finish**

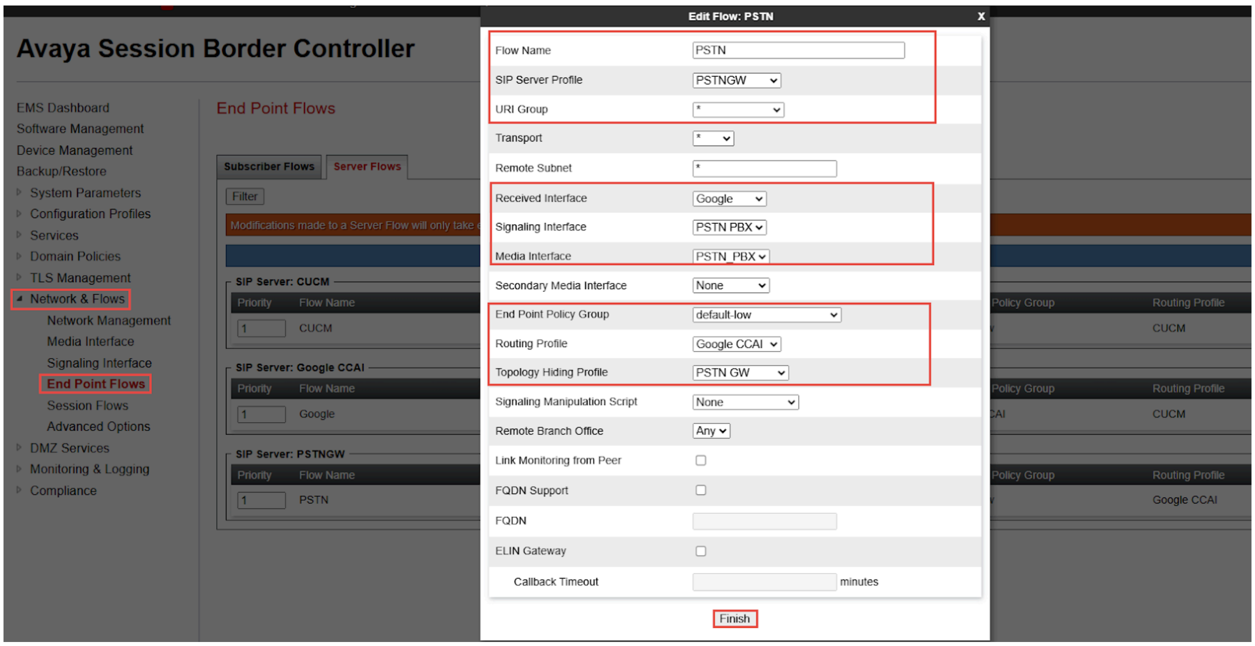

Figure 53: Server Flow for PSTN Gateway

End point flow for **Google CES**

- Navigate: **Network & Flows** □ **End Point Flows** □ **Server Flows**
- Set SIP Server: **Google CCAI**

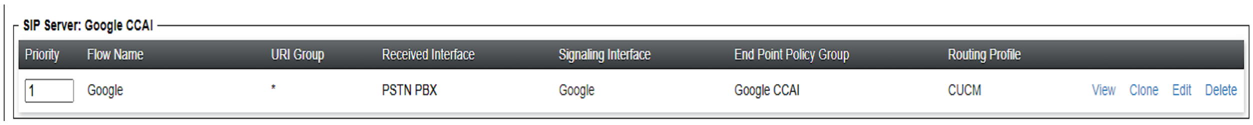

Figure 54: Server Flow for Google CES

- Set Flow Name: **Google**
- Select SIP Server profile: **Google CCAI**
- Select URI Group: *
- Select Received Interface: **PSTN PBX**
- Select Signaling Interface: **Google**
- Select Media Interface: **Google**
- Select End Policy Group: **Google CCAI**
- Select Routing Profile: **CUCM**
- Select Topology Hiding Profile: **Google CCAI**
- Signaling Manipulation Script: **Google CCAI_SR**
- Click **Finish**

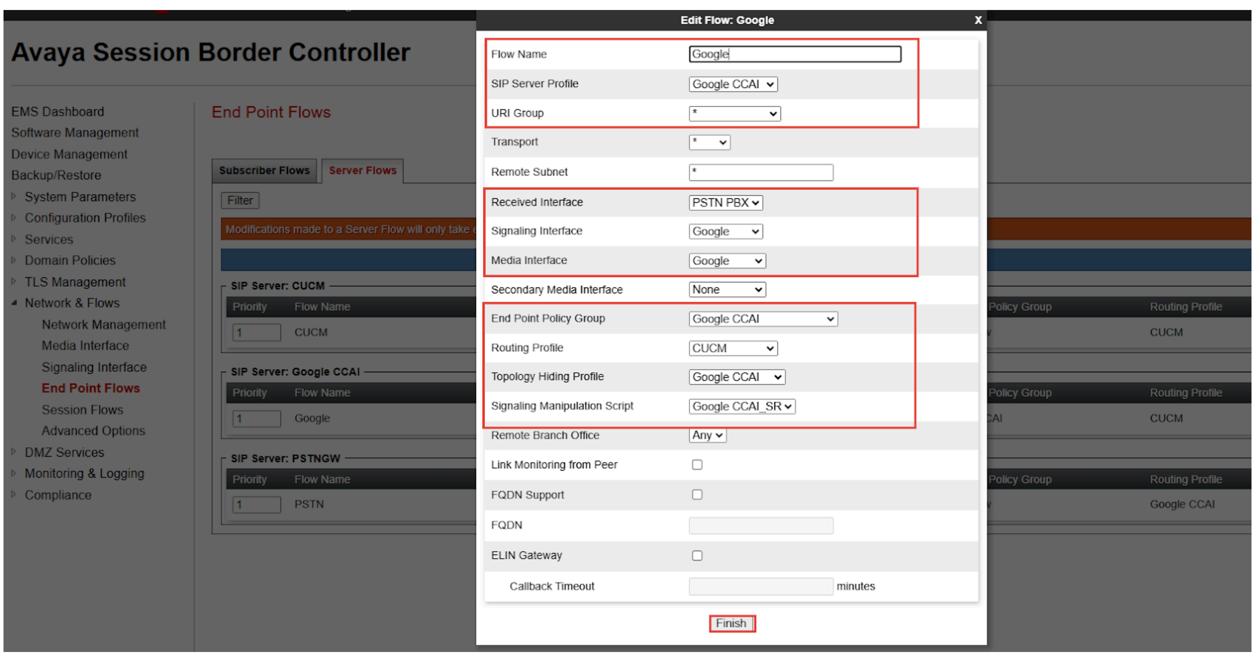

Figure 55: Server Flow for Google CES (Cont.)

End point flow for **OnPrem PBX**

- Navigate: **Network & Flows** □ **End Point Flows** □ **Server Flows**
- Set SIP Server: **CUCM**

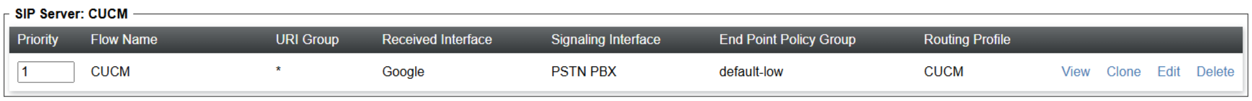

Figure 56: Server Flow for OnPrem PBX

- Set Flow Name: **CUCM**
- Select SIP Server profile: **CUCM**
- Select URI Group: *
- Select Received Interface: **Google**
- Select Signaling Interface: **PSTN PBX**
- Select Media Interface: **Google**
- Select End Policy Group: **default-low**
- Select Routing Profile: **CUCM**
- Select Topology Hiding Profile: **CUCM**
- Signaling Manipulation Script: **CUCM**
- Click **Finish**

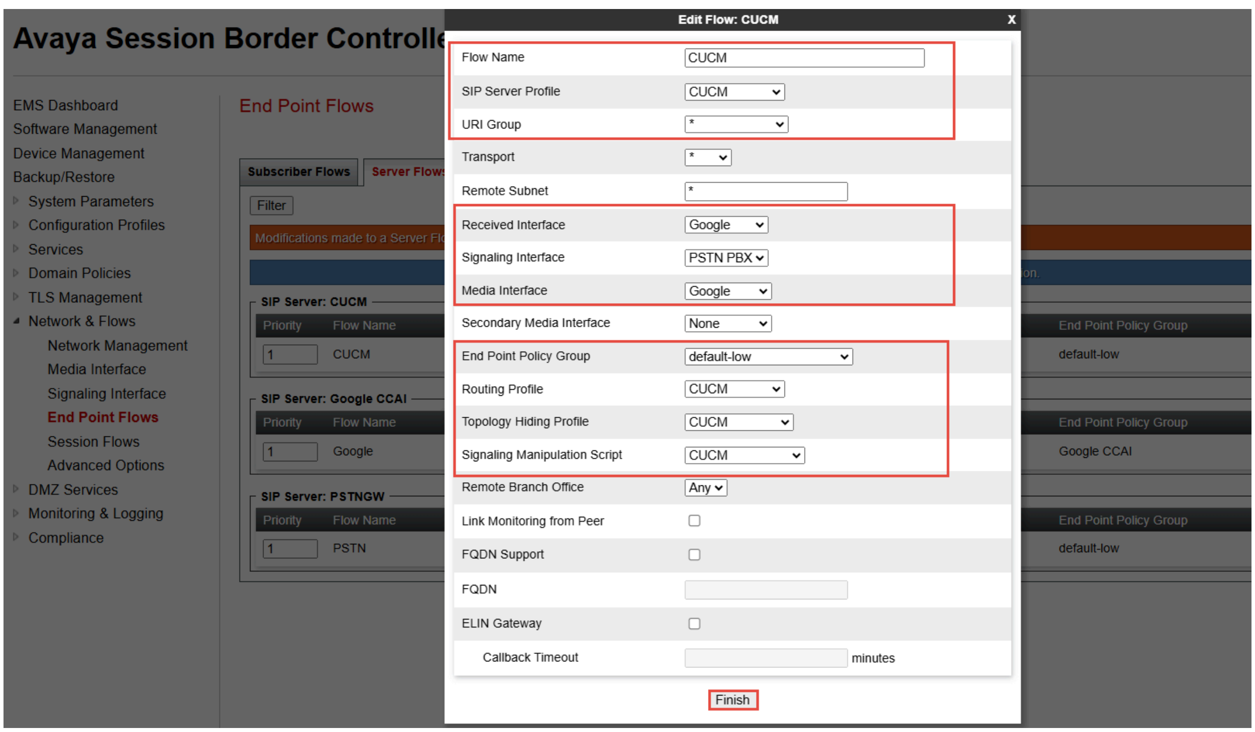

Figure 57: Server Flow for OnPrem PBX (Cont.)

7.4.13 TLS Configuration

Configure TLS management for Google CES

- Navigate: **TLS management** □ **Certificates**. Click **Generate CSR**

Figure 58: Generate CSR

- Set Country Name: **US**
- Set State/Province Name: **Texas**
- Set Locality Name: **Plano**
- Set Organization name: **Tekvizion**
- Set Organizational Unit: **lab**
- Set Common Name: **sbc8.tekvizionlabs.com**
- Set Algorithm: **SHA256**
- Select Key Size (Modulus Length): **2048 bits**

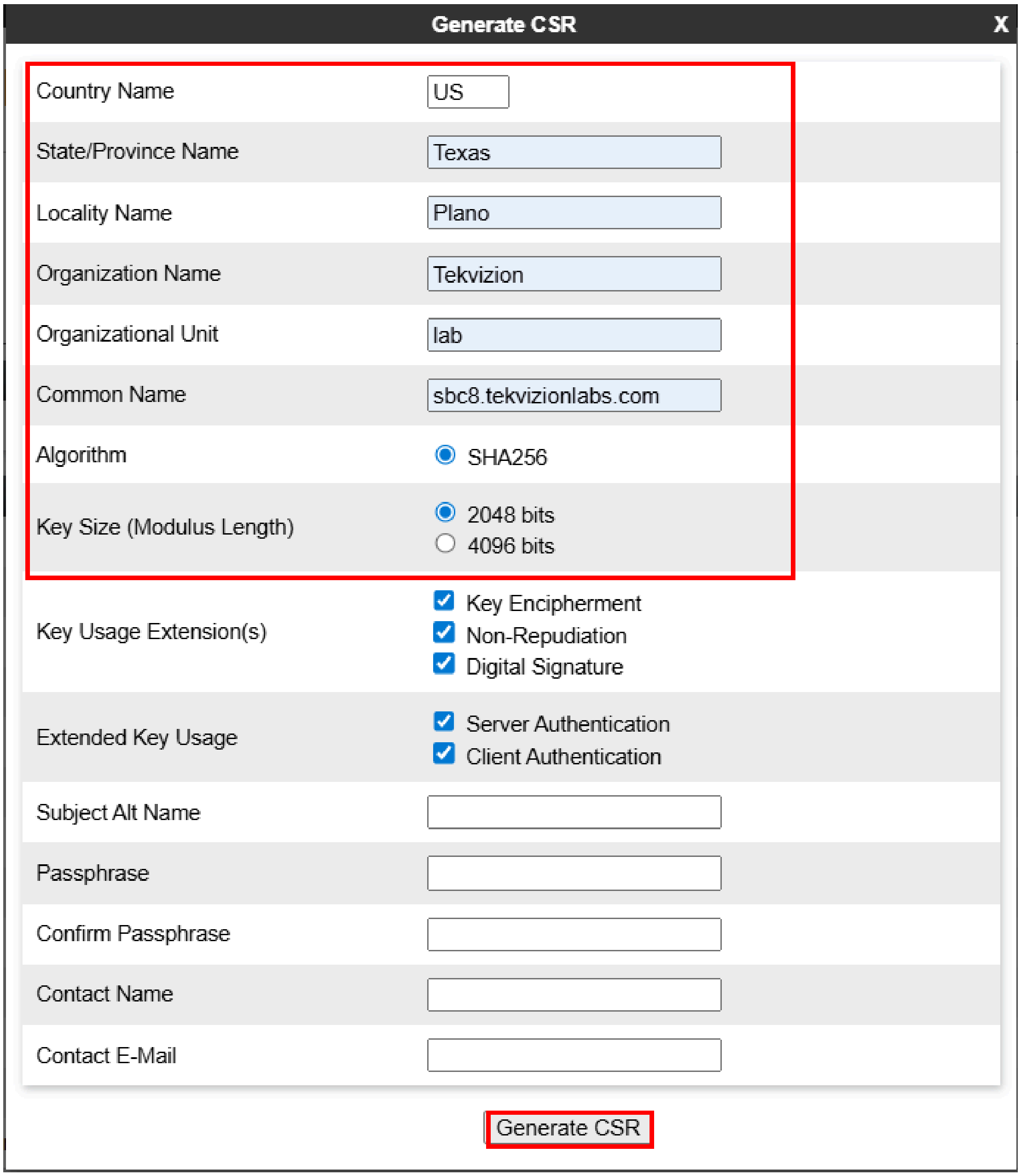

Figure 59: Generate CSR (Cont.)

Upload Google Certificate:

Download the Google Root Certificates from the following link https://pki.goog/repository/ and select the label GTS Root R1 only

- Navigate: **TLS management** □ **Certificates**. Click **Install**

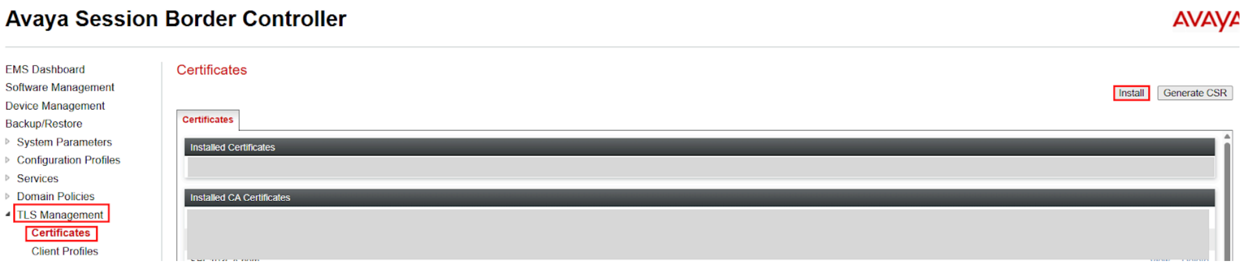

Figure 60: Certificate installation

- Set Type: Select **CA Certificate**
- Set Name: **GTS Root R1**
- Set Allow Weak Certificate/Key: **Checked**
- Set Certificate File: Click Choose File to select **GTS Root R1.pem**
- Click **Upload**

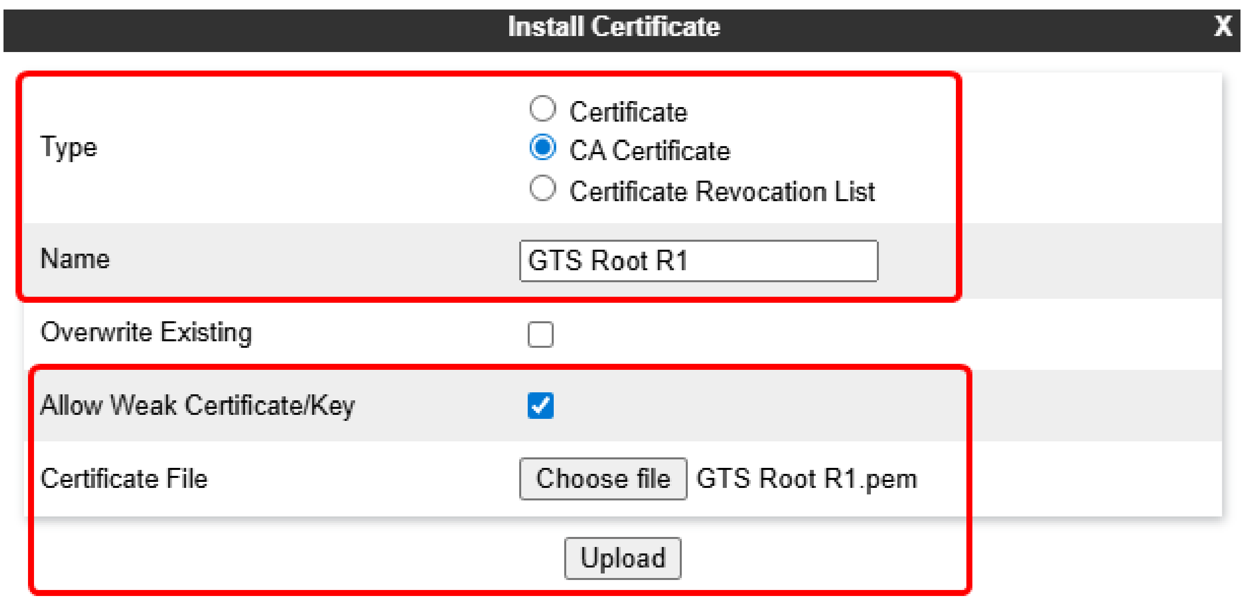

Figure 61: GTS Root R1

Upload SBC intermediate certificates:

- Set Type: **CA Certificate**
- Set Name: **GoDaddy_Root**
- Set Allow Weak Certificate/Key: **Checked**
- Set Certificate File: Click Choose File to select **Go_Daddy_Root.cer**
- Click **Upload**

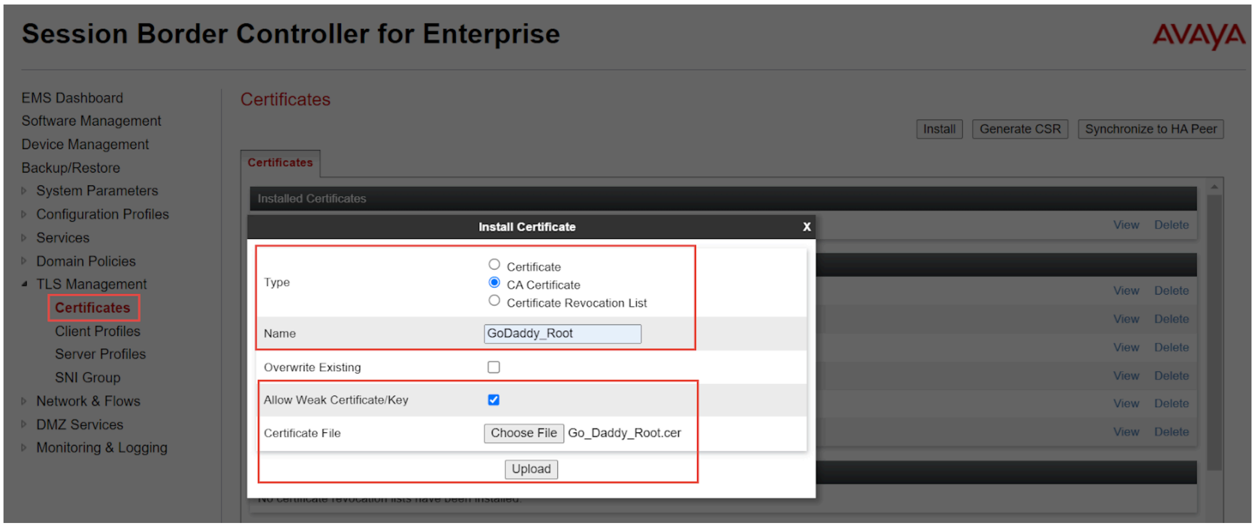

Figure 62: Upload GoDaddy Root CA

- Set Type: **CA Certificate**
- Set Name: **Go_Daddy_Secure**
- Set Allow Weak Certificate/Key: **Checked**
- Set Certificate File: Click Choose File to select **Go_Daddy_Secure.cer**
- Click **Upload**

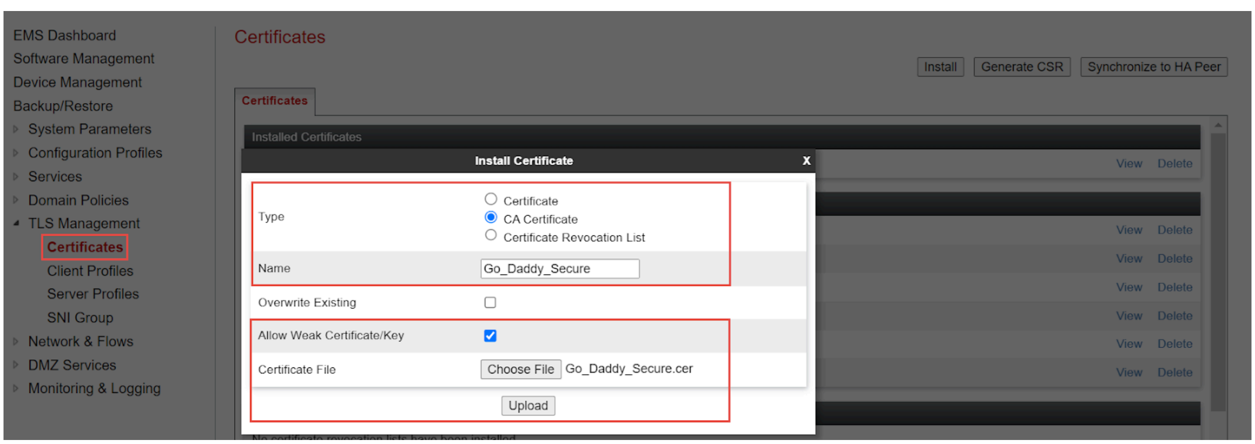

Figure 63: Upload GoDaddy Secure CA

- Set Type: **Certificate**
- Set Name: **sbc8**
- Set Allow Weak Certificate/Key: **Checked**
- Set Certificate File: Click Choose File to select **23xxxx.pem**
- Select Key: **Use Existing Key**
- Select Key File: **sbc8.key** from drop down
- Click **Upload**

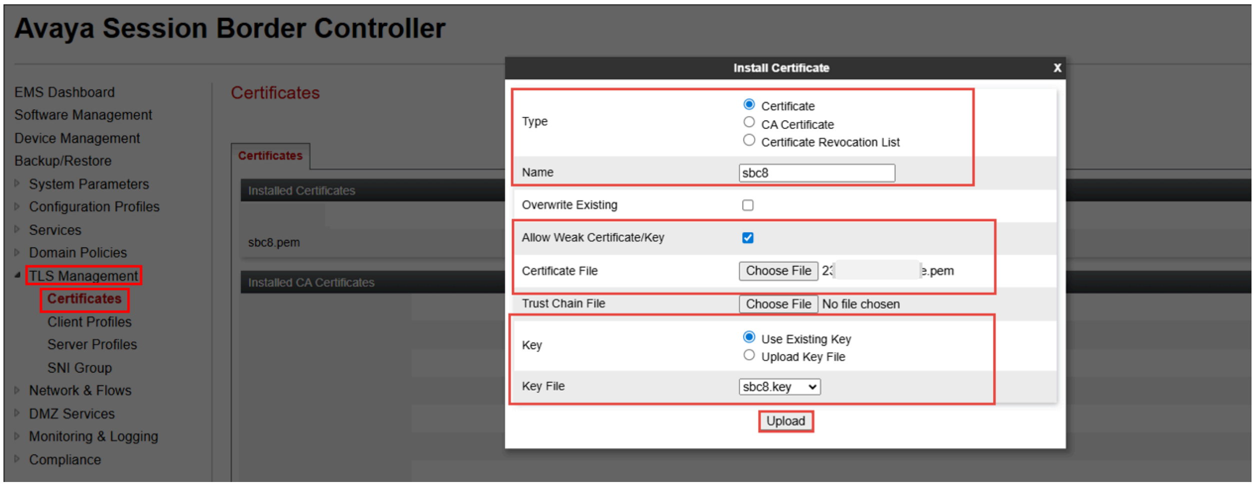

Figure 64: Upload Avaya SBC server Certificate

Client Profile for **Google CES**

- Navigate: **TLS management** □ **Client Profiles**. Click **Add**
- Set Profile Name: **Google**
- Set Certificate: select server certificate **sbc8.pem**
- Set Peer Certificate Authorities: Select **GTSRoot1.pem**
- Set Verification Depth: **5**
- Click **Next**

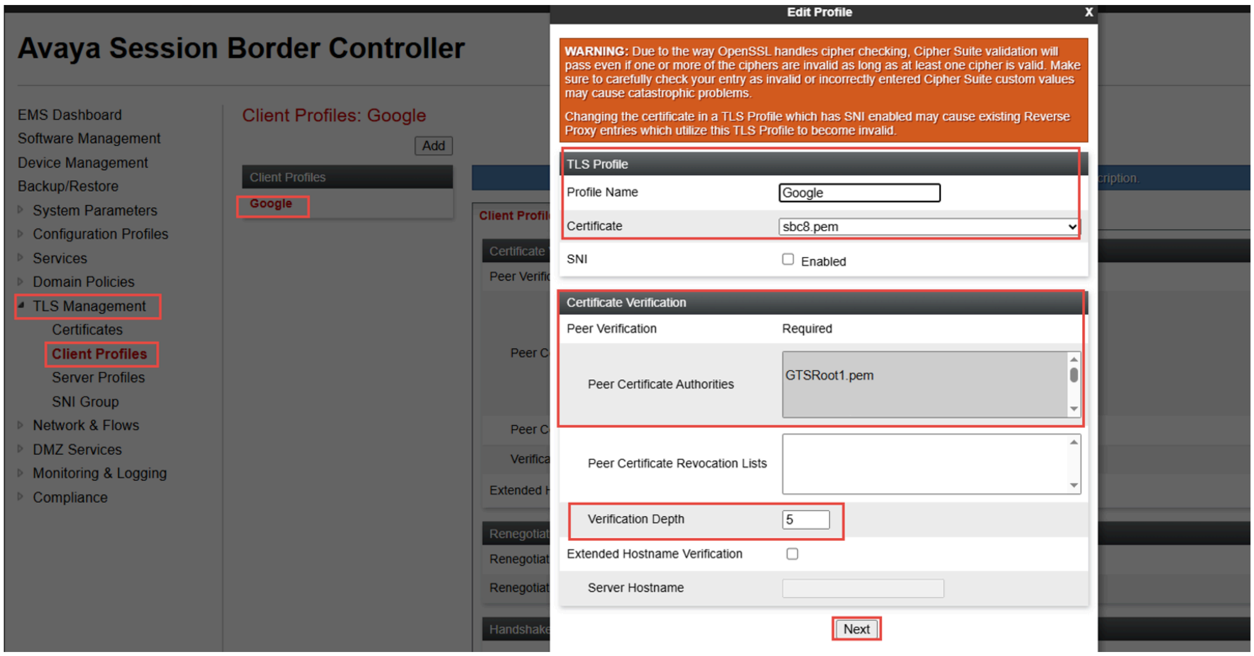

Figure 65: Client Profile Google CES

Server Profile for **Google CES**

- Navigate: **TLS management** □ **Server Profiles**. Click Add
- Set Profile Name: **Google**
- Set Certificate: **sbc8.pem**
- Click on **Next**

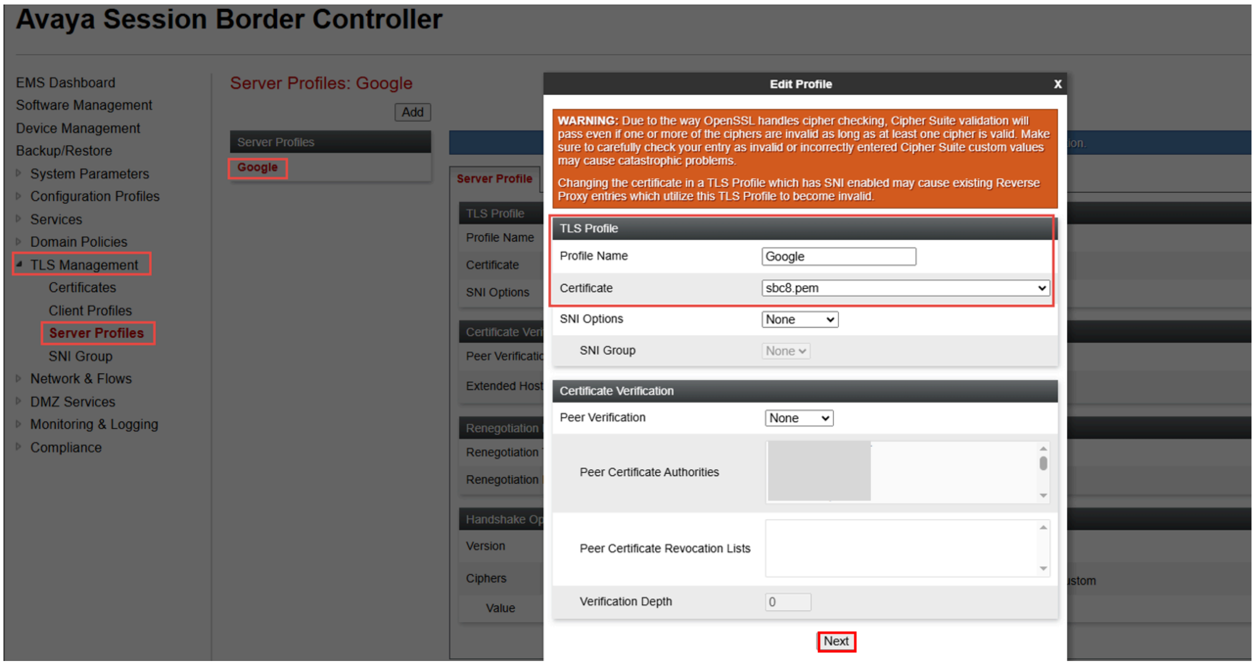

Figure 66: Server Profile towards Google CES

- Set Version: Select **TLS 1.2** versions
- Select Ciphers: Default
- Click on **Finish**

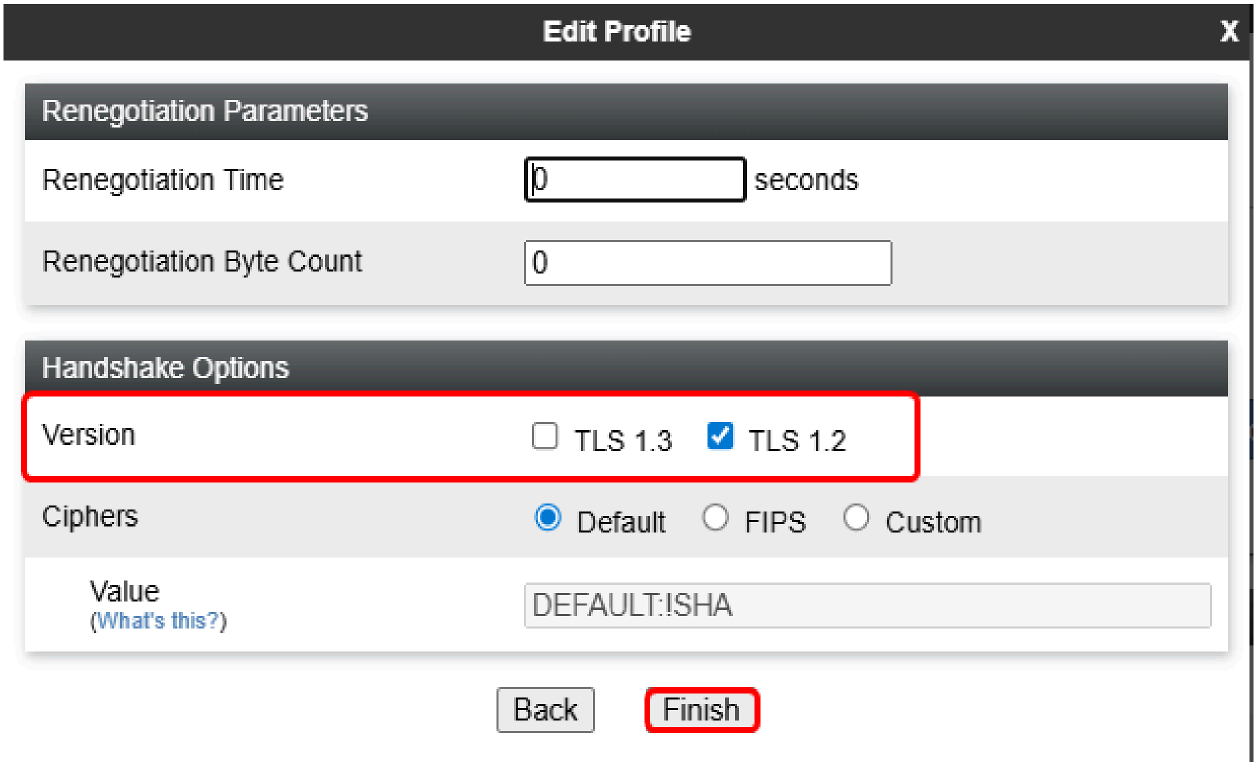

Figure 67: Server Profile towards Google CES (Cont.)

8 SIP INVITE To Google CES

8.1 SIP INVITE for SIPREC call

```
INVITE sip:+13614  @us.telephony.goog:5672;transport=tls SIP/2.0
From: "Pradeep Gopal" <sip:2145  @192.65.  >;tag=981EFC88-230A
To: <sip:+13614  @us.telephony.goog:5672;transport=tls>
CSeq: 4360 INVITE
Call-ID: a95d66761d308966e2bbc50433215103
Contact: <sip:192.65.  :5061;transport=tls>;+sip.src
Record-Route: <sip:192.65.  :5061;ipcs-line=4132;lr;transport=tls>
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK REFER, INFO, REGISTER
Supported: 100rel, replaces
Max-Forwards: 69
Via: SIP/2.0/TLS 192.65.  :5061;branch=z9hG4bK-s1632-000655365869-1--s1632-
Expires: 180
Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898  /conversations/
Sr_a95  .03>;purpose=Goog-ContactCenter-Conversation
Require: siprec
Timestamp: 1758101718
Allow-Events: telephone-event
P-Asserted-Identity: "Pradeep Gopal" <sip:2145  @192.65.  >
Remote-Address: MTAuNjQuMS43MjoxNzI1OToxOjE=
Content-Disposition: session;handling=required
Content-Type: multipart/mixed;boundary=foobar
Content-Length: 2284

--foobar
Content-Type: application/sdp

v=0
o=- 4132 1 IN IP4 10.64.
s=SIP
c=IN IP4 192.65.
t=0 0
m=audio 35160 RTP/SAVP 0 96 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=label:10
a=sendonly
a=rtpmap:96 opus/48000/2
a=fmtp:96 maxplaybackrate=16000;sprop-maxcapturerate=16000;maxaveragebitrate=20000;stereo=0;useinbandfec=0;usedtx=0;cbr=0;sprop-stereo=0
a=ptime:20
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:IPWiw5iQFrQcIQb+P86SnUROXI7evnyOvSfOgNP+
m=audio 35162 RTP/SAVP 0 96 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=label:20
a=sendonly
a=rtpmap:96 opus/48000/2
a=fmtp:96 maxplaybackrate=16000;sprop-maxcapturerate=16000;maxaveragebitrate=20000;stereo=0;useinbandfec=0;usedtx=0;cbr=0;sprop-stereo=0
a=ptime:20
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:s18wuOtpqTzJ+CItD+kw5ZfVf3mEJdGUJomop34a
```

The INVITE Request-URI should include e.164 number obtained from Google and it should have respective regional host name with SIP Signaling port :5672

Google requires the Call-Info header, and it must contain a conversation ID. The conversation ID is unique, and the format of the conversation ID follows the regex "[a-zA-Z][a-zA-Z0-9_-]*" and is assigned for each call.

dialogflow.googleapis.com/v2beta1 - API endpoint
projects/ccai-3898XX - Google Cloud CCAI project ID
conversations/Sr_xxxx - The unique conversation session ID that is assigned for that each call

The connection IP toward Google CCAI must be a public IP, not a private one.

For SIPREC, there can be single media lines with *a=sendonly*, for GTP there will be a multiple media line with *a=senrecy*
Encrypted SRTP and the allocated port range should be used (Port: 16384-32767) else CES will not receive audio.

It must be a supported crypto suite by Google.

Figure 68: SIPREC call

8.2 SIP INVITE for GTP call

```
INVITE sip:+131494  @us.telephony.goog:5672 SIP/2.0
From: "Kanitkar" <sip:+121455  @192.65.  >;tag=BC91BAE8-1784
To: <sip:+131494  @us.telephony.goog:5672>
CSeq: 101 INVITE
Call-ID: 2891181e459655649e5e9861e89f3a44
Contact: <sip:2145509018@192.65.  :5061;transport=tls>
Record-Route: <sip:192.65.  :5061;ipcs-line=705;lr;transport=tls>
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK REFER, SUBSCRIBE, NOTIFY,
Supported: 100rel, timer, resource-priority, replaces
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 69
Via: SIP/2.0/TLS 192.65.  :5061;branch=z9hG4bK-s1632-001787678602-1--s
Via: SIP/2.0/TCP 10.64.  :5060;branch=z9hG4bK6BDD237C
Expires: 180
Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/ccai-389  /conversations/
Sr 2  44>;purpose=Goog-ContactCenter-Conversation
Date: Wed, 24 Sep 2025 11:27:09 GMT
Timestamp: 1758713229
Allow-Events: telephone-event
P-Asserted-Identity: "Kanitkar" <sip:2145  @192.65.  >
Min-SE:  1800
Remote-Address: MTAuNjQuMS43MjoxNjc0MzoxOjE=
Content-Disposition: session;handling=required
Content-Type: application/sdp
Cisco-Guid: 1040042236-2557481456-3134219286-2368317232
Content-Length: 375

v=0
o=CiscoSystemsSIP-GW-UserAgent 2001 8907 IN IP4 10.64.
s=SIP
c=IN IP4 192.65.
t=0 0
m=audio 35168 RTP/SAVP 101 0 8 19
c=IN IP4 192.65.
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=rtpmap:19 CN/8000
a=ptime:20
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:7Y5aL7qOjXJfgXt5/4DbumxgJL8zwK/Lto+N3x7I
```

The INVITE Request-URI should include e.164 number obtained from Google and it should have respective regional host name with SIP Signaling port :5672

Google requires the Call-Info header, and it must contain a conversation ID. The conversation ID is unique, and the format of the conversation ID follows the regex "[a-zA-Z][a-zA-Z0-9_-]*" and is assigned for each call.

dialogflow.googleapis.com/v2beta1 - API endpoint
projects/ccai-3898XX - Google Cloud CCAI project ID
conversations/Sr_xxxx - The unique conversation session ID that is assigned for that each call

The connection IP toward Google CCAI must be a public IP, not a private one.

For GTP, there can be single media lines with *a=sendrecv*, for SIPREC there will be a multiple media line with *a=sendonly*
Encrypted SRTP and the allocated port range should be used (Port: 16384-32767) else CES will not receive audio.

It must be a supported crypto suite by Google.

Figure 69: GTP call

9 Summary of Tests and Results

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
35	UUI header test	Use the UUI header as opposed to Call-Info header to send conversation id	Call should process as normal, and recording under conversation ID derived from UUI header as opposed to Call-Info	PASSED	Calls get successfully connected to Live agent when using UUI header instead of Call-Info header.
36	Keep_conversation_running=TRUE test	ConversationProfile needs to have SipConfig set with keepConversationRunning = TRUE. Send first call with a Call-Info header and have a call for 2 turns. End the call. Send second call with the SAME Call-Info header as above and have a call for 3 turns. End the call.	Two calls having the same Call-Info has both conversation details.	PASSED	Both call transcripts are present for the same conversation session id.
37	Live Agent Transfer	Call goes to virtual agent, initial live agent handoff and verify outgoing SIP INVITE, call connection and disconnection		PASSED	Call gets connected successfully to BOT and INVITE sent successfully to connect with Agent.
38	Live Agent Transfer	Call goes to virtual agent, initial live agent handoff and verify outgoing SIP INVITE, call connection and disconnection. Make calls to		PASSED	Call was connected successfully with live agent and when performing conversation "Speak to an agent", a new

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		+1314944XXXX, "speak to an agent", +1-972-852-XXXX should then ring and get connected as the agent			INVITE was sent from Google to transfer to an agent and call gets connected successfully with both-way audio.
39	UUI headers	Call goes to virtual agent, say "end the call", validate that the SIP BYE has a UUI header Make calls to +1314944XXXX, "end the call", check SIP BYE and ensure there is one or more (identify if there are 3 or 1) UUI headers with purpose Goog-Session-Param		PASSED	Calls get connected successfully to live agents and when performing conversation "End the call", the call gets disconnected normally with 3 UUI header.
40	SIP REFER	Call goes to virtual agent, say "send a sip refer", validate that a SIP REFER is received to 972-852-XXXX. Make calls to +1314944XXXX, "send a sip refer", SIP REFER should be received with refer to set to 972-852-XXXX		PASSED	REFER request and the transfer are successful.