

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



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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.

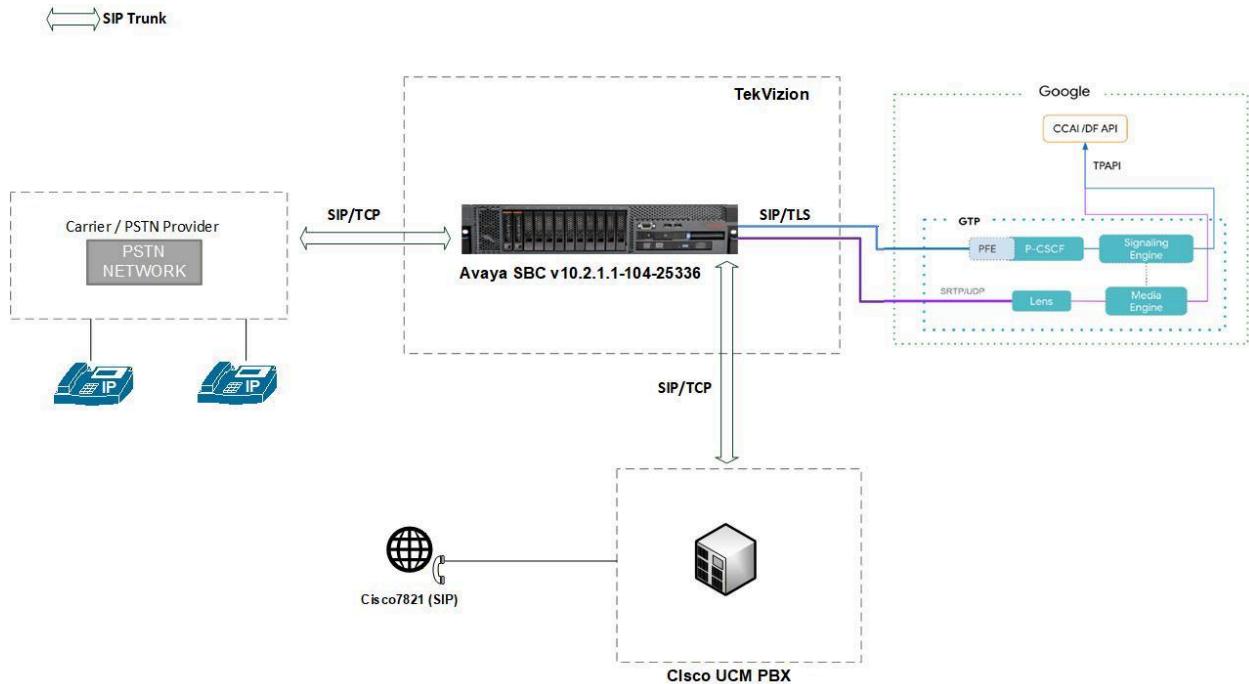


Figure 1: SIP Trunk Lab Reference Network

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.

The image shows the Avaya SBC Device Management page. The top navigation bar includes 'Device: SA', 'Alarms 15', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The left sidebar has 'EMS' and 'SA' selected. The main content area is titled 'Device Management' and shows a table of devices. The table has columns: 'Device Name', 'Management IP', 'Version', and 'Status'. One row is shown: 'SA' with Management IP '10.80.11.222', Version '10.2.1.1-104-25336', and Status 'Commissioned'. Action buttons for 'Reboot', 'Shutdown', 'Restart Application', 'View', 'Edit', and 'Uninstall' are shown to the right of the table.

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**

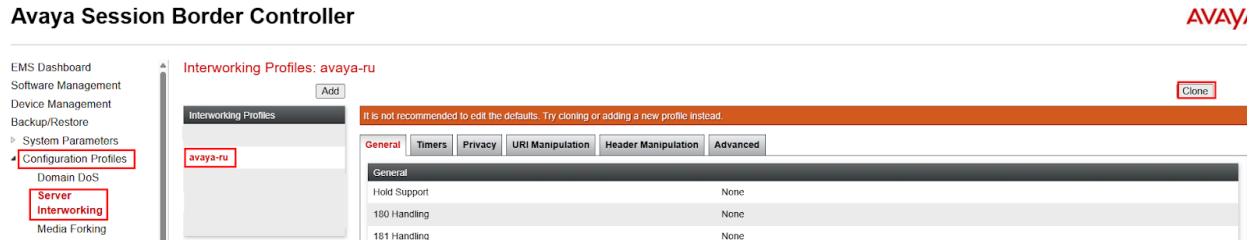


Figure 4: Server Interworking Profile for OnPrem PBX

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

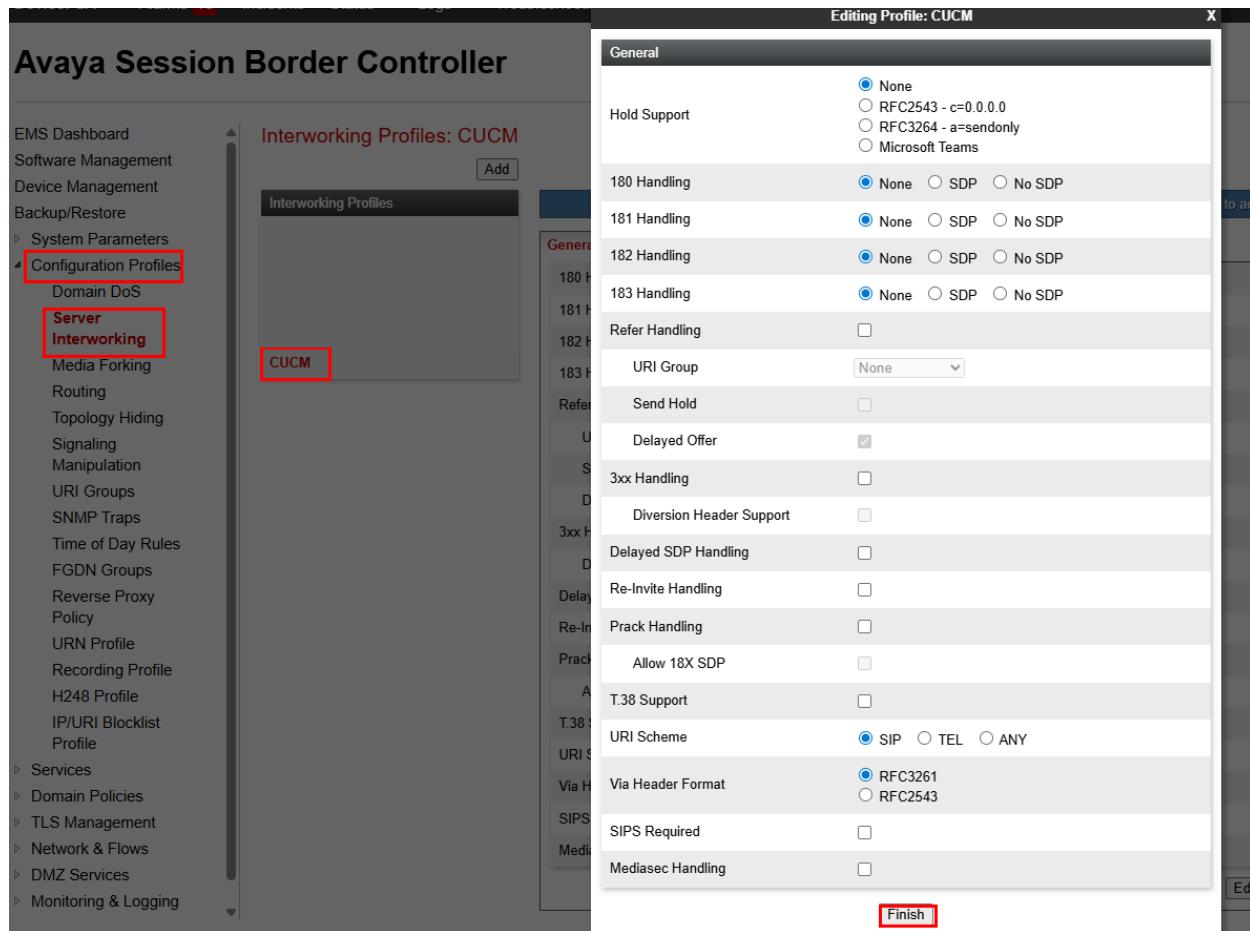


Figure 5: Server Interworking Profile for OnPrem PBX (Cont.)

Server Interworking for Google CES

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

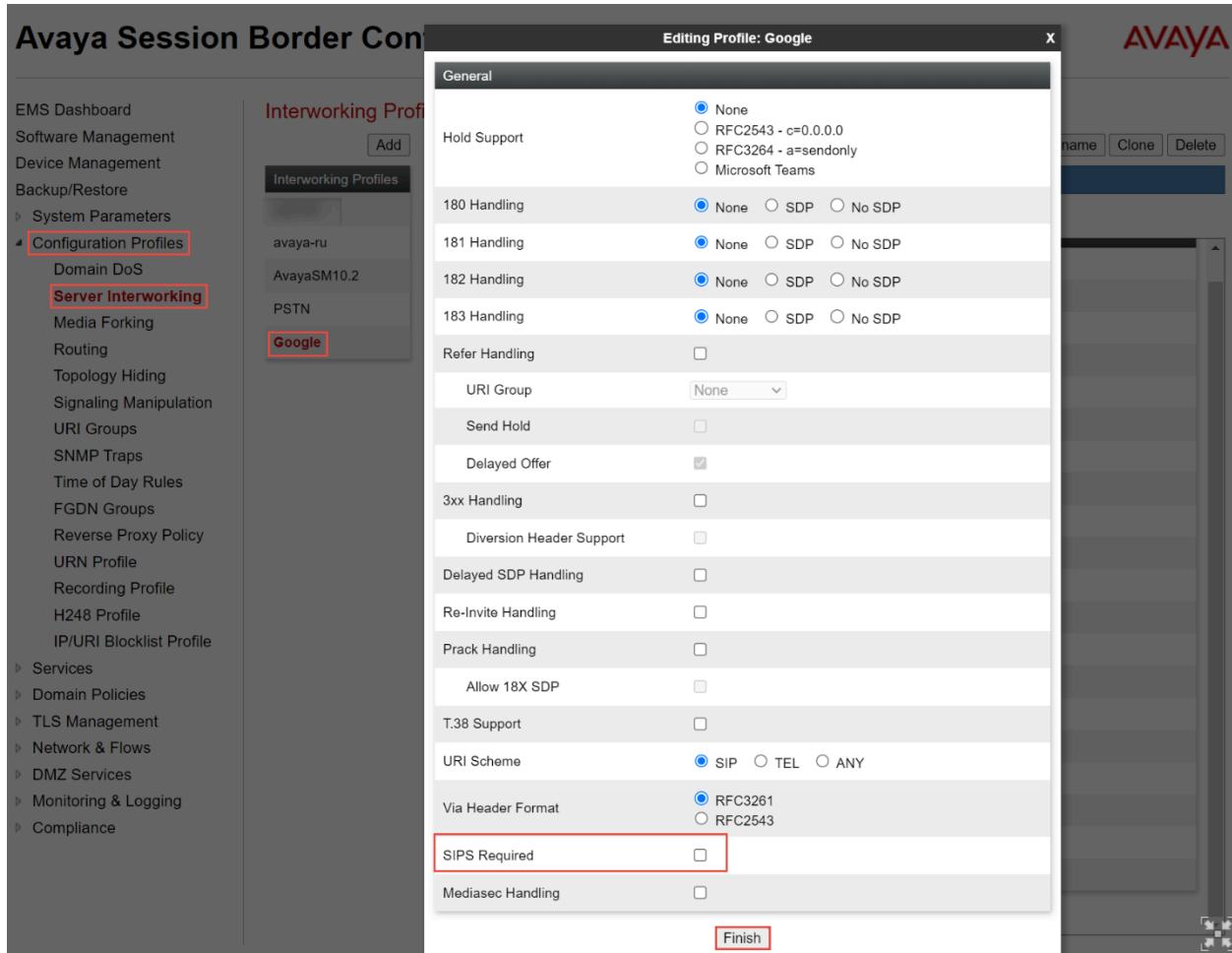
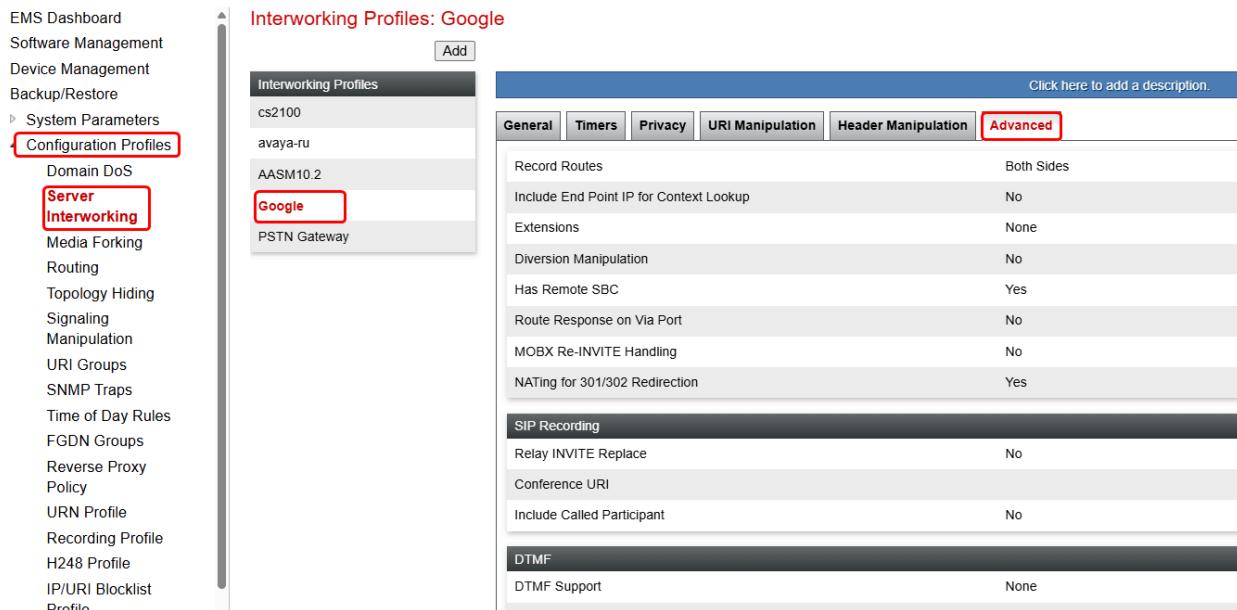


Figure 6: Server Interworking Profile for Google CES

Avaya Session Border Controller



The screenshot shows the EMS Dashboard of the Avaya Session Border Controller. On the left, a navigation tree is visible with several nodes expanded, including 'System Parameters' and 'Configuration Profiles'. Under 'Configuration Profiles', 'Server Interworking' is selected and highlighted with a red box. The main content area is titled 'Interworking Profiles: Google'. It shows a list of profiles: 'cs2100', 'avaya-ru', 'AASM10.2', and 'Google', with 'Google' also highlighted with a red box. To the right, a detailed configuration page for the 'Google' profile is displayed. The 'Advanced' tab is selected. The configuration includes sections for 'General', 'Timers', 'Privacy', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'Advanced' section contains the following settings:

Setting	Value
Record Routes	Both Sides
Include End Point IP for Context Lookup	No
Extensions	None
Diversion Manipulation	No
Has Remote SBC	Yes
Route Response on Via Port	No
MOBX Re-INVITE Handling	No
NATing for 301/302 Redirection	Yes
SIP Recording	
Relay INVITE Replace	No
Conference URI	
Include Called Participant	No
DTMF	
DTMF Support	None

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**

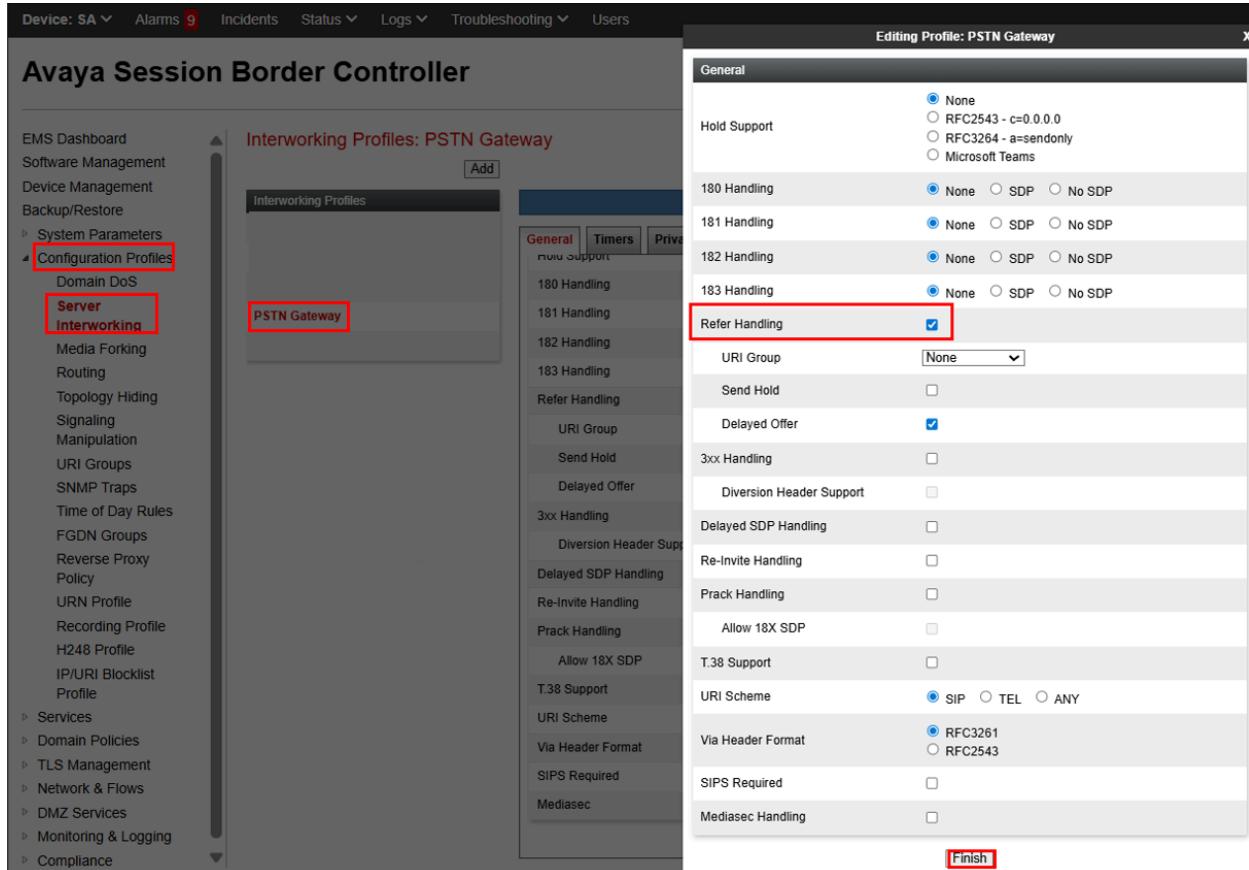


Figure 8: Server Interworking Profile for PSTN Gateway

Avaya Session Border Controller

The screenshot shows the EMS Dashboard of the Avaya Session Border Controller. On the left, a sidebar lists various configuration profiles. The 'Server Interworking' profile is selected and highlighted with a red box. The main content area is titled 'Interworking Profiles: PSTN Gateway'. It shows a list of profiles: 'AASM10.2', 'Google', and 'PSTN Gateway', with 'PSTN Gateway' also highlighted with a red box. A large 'Add' button is at the top right. Below the profiles, there are several tabs: General, Timers, Privacy, URI Manipulation, Header Manipulation, and Advanced (which is selected and highlighted with a red box). The 'Advanced' tab contains configuration settings for various SIP-related features. A note 'Click here to add a description' is at the top right of the configuration area.

Setting	Value
Record Routes	Both Sides
Include End Point IP for Context Lookup	No
Extensions	None
Diversion Manipulation	No
Has Remote SBC	Yes
Route Response on Via Port	No
MOBX Re-INVITE Handling	No
NATing for 301/302 Redirection	Yes
SIP Recording	
Relay INVITE Replace	No
Conference URI	
Include Called Participant	No
DTMF	
DTMF Support	None
Adaptive Inband Detection	No

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**

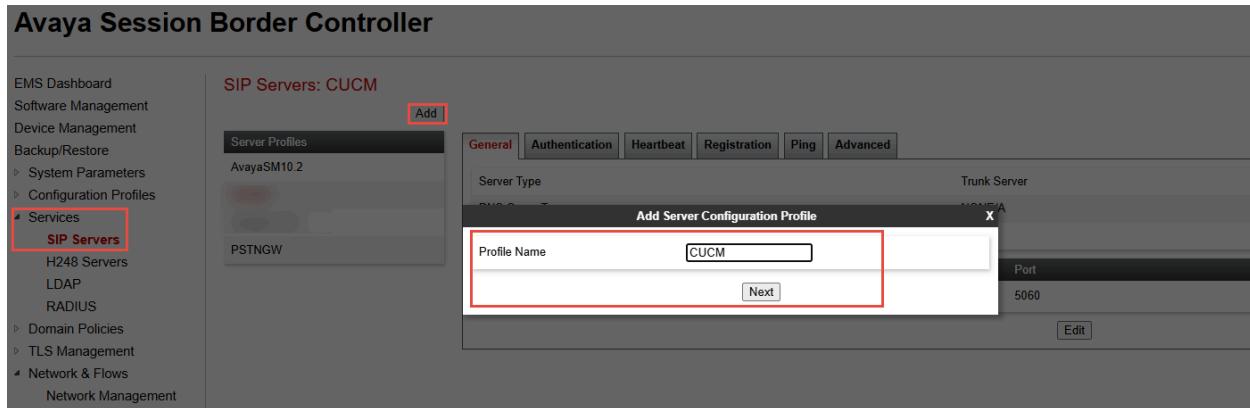


Figure 10: SIP Server for OnPrem PBX

- 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**

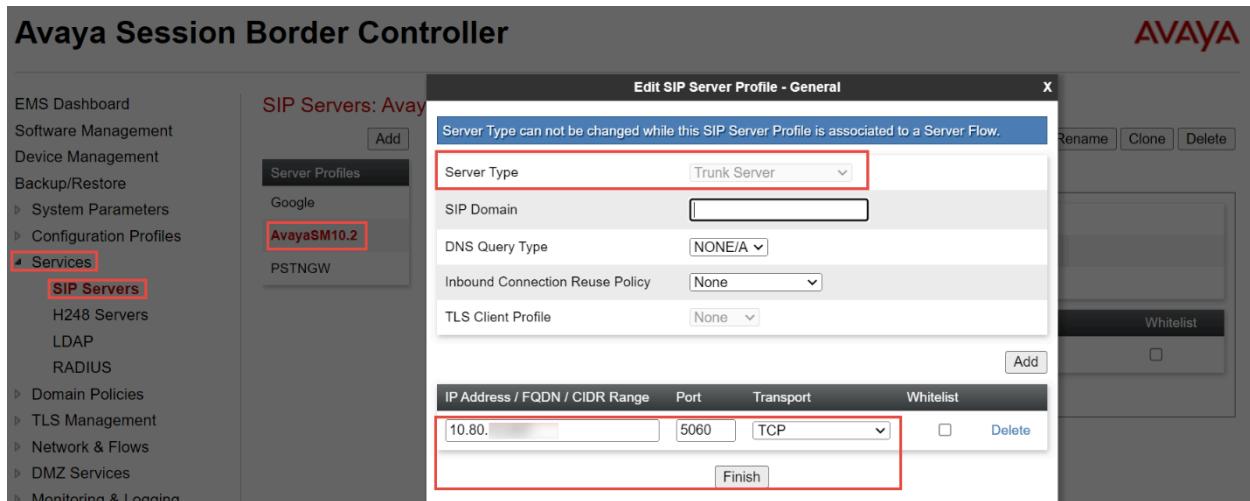


Figure 11: SIP Server for OnPrem PBX (Cont.)

- 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**

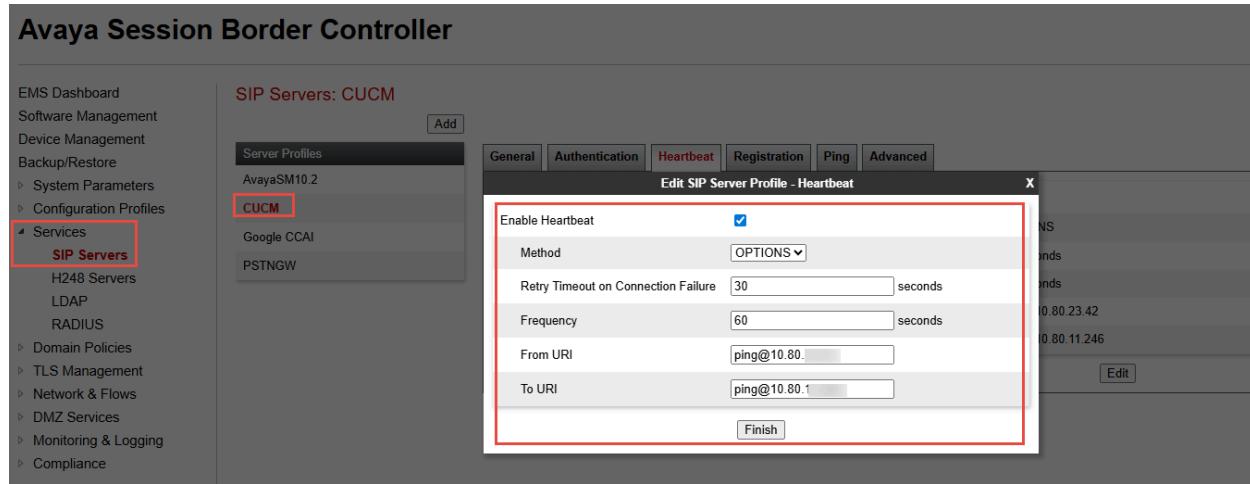


Figure 12: SIP Server for OnPrem PBX (Cont.)

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

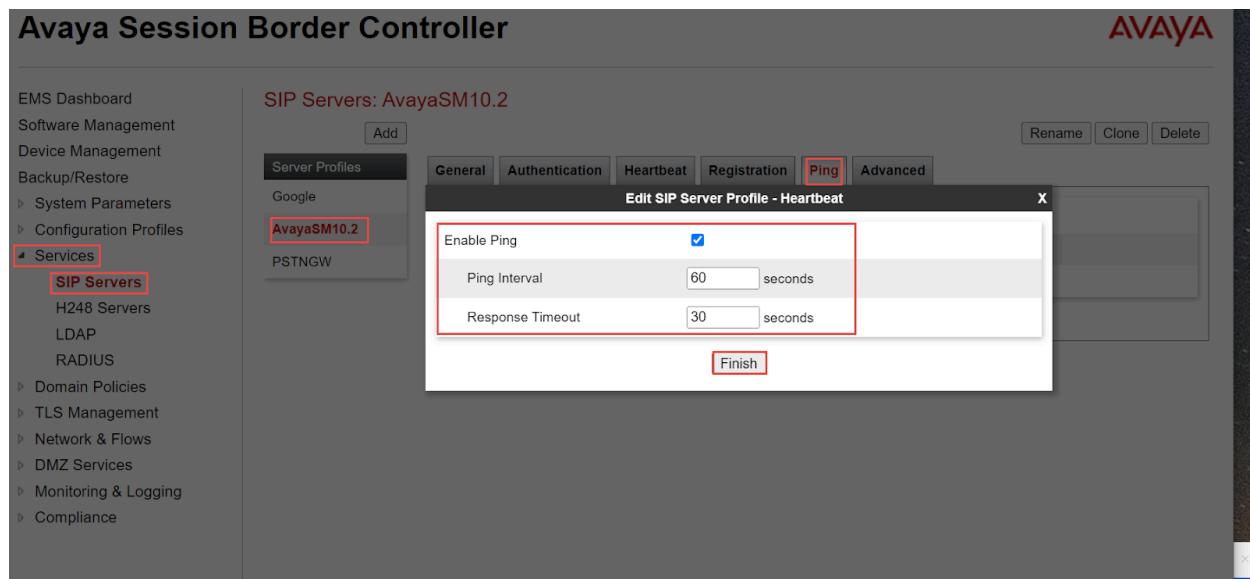


Figure 13: SIP Server for OnPrem PBX (Cont.)

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

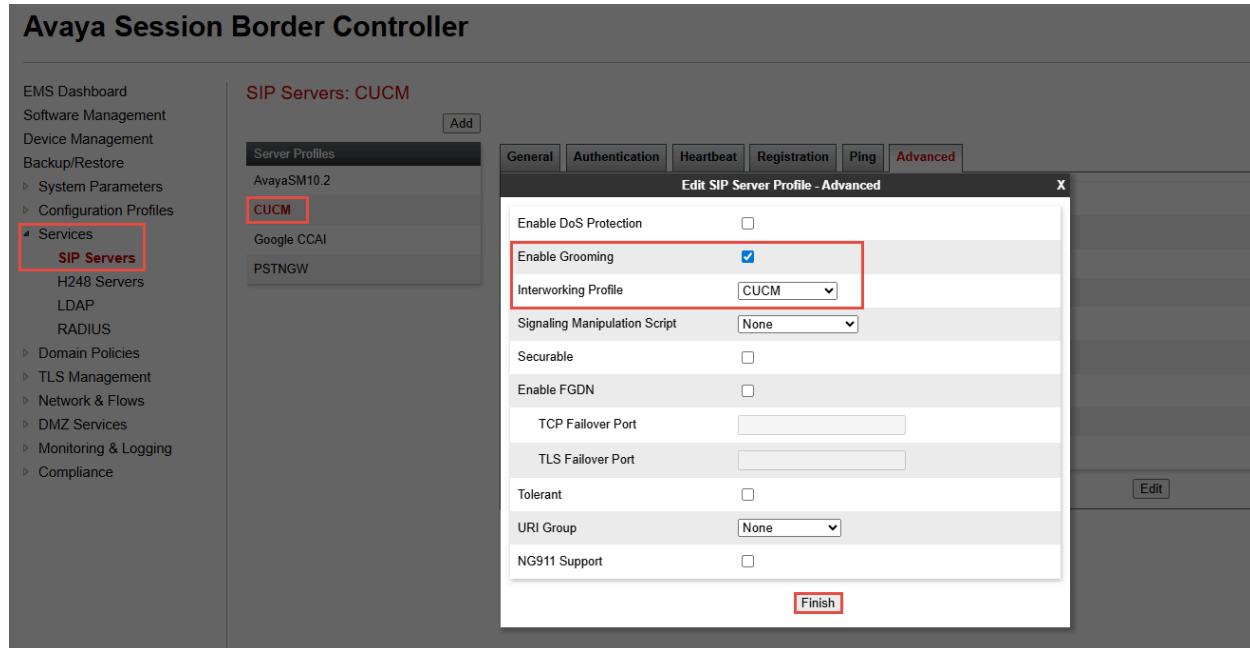


Figure 14: SIP Server for OnPrem PBX (Cont.)

SIP Server for Google CES

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

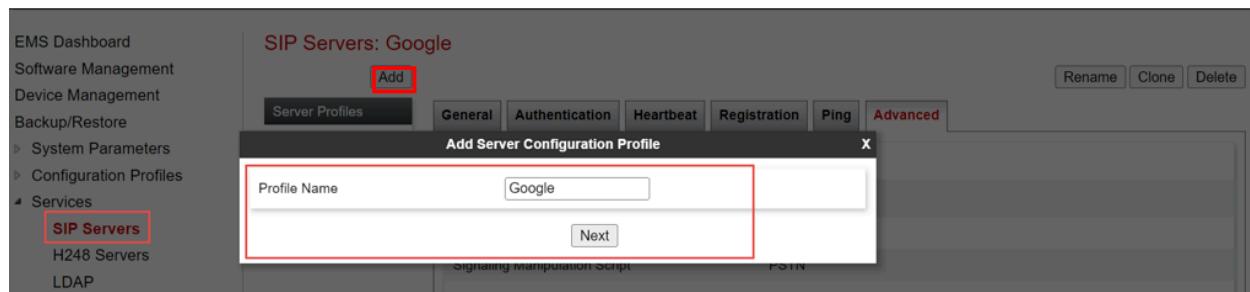


Figure 15: SIP Server for Google CES

- 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**

Avaya Session Border Controller

EMS Dashboard

Software Management

Device Management

Backup/Restore

System Parameters

Configuration Profiles

Services

SIP Servers

H248 Servers

LDAP

RADIUS

Domain Policies

TLS Management

Network & Flows

DMZ Services

Monitoring & Logging

SIP Servers: Google

Add

Server Profiles

Google

Edit SIP Server Profile - General

Server Type: Trunk Server

SIP Domain: (empty)

DNS Query Type: NONE/A

Inbound Connection Reuse Policy: None

TLS Client Profile: Google

IP Address / FQDN / CIDR Range: us.telephony.goog

Port: 5672

Transport: TLS

Whitelist: (checkbox)

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**

Edit SIP Server Profile - Heartbeat

Enable Heartbeat	<input checked="" type="checkbox"/>
Method	OPTIONS <input type="button" value="▼"/>
Retry Timeout on Connection Failure	30 <input type="text"/> seconds
Frequency	30 <input type="text"/> seconds
From URI	ping@192.65. <input type="text"/>
To URI	ping@us.telephony.goog <input type="text"/>
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**

Avaya Session Border Controller

EMS Dashboard

Software Management

Device Management

Backup/Restore

System Parameters

Configuration Profiles

Services

SIP Servers

H248 Servers

LDAP

RADIUS

Domain Policies

TLS Management

SIP Servers: Google

Add	Rename Clone Delete
Server Profiles	General Heartbeat Registration Ping Advanced
Google	AvayaSM10.2 PSTNGW

Edit SIP Server Profile - Heartbeat

Enable Ping	<input checked="" type="checkbox"/>
Ping Interval	60 <input type="text"/> seconds
Response Timeout	30 <input type="text"/> seconds
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**

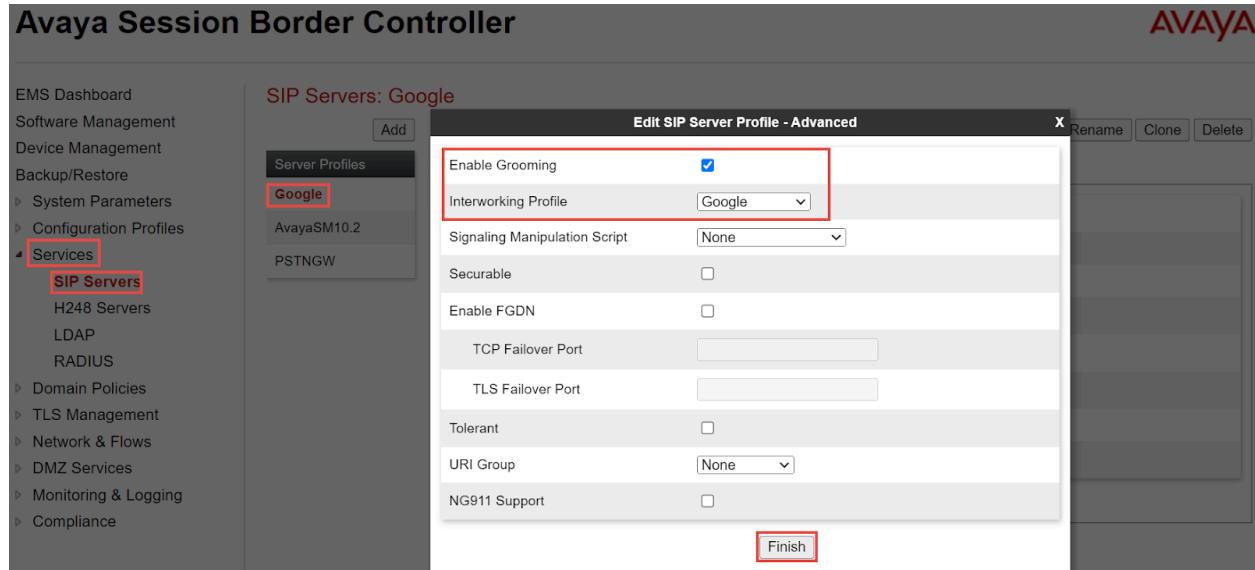


Figure 19: SIP Server for Google CES (Cont.)

SIP Server for PSTN Gateway

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

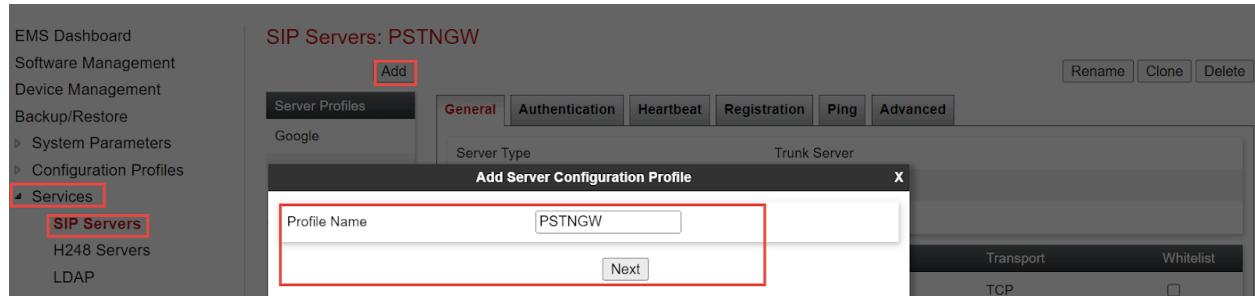


Figure 20: SIP Server for PSTN Gateway

- 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**

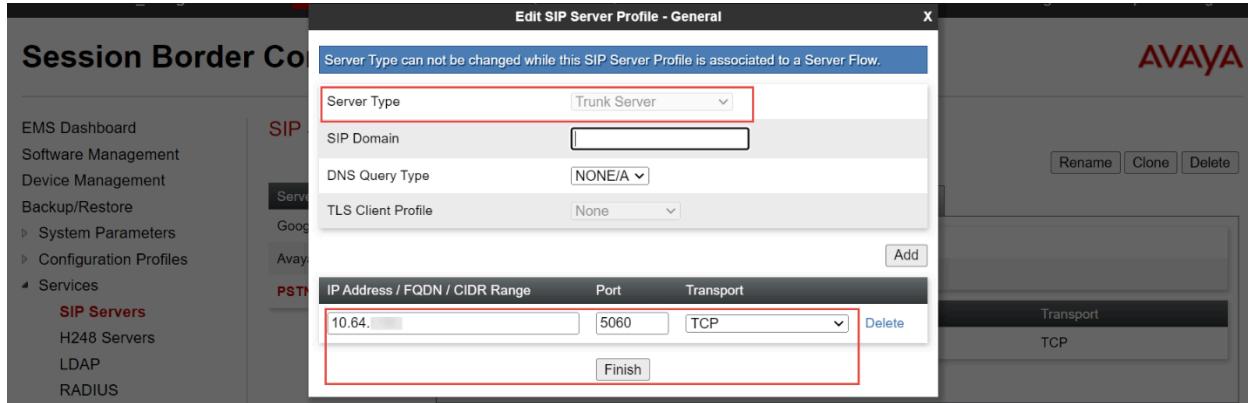


Figure 21: SIP Server for PSTN Gateway (Cont.)

- 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**

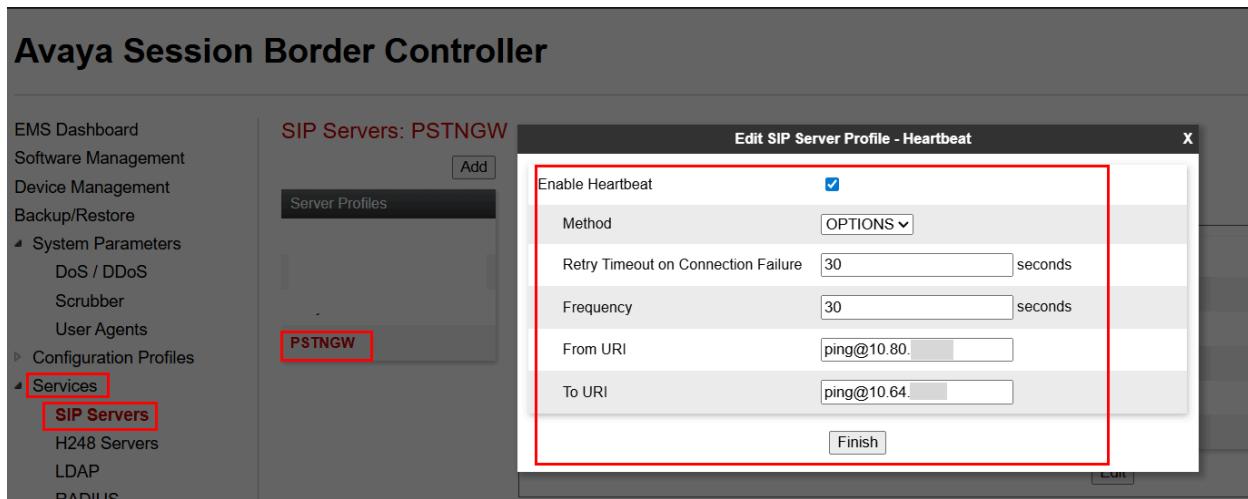


Figure 22: SIP Server for PSTN Gateway (Cont.)

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

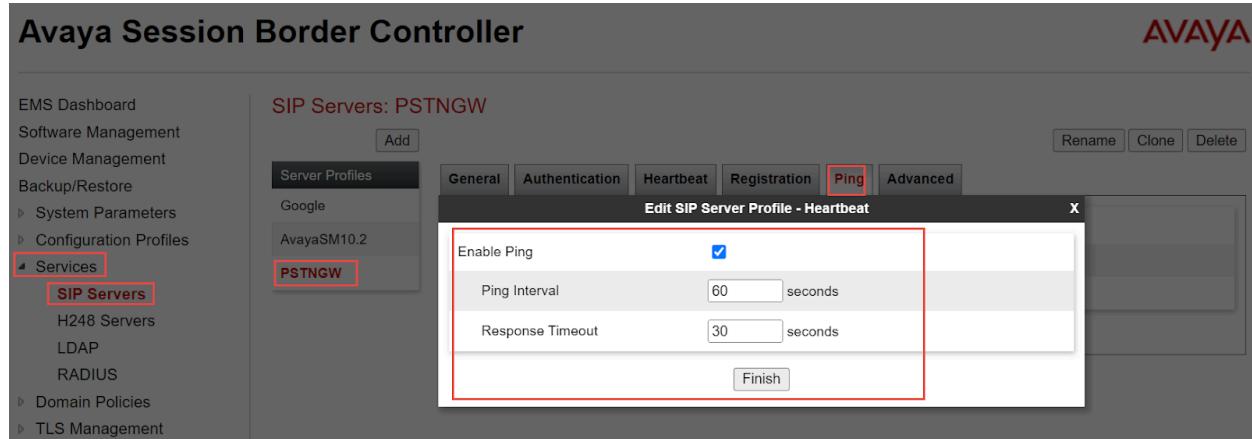


Figure 23: SIP Server for PSTN Gateway (Cont.)

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

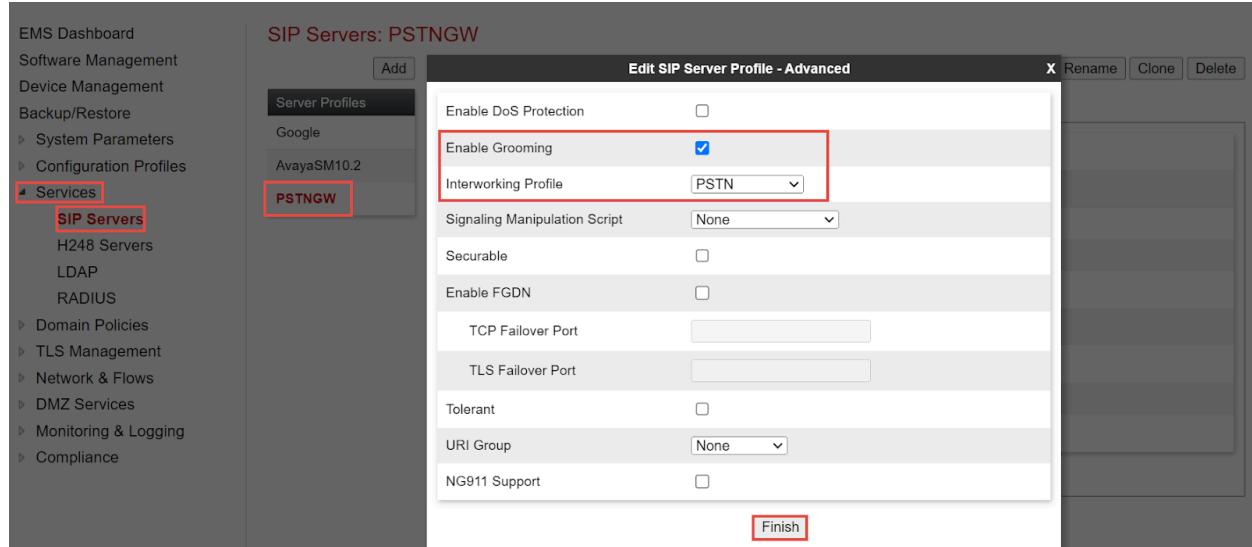


Figure 24: SIP Server for PSTN Gateway (Cont.)

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**

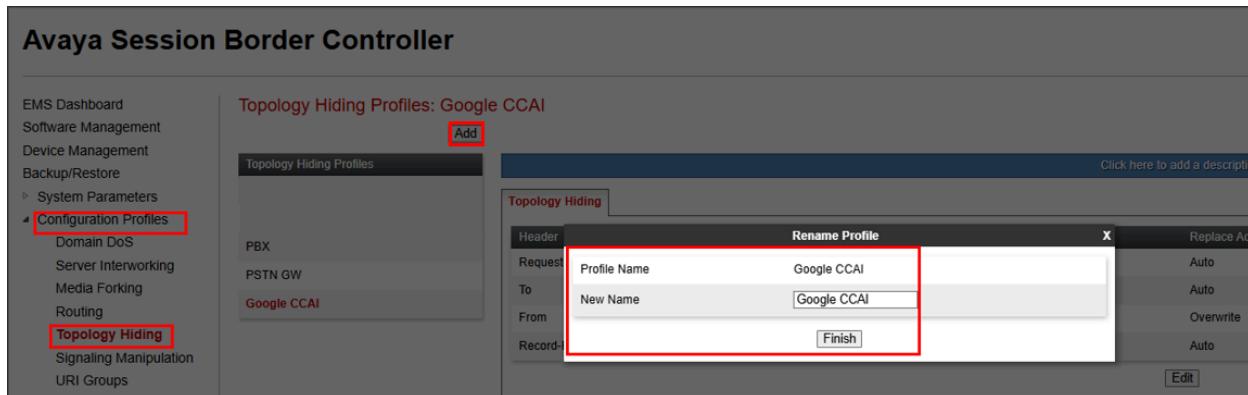


Figure 25: Topology Hiding for Google CES

- 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**

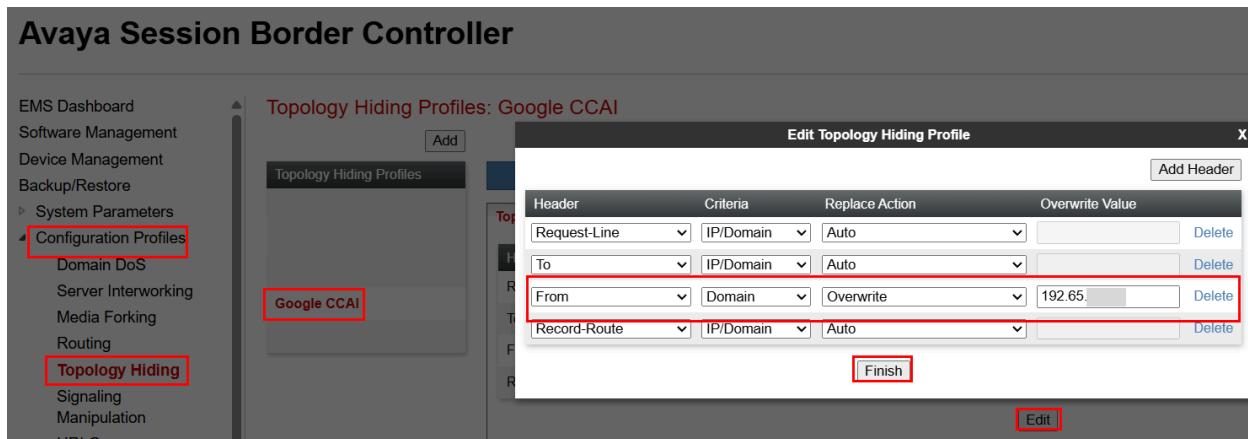


Figure 26: Topology Hiding for Google CES (Cont.)

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**

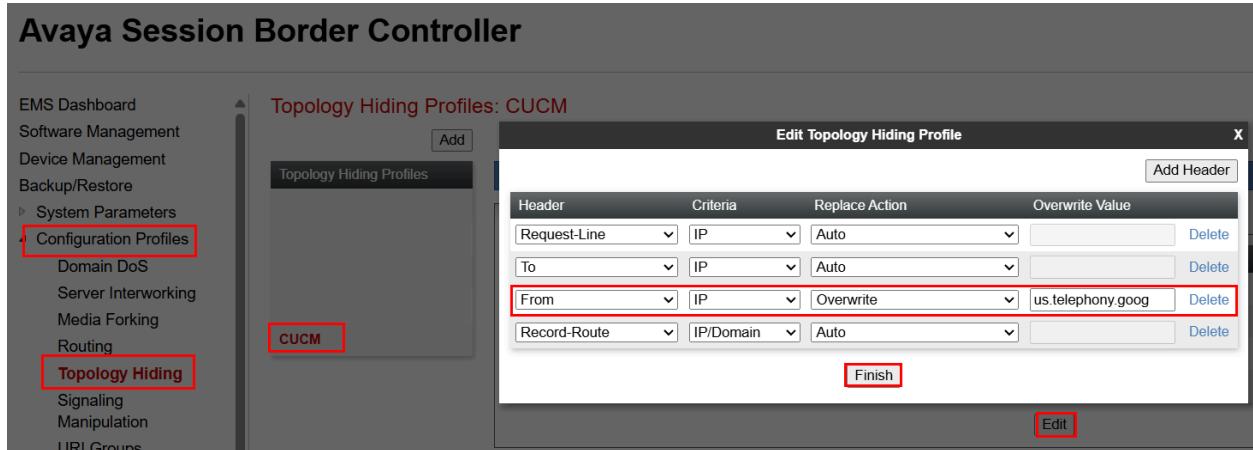


Figure 27: Topology Hiding for OnPrem PBX

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**

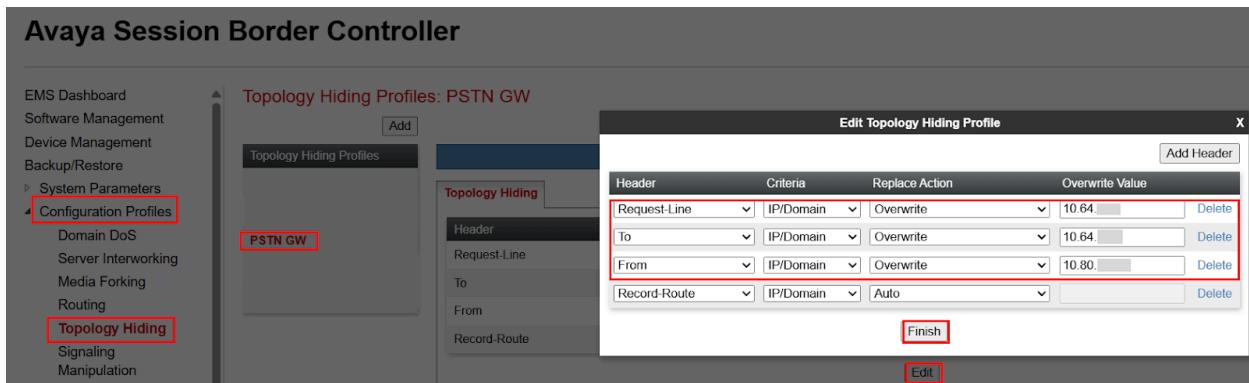


Figure 28: Topology Hiding for PSTN Gateway

7.4.5 Routing

Routing for OnPrem PBX

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

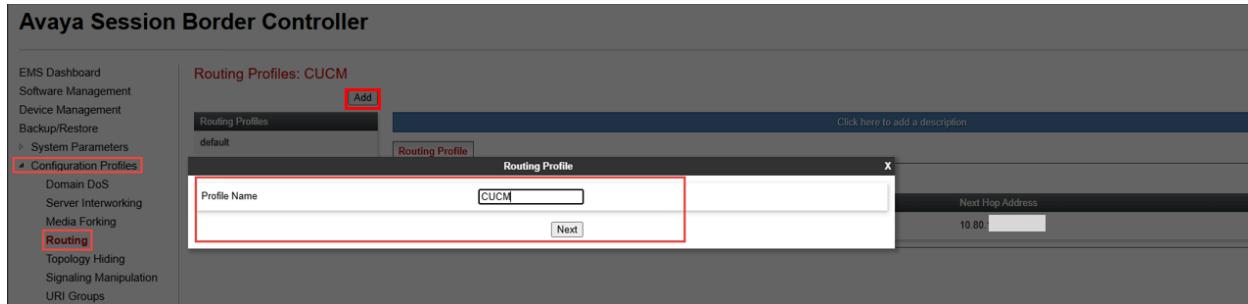


Figure 29: Routing for OnPrem PBX

- 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**

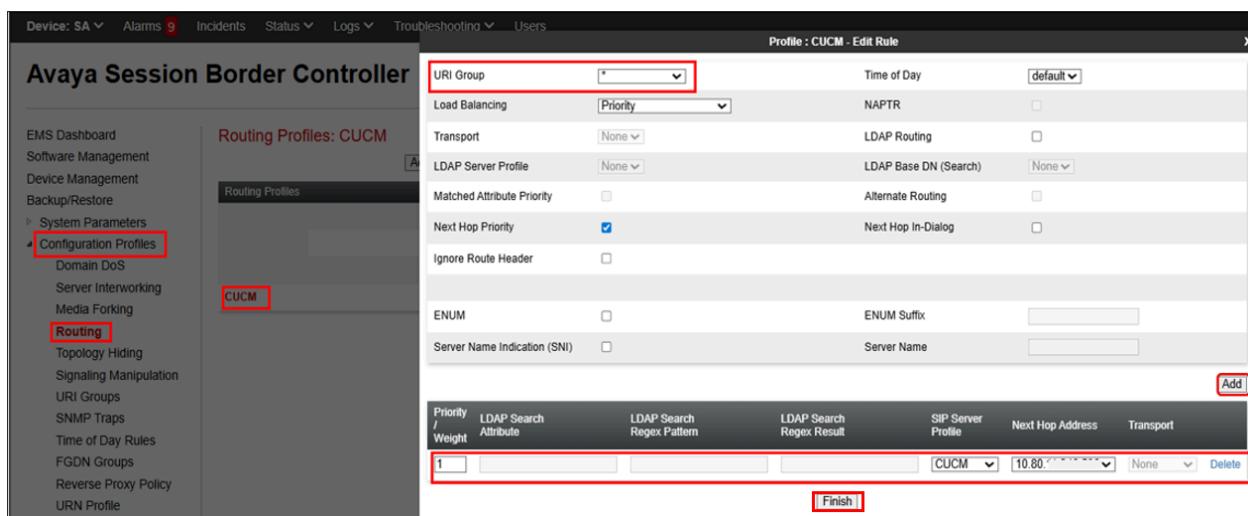


Figure 30: Routing for OnPrem PBX (Cont.)

Routing for PSTN Gateway

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

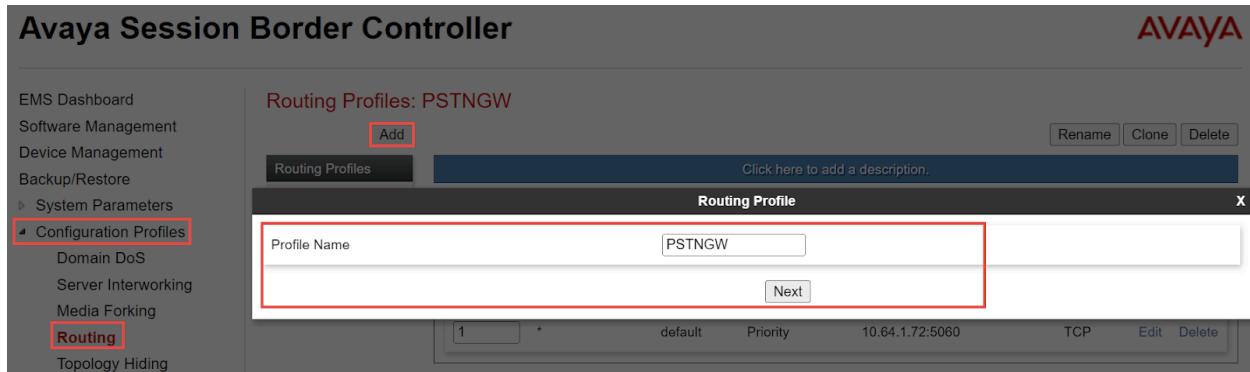


Figure 31: Routing for PSTN Gateway

- 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**

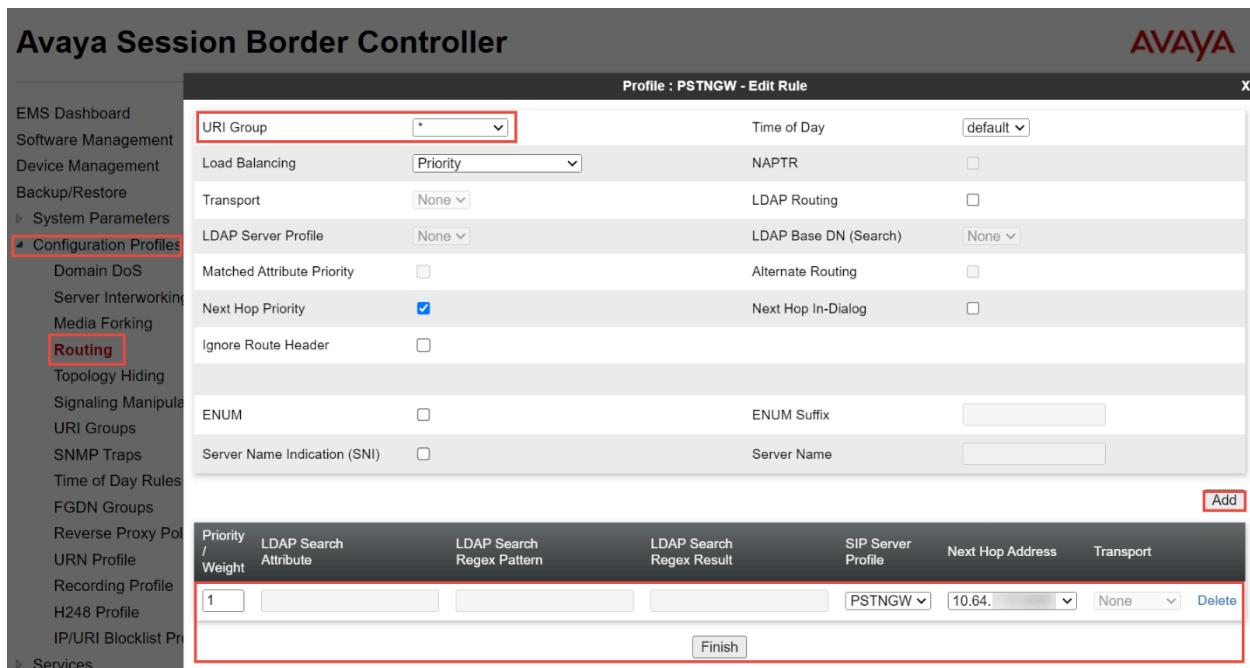


Figure 32: Routing for PSTN Gateway (Cont.)

Routing for Google CES

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

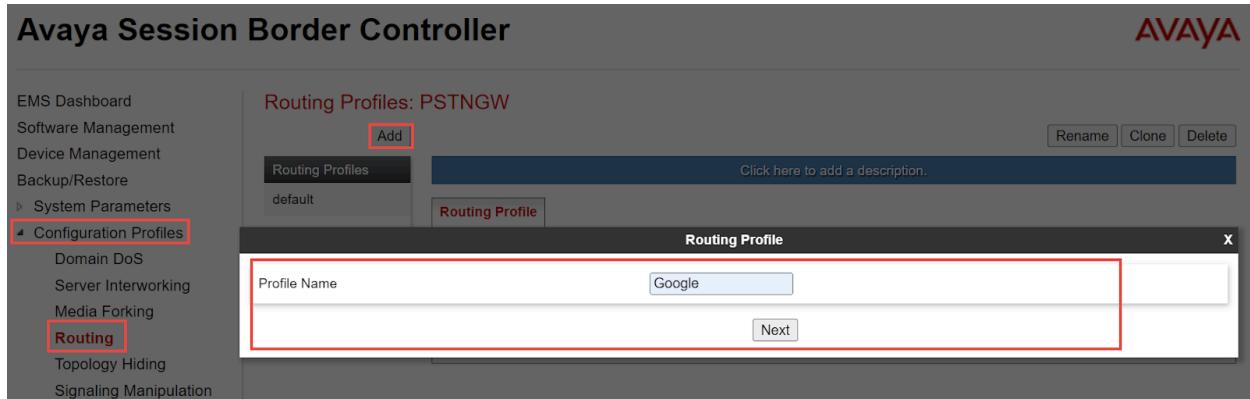


Figure 33: Routing for Google CES

- 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**

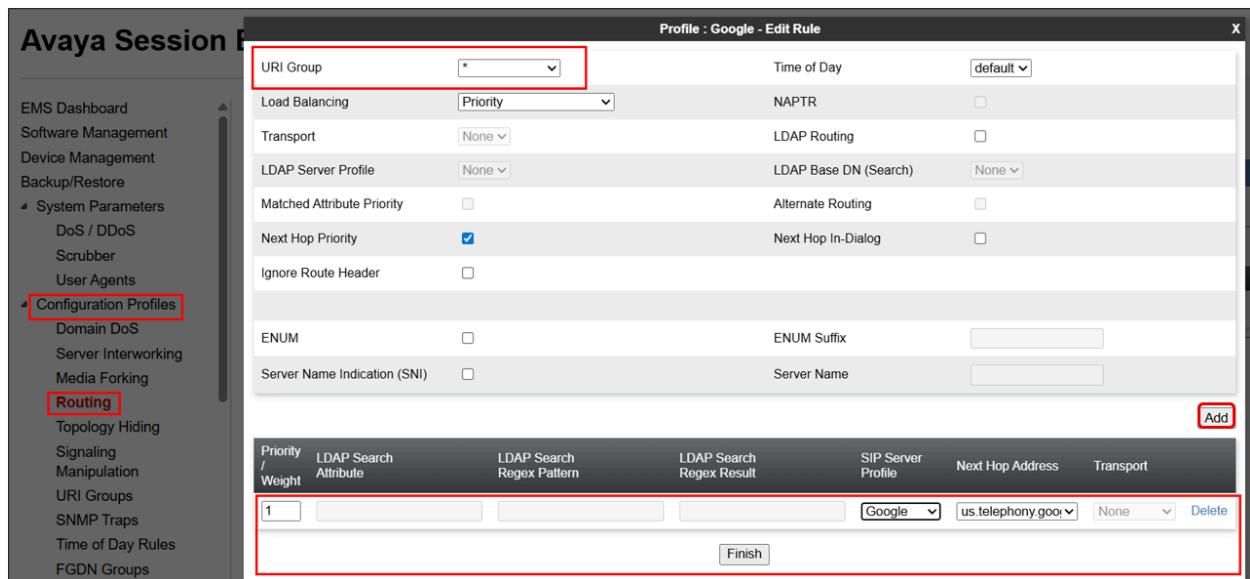


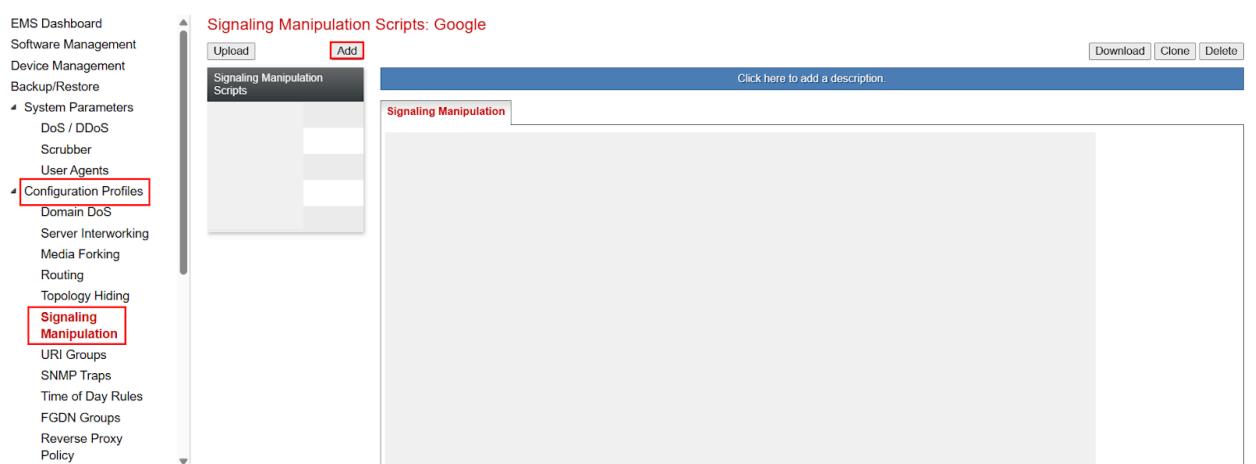
Figure 34: Routing for Google CES (Cont.)

7.4.6 Signaling Manipulation

Signaling Manipulation for Google CES

- Navigate: Configuration Profiles □ Signaling Manipulation
- Click Add

Avaya Session Border Controller 



EMS Dashboard
Software Management
Device Management
Backup/Restore
System Parameters
DoS / DDoS
Scrubber
User Agents
Configuration Profiles
Domain DoS
Server Interworking
Media Forking
Routing
Topology Hiding
Signaling Manipulation
URI Groups
SNMP Traps
Time of Day Rules
FGDN Groups
Reverse Proxy
Policy

Signaling Manipulation Scripts

Click here to add a description.

Signaling Manipulation

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-38XXXX/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**

Signaling Manipulation Editor

AVAYA

Title Save

```

1 within session "all"
2 {
3   act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING" and %METHOD="INVITE"
4   {
5     %aor = %HEADERS["Call-ID"][1];
6     %baseUri = "<http:&slash;dialogflow.googleapis.com/v2beta1/projects/ccai-3898XX/conversations/Sr_";
7     append( %baseUri, %aor);
8     %newUri = ">purpose=Goog-ContactCenter-Conversation";
9     append( %baseUri, %newUri);
10    %HEADERS["Call-Info"][] = %baseUri;
11    %HEADERS["Call-Info"][]1.URI.regex_replace("&slash;","");
12    %HEADERS["Request-Line"][]1.URI.USER.regex_replace("^.*", "+1314944XXXX");
13    %HEADERS["TO"][]1.URI.USER.regex_replace("^.*", "+1314944XXXX");
14    %HEADERS["FROM"][]1.URI.USER.regex_replace("^.*.....", "+1214550XXXX");
15  }
16
17
18

```

Figure 36: Signaling Manipulation Editor - Google CES

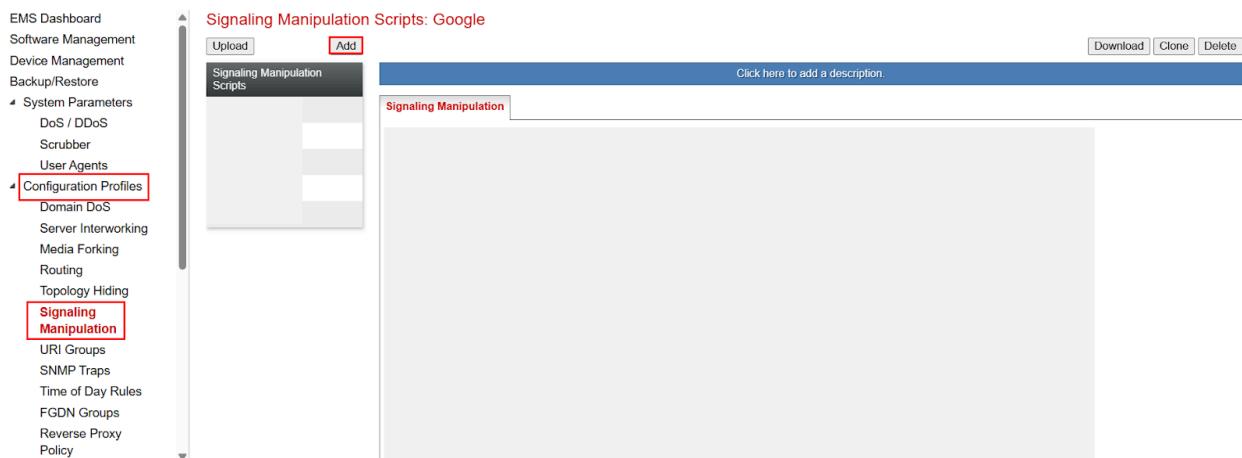
Sigma Script:

```
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

Avaya Session Border Controller



EMS Dashboard
Software Management
Device Management
Backup/Restore
System Parameters
DoS / DDoS
Scrubber
User Agents
Configuration Profiles
Domain DoS
Server Interworking
Media Forking
Routing
Topology Hiding
Signaling Manipulation
URI Groups
SNMP Traps
Time of Day Rules
FGDN Groups
Reverse Proxy
Policy

Signaling Manipulation Scripts: Google

Upload Add Download Clone Delete

Click here to add a description.

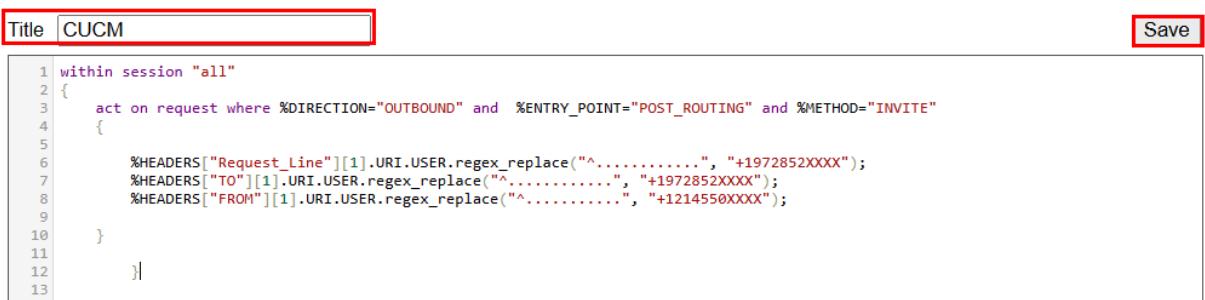
Signaling Manipulation

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**



Signaling Manipulation Editor



Title CUCM Save

```
1 within session "all"
2 {
3     act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING" and %METHOD="INVITE"
4     {
5
6         %HEADERS["Request_Line"][1].URI.USER.regex_replace("^.....", "+1972852XXXX");
7         %HEADERS["TO"][1].URI.USER.regex_replace("^.....", "+1972852XXXX");
8         %HEADERS["FROM"][1].URI.USER.regex_replace(".....", "+1214550XXXX");
9
10    }
11
12
13 }
```

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**

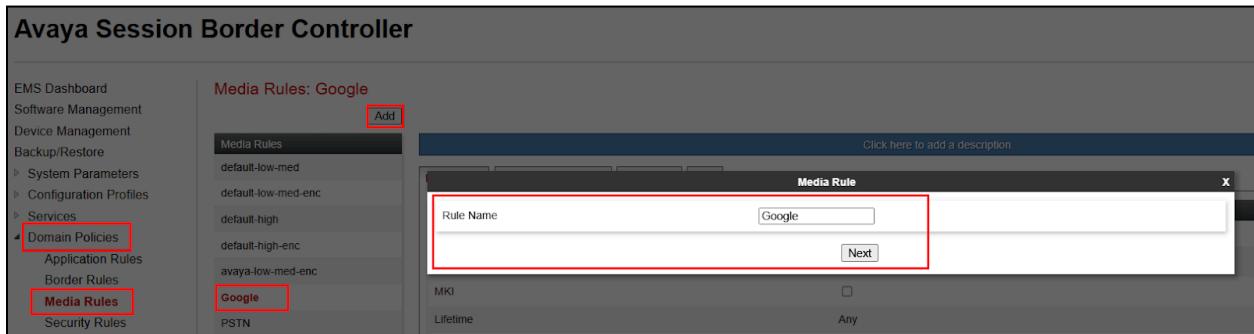


Figure 39: Media Rules

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

Media Encryption

Audio Encryption

Preferred Format #1	SRTP_AES_CM_128_HMAC_SHA1_80
Preferred Format #2	NONE
Preferred Format #3	NONE
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime Leave blank to match any value.	2 ⁸ <input type="text"/>
Interworking	<input checked="" type="checkbox"/>
Symmetric Context Reset	<input type="checkbox"/>
Key Change in New Offer	<input type="checkbox"/>

Video Encryption

Preferred Format #1	SRTP_AES_CM_128_HMAC_SHA1_80
Preferred Format #2	NONE
Preferred Format #3	NONE
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime Leave blank to match any value.	2 ⁸ <input type="text"/>
Interworking	<input type="checkbox"/>
Symmetric Context Reset	<input type="checkbox"/>
Key Change in New Offer	<input type="checkbox"/>

Miscellaneous

Capability Negotiation	<input type="checkbox"/>
------------------------	--------------------------

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**

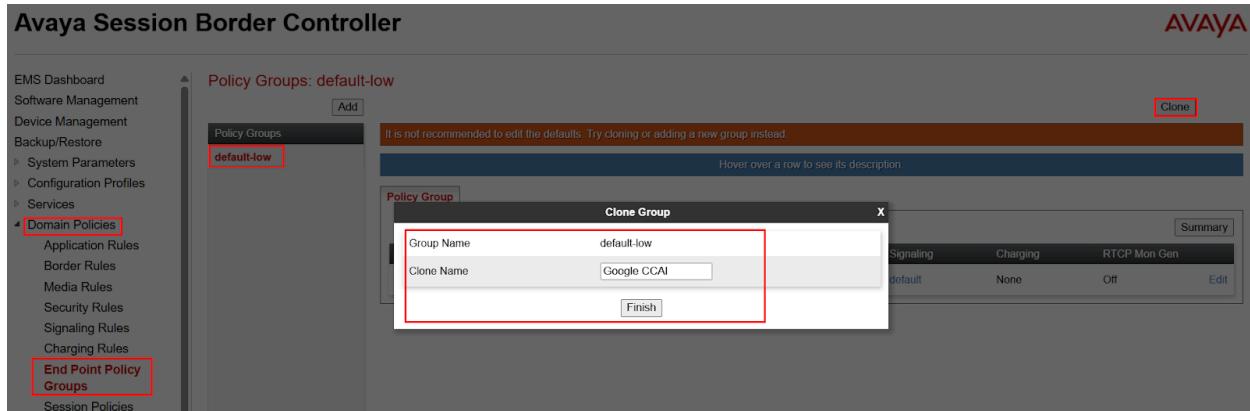


Figure 42: End Point Policy Group for Google CES

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

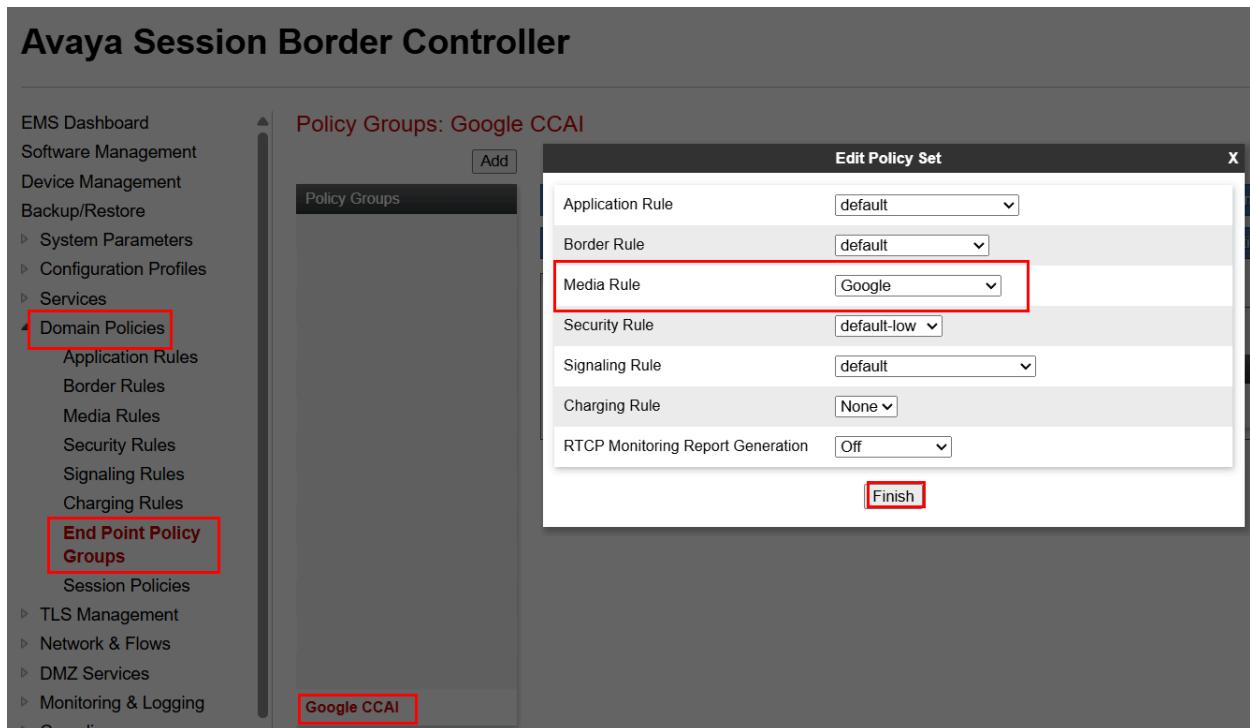


Figure 43: End Point Policy Group for Google CES

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**

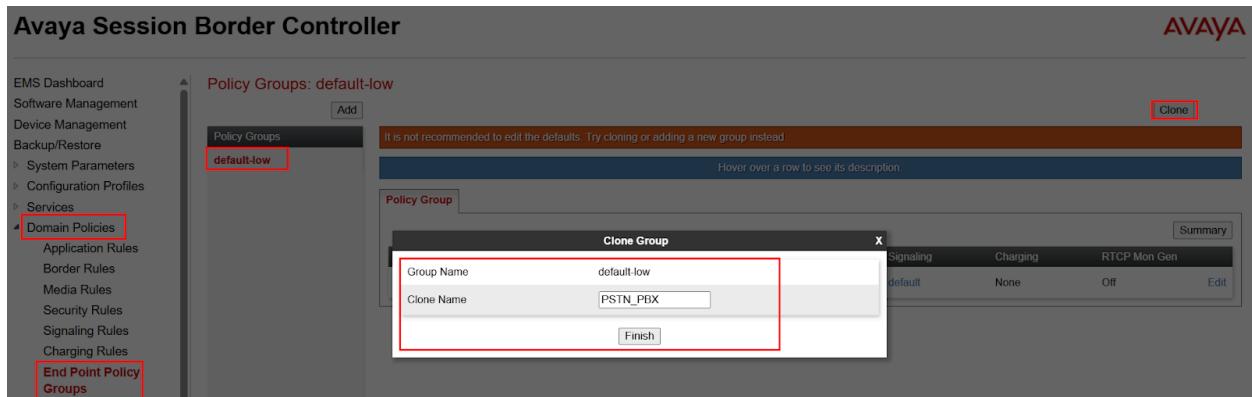


Figure 44: End Point Policy Group for PSTN Gateway and OnPrem PBX

- Click **Finish**

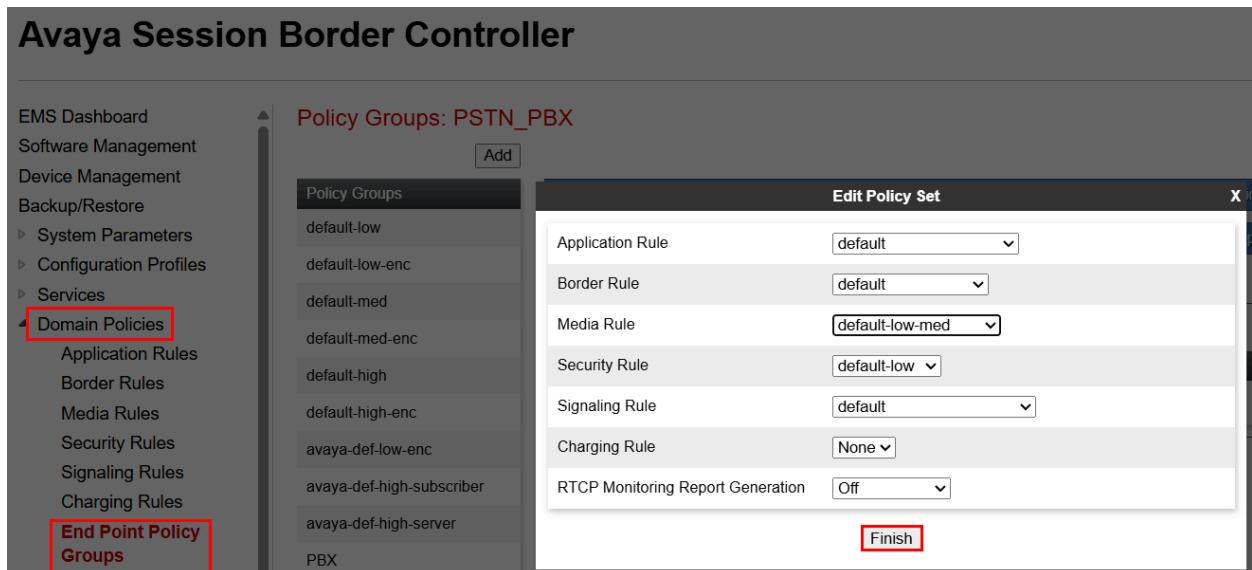


Figure 45: End Point Policy Group for PSTN Gateway and OnPrem PBX (Cont.)

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**

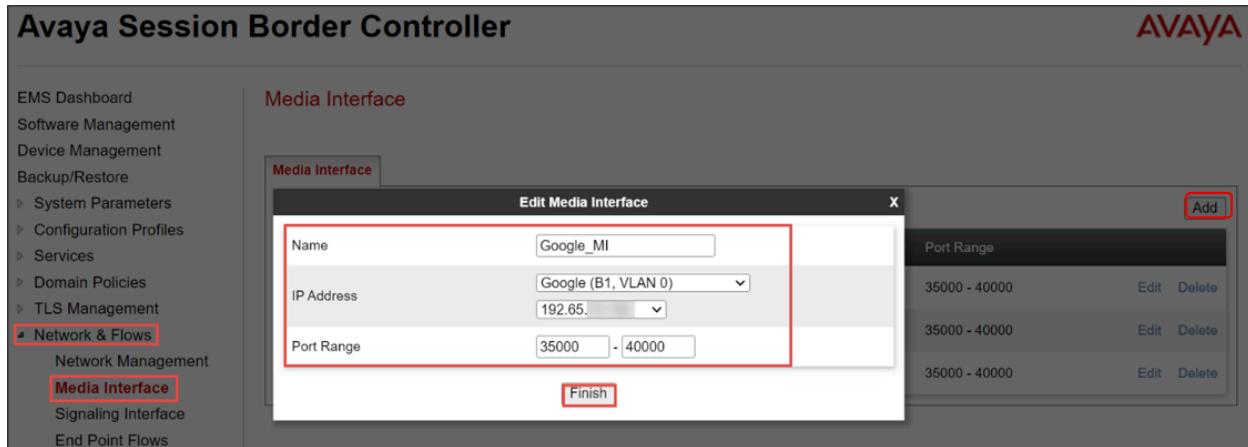


Figure 46: Media Interface Facing Google CES

- 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**

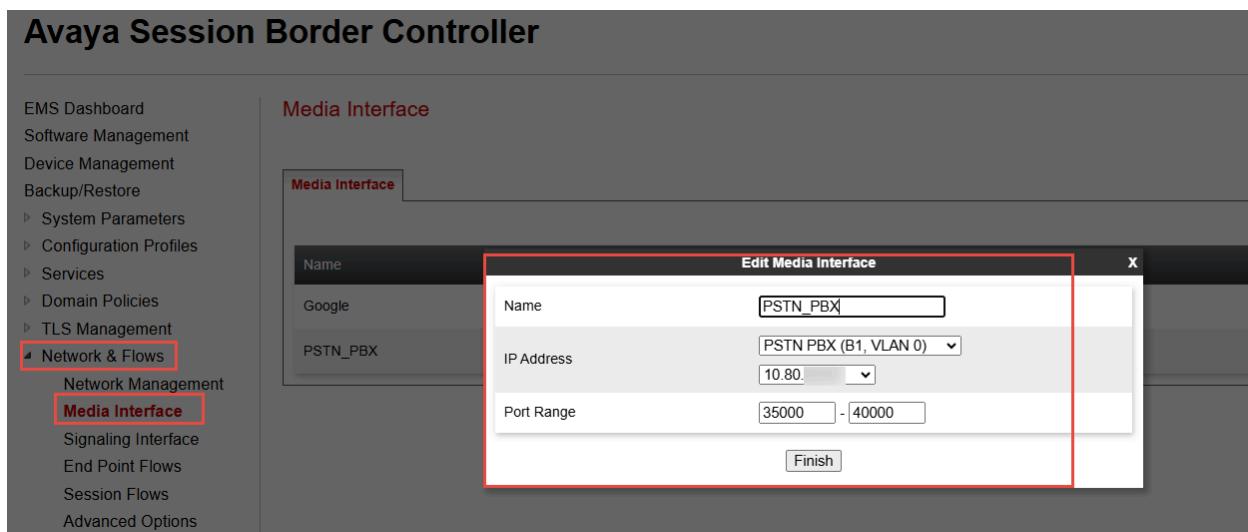


Figure 47: Media Interface Facing PSTN Gateway and OnPrem PBX

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**

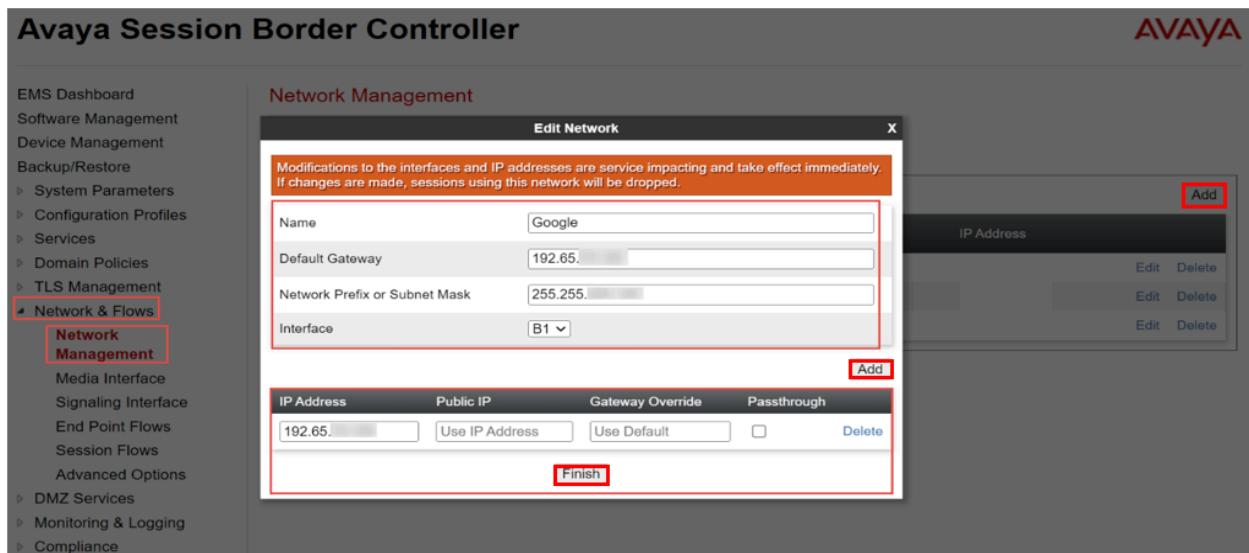


Figure 48: Network Management Facing Google CES

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**

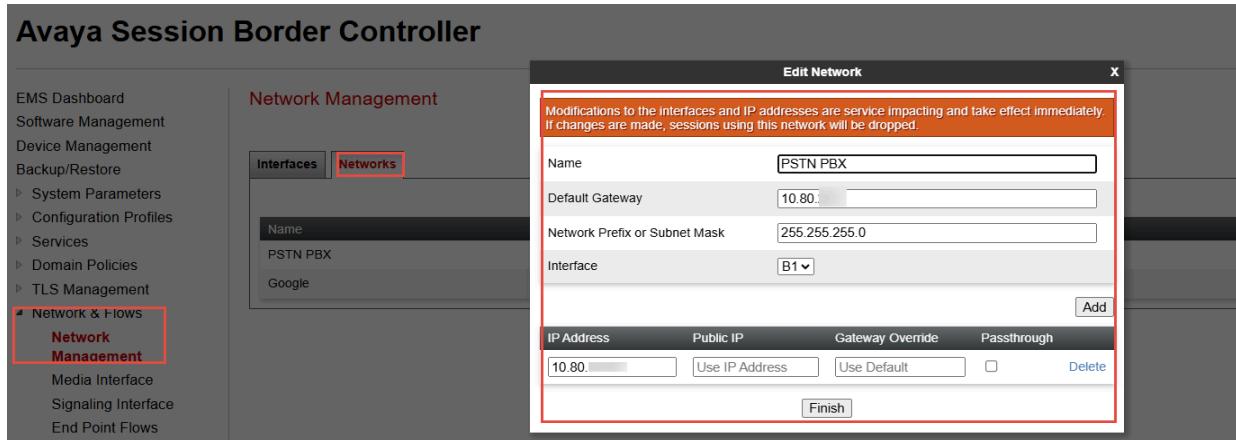


Figure 49: Network Management Facing PSTN Gateway and OnPrem PBX

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**

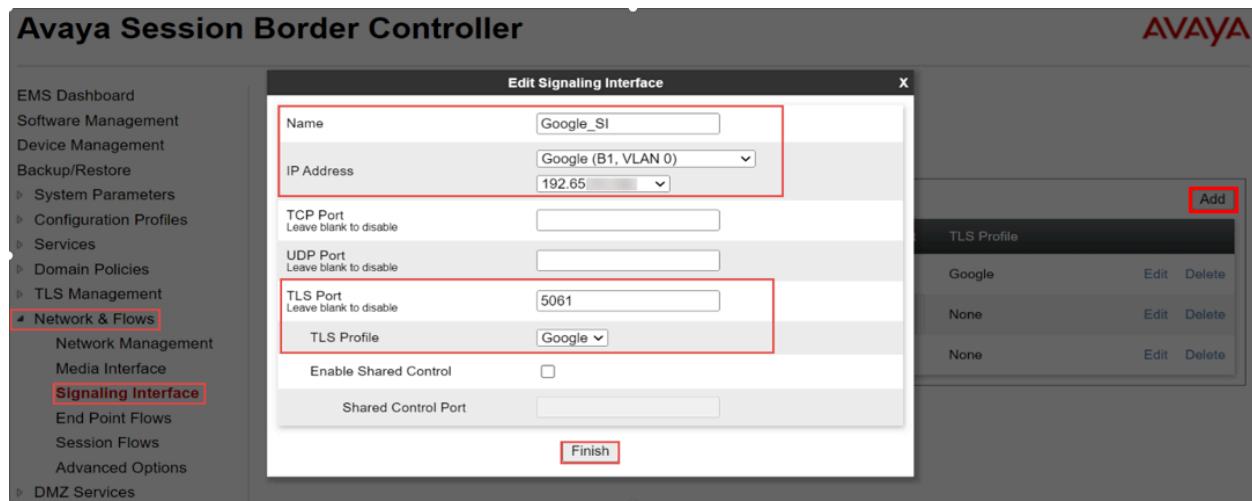


Figure 50: Signaling Interface Facing Google CES

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**

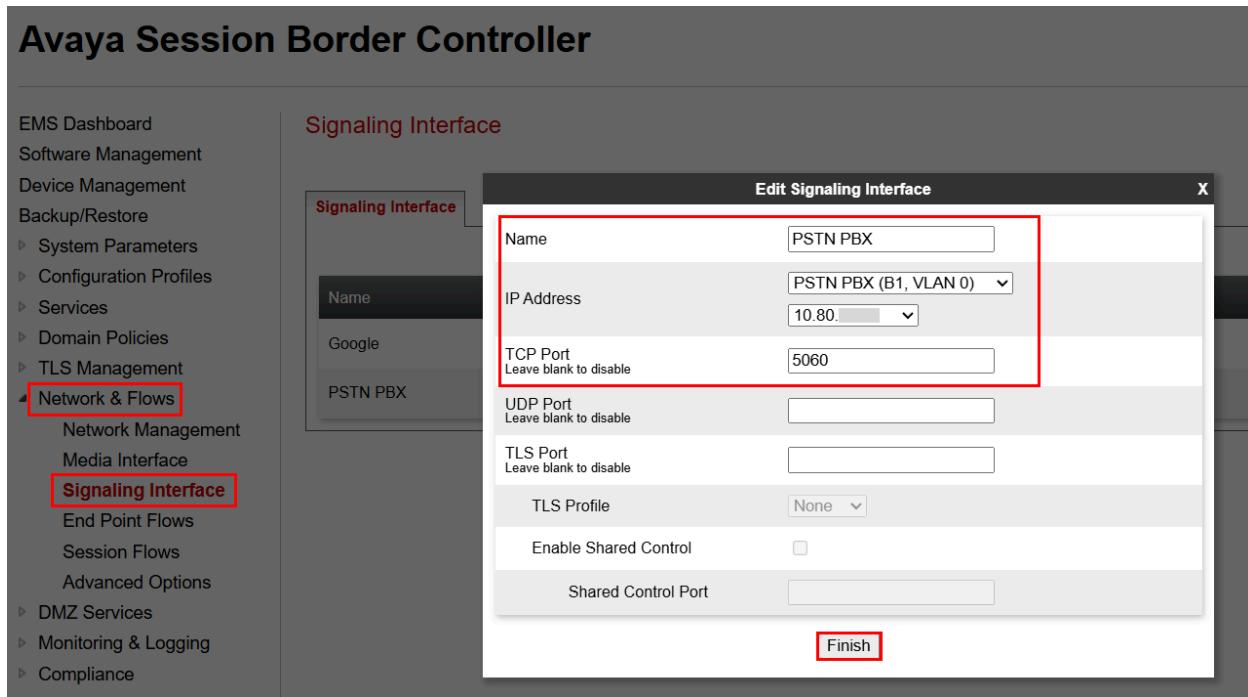


Figure 51: Signaling Interface Facing PSTN Gateway and OnPrem PBX

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**

Avaya Session Border Controller

End Point Flows

Server Flows Server Flows

Modifications made to a Server Flow will only take effect on new sessions.

SIP Server: CUCM

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile	View	Clone	Edit	Delete
1	CUCM	*	Google	PSTN PBX	default-low	CUCM				

SIP Server: Google CCAI

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile	View	Clone	Edit	Delete
1	Google	*	PSTN PBX	Google	Google CCAI	CUCM				

SIP Server: PSTNGW

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile	View	Clone	Edit	Delete
1	PSTN	*	Google	PSTN PBX	default-low	Google CCAI				

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**

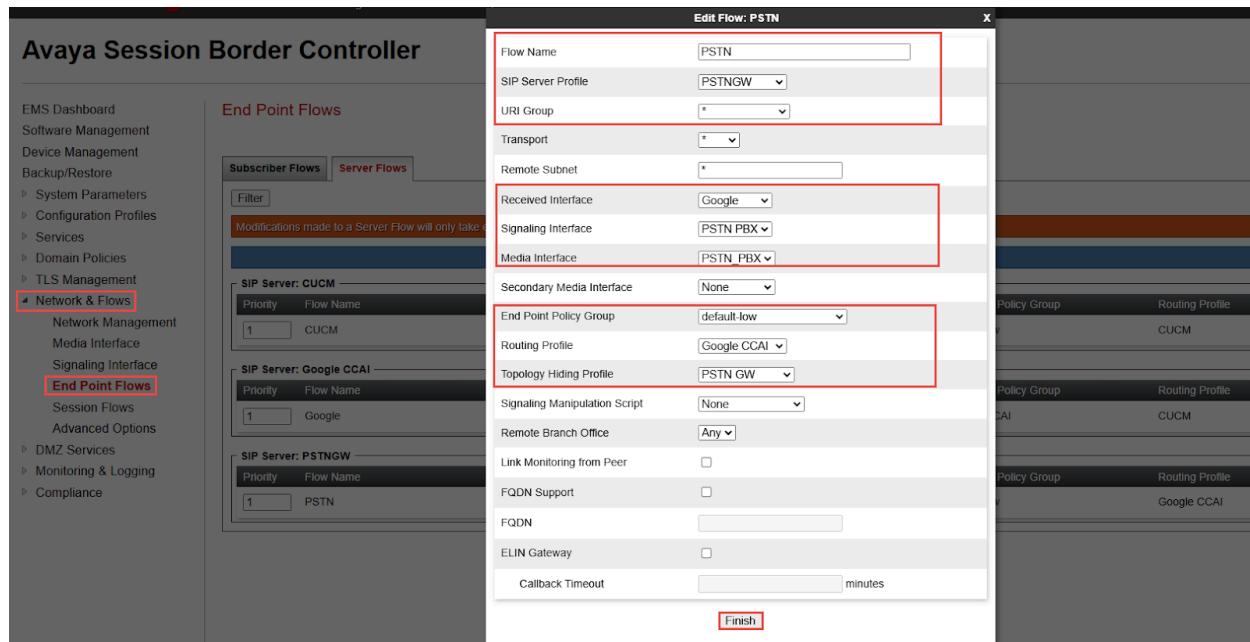


Figure 53: Server Flow for PSTN Gateway

End point flow for Google CES

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

SIP Server: Google CCAI						
Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile
1	Google	*	PSTN PBX	Google	Google CCAI	CUCM

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**

Avaya Session Border Controller

EMS Dashboard

Software Management

Device Management

Backup/Restore

System Parameters

Configuration Profiles

Services

Domain Policies

TLS Management

Network & Flows

Network Management

Media Interface

Signaling Interface

End Point Flows

Subscriber Flows Server Flows

Filter

Modifications made to a Server Flow will only take effect when you click Finish.

SIP Server: CUCM

Priority	Flow Name
1	CUCM

SIP Server: Google CCAI

Priority	Flow Name
1	Google

SIP Server: PSTN

Priority	Flow Name
1	PSTN

Edit Flow: Google

Flow Name: Google

SIP Server Profile: Google CCAI

URI Group: *

Transport: *

Remote Subnet: *

Received Interface: PSTN PBX

Signaling Interface: Google

Media Interface: Google

Secondary Media Interface: None

End Point Policy Group: Google CCAI

Routing Profile: CUCM

Topology Hiding Profile: Google CCAI

Signaling Manipulation Script: Google CCAI_SR

Remote Branch Office: Any

Link Monitoring from Peer:

FQDN Support:

FQDN:

ELIN Gateway:

Callback Timeout: minutes

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**

SIP Server: CUCM						
Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile
1	CUCM	*	Google	PSTN PBX	default-low	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**

The screenshot shows the Avaya SBC EMS interface. On the left, the navigation menu includes 'EMS Dashboard', 'Software Management', 'Device Management', 'Backup/Restore', 'System Parameters', 'Configuration Profiles', 'Services', 'Domain Policies', 'TLS Management', 'Network & Flows' (selected), 'Network Management', 'Media Interface', 'Signaling Interface', 'End Point Flows' (selected), 'Session Flows', 'Advanced Options', 'DMZ Services', 'Monitoring & Logging', and 'Compliance'. The 'Network & Flows' section is expanded, showing 'SIP Server: CUCM', 'SIP Server: Google CCAI', and 'SIP Server: PSTN GW'. The 'SIP Server: CUCM' table has one row with Priority 1 and Flow Name CUCM. The 'Edit Flow: CUCM' dialog box is open, showing fields for 'Flow Name' (CUCM), 'SIP Server Profile' (CUCM), 'URI Group' (*), 'Transport' (dropdown), 'Remote Subnet' (*), 'Received Interface' (Google), 'Signaling Interface' (PSTN PBX), 'Media Interface' (Google), 'Secondary Media Interface' (None), 'End Point Policy Group' (default-low), 'Routing Profile' (CUCM), 'Topology Hiding Profile' (CUCM), 'Signaling Manipulation Script' (CUCM), 'Remote Branch Office' (Any), 'Link Monitoring from Peer' (checkbox), 'FQDN Support' (checkbox), 'FQDN' (text input), 'ELIN Gateway' (checkbox), 'Callback Timeout' (text input), and a 'Finish' button. The 'Edit Flow' dialog box has a red border around the 'Flow Name' and 'SIP Server Profile' fields, and a red box highlights the 'Received Interface' and 'Signaling Interface' fields.

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**

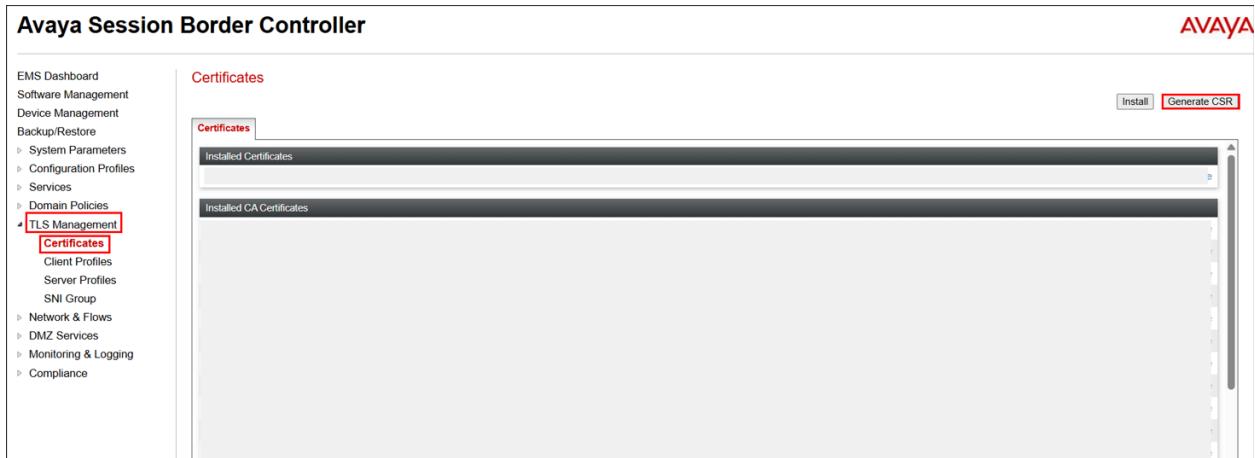


Figure 58: Generate CSR

- 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**

Generate CSR

Country Name	US
State/Province Name	Texas
Locality Name	Plano
Organization Name	Tekvizion
Organizational Unit	lab
Common Name	sbc8.tekvizionlabs.com
Algorithm	<input checked="" type="radio"/> SHA256
Key Size (Modulus Length)	<input checked="" type="radio"/> 2048 bits <input type="radio"/> 4096 bits
Key Usage Extension(s)	<input checked="" type="checkbox"/> Key Encipherment <input checked="" type="checkbox"/> Non-Repudiation <input checked="" type="checkbox"/> Digital Signature
Extended Key Usage	<input checked="" type="checkbox"/> Server Authentication <input checked="" type="checkbox"/> Client Authentication
Subject Alt Name	
Passphrase	
Confirm Passphrase	
Contact Name	
Contact E-Mail	
Generate CSR	

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**

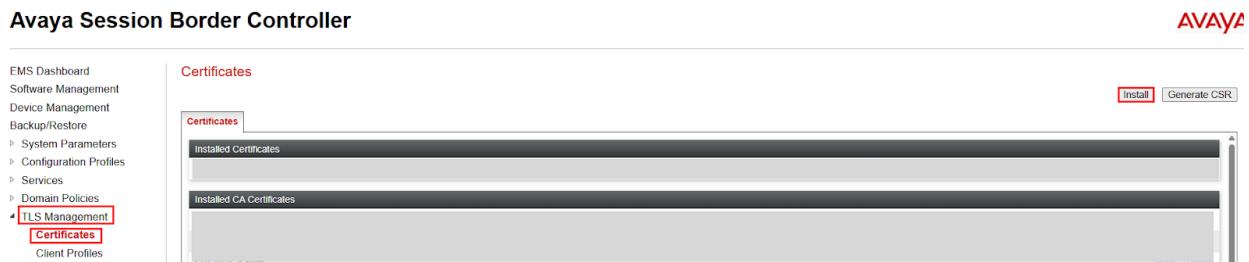


Figure 60: Certificate installation

- 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**

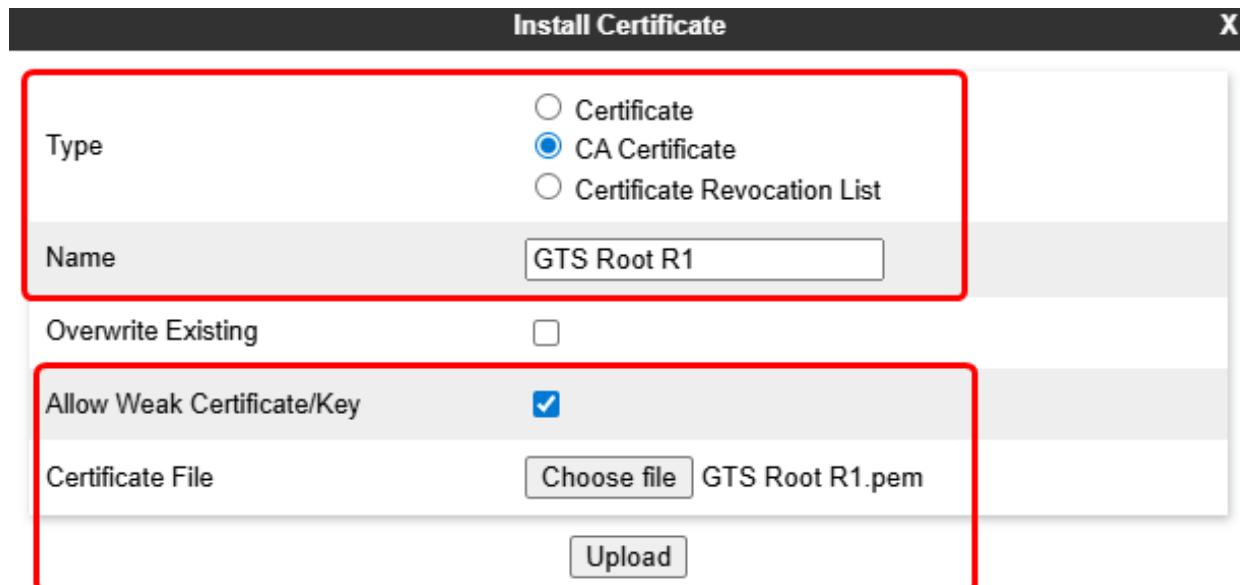


Figure 61: GTS Root R1

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**

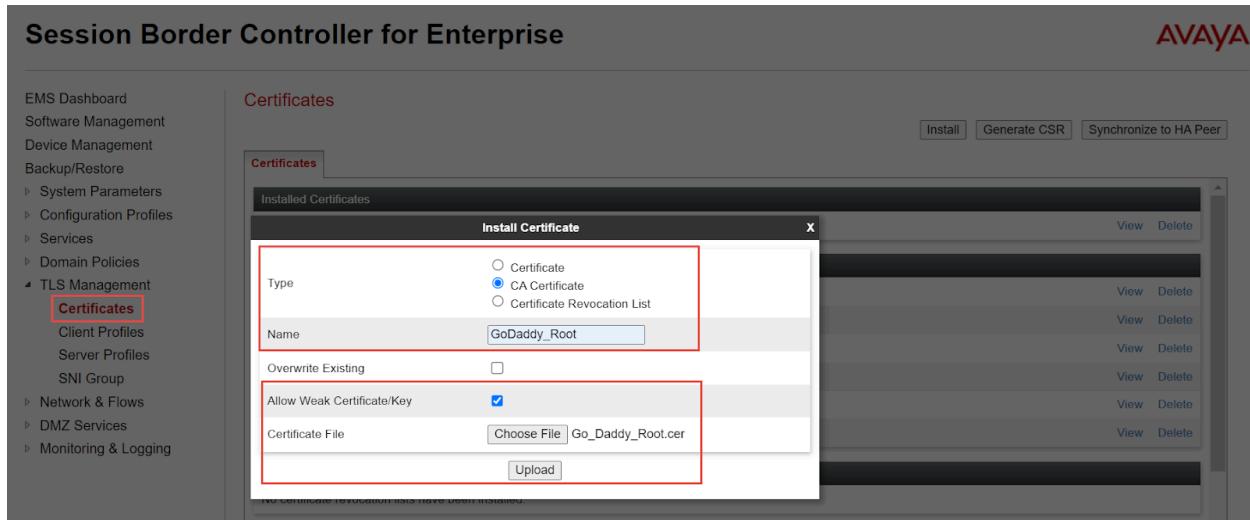


Figure 62: Upload GoDaddy Root CA

- 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**

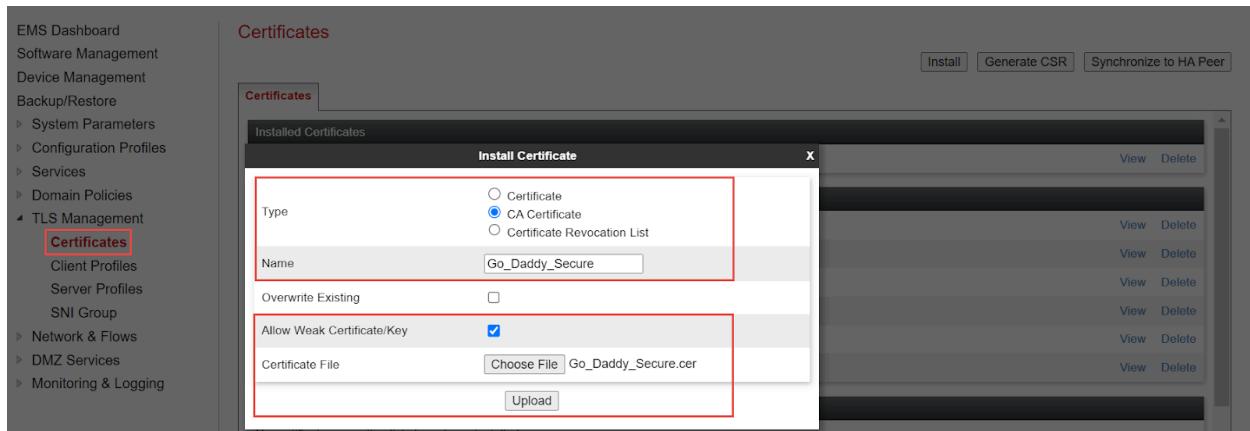


Figure 63: Upload GoDaddy Secure CA

- 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**

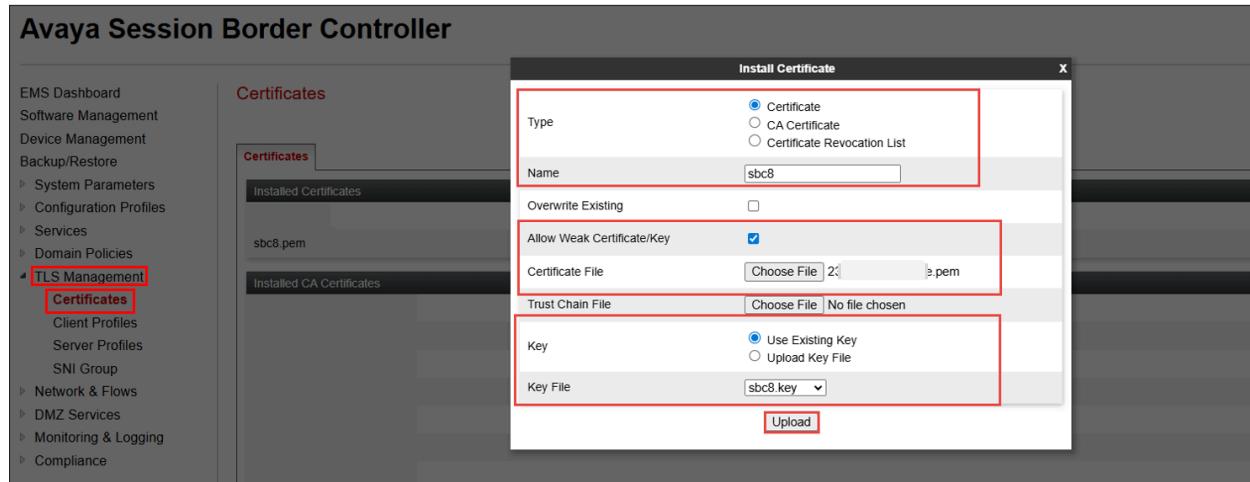


Figure 64: Upload Avaya SBC server Certificate

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**

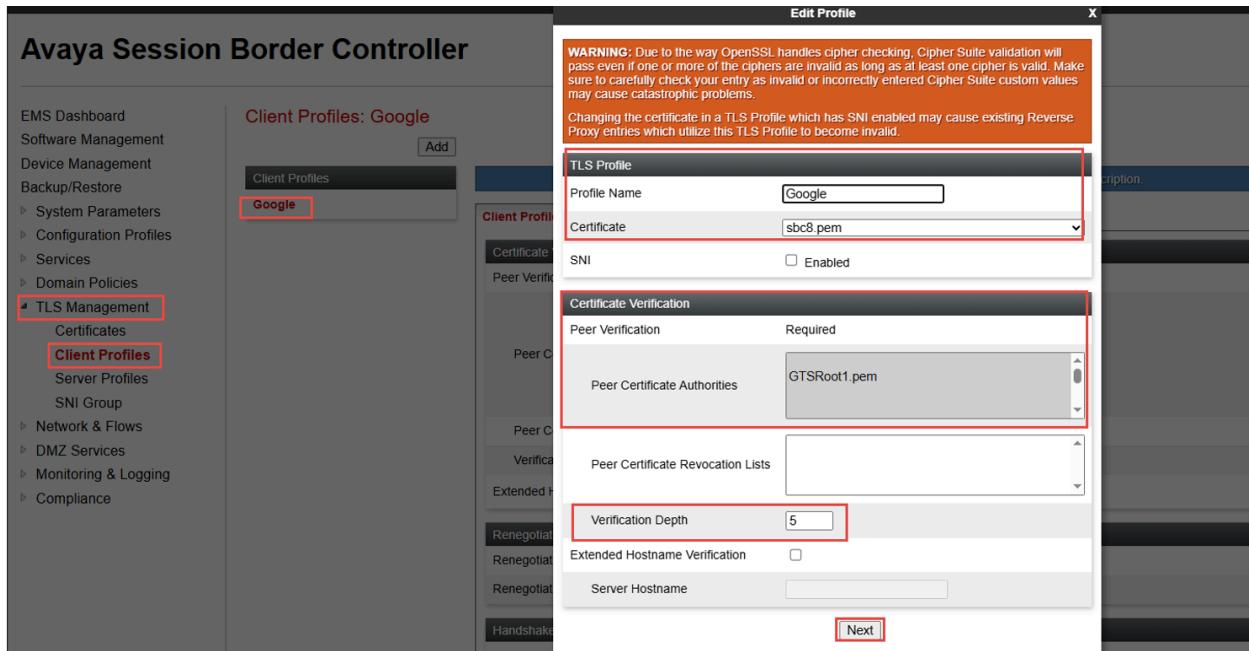


Figure 65: Client Profile Google CES

Server Profile for Google CES

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

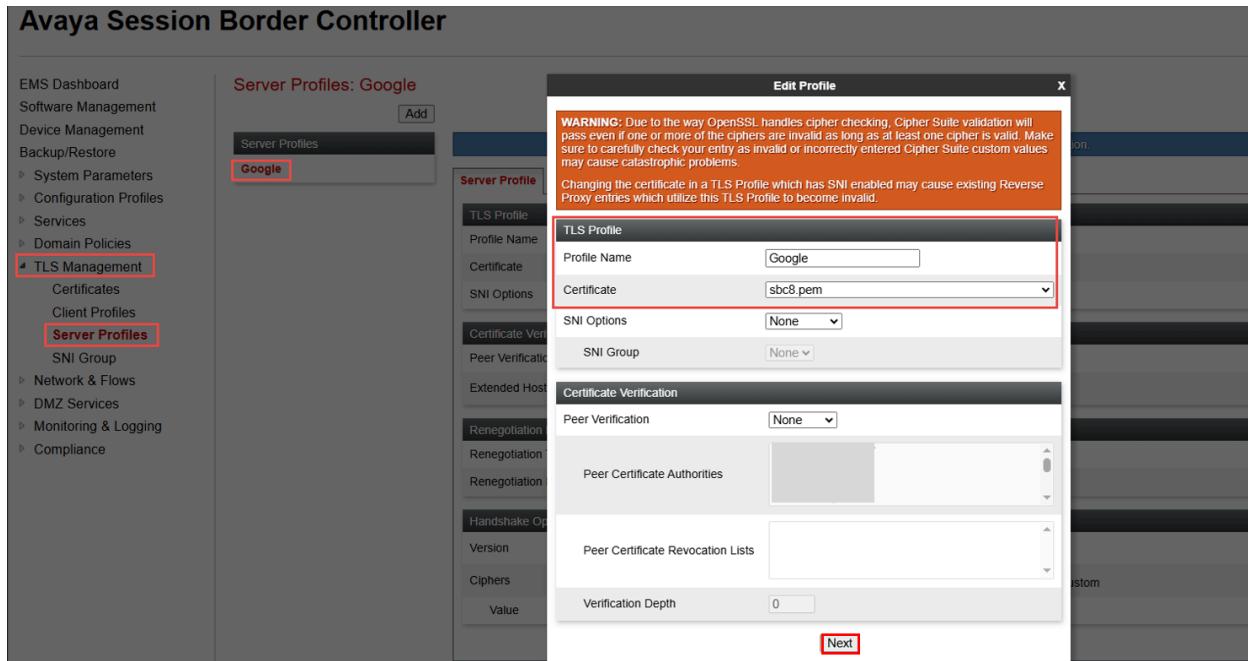


Figure 66: Server Profile towards Google CES

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

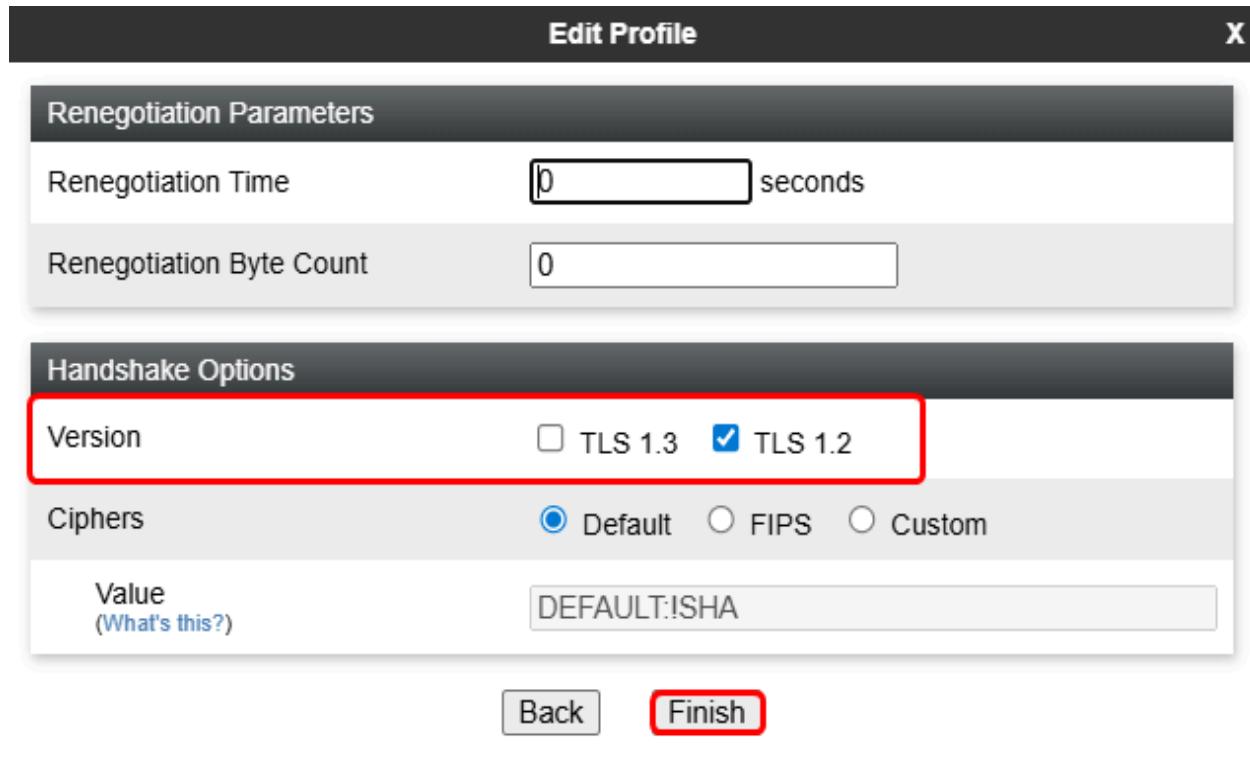


Figure 67: Server Profile towards Google CES (Cont.)

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@192.65.192.65>;tag=981EFC88-230A
To: <sip:+13614@us.telephony.goog:5672;transport=tls>
CSeq: 4360 INVITE
Call-ID: a95d66761d308966e2bbc50433215103
Contact: <sip:192.65.192.65:5061;transport=tls>;sip.src
Record-Route: <sip:192.65.192.65:5061;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 192.65.192.65:5061;branch=z9hg4bK-s1632-000655365869-1-s1632-Expires: 180
Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898/conversations/Sr_a9503>;purpose=Goog-ContactCenter-Conversation
Request: SIP/2.0
Timestamp: 1758101718
Allow-Events: telephone-event
P-Asserted-Identity: "Pradeep Gopal" <sip:2145@192.65.192.65>
Remote-Address: MTAuHjQUMS43MjoxNzI0ToxOjE=
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 10.64.192.65
s=SIP
c=IN IP4 192.65.192.65
t=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+kw52fVf3mEJdGUJomop34a
```

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

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...@192.65.192.65>;tag=BC91BAE8-1784
To: <sip:+131494@us.telephony.goog:5672>
CSeq: 101 INVITE
Call-ID: 2891181e459655649e5e9861e89f3a44
Contact: <sip:21455090180192.65.192.65:5061;transport=tls>
Record-Route: <sip:192.65.192.65:5061;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 192.65.192.65:5061;branch=z9hg4bK-s1632-001707678602-1-s
Via: SIP/2.0/TCP 10.64.192.65:5060;branch=z9hg4bK6BBD237C
Expires: 180
Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898/conversations/Sr_2f144>;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@192.65.192.65>
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 10.64.192.65
s=SIP
c=IN IP4 192.65.192.65
t=0
m=audio 35168 RTP/SAVP 101 0 8 19
c=IN IP4 192.65.192.65
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:7Y5aL7q0jXJfgXt5/4DbumxgJL8zwK/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

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
		<p>+1314944XXXX, "speak to an agent", +1-972-852-XXXX should then ring and get connected as the agent</p>			INVITE was sent from Google to transfer to an agent and call gets connected successfully with both-way audio.
39	UUI headers	<p>Call goes to virtual agent, say "end the call", validate that the SIP BYE has a UUI header</p> <p>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</p>		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	<p>Call goes to virtual agent, say "send a sip refer", validate that a SIP REFER is received to 972-852-XXXX.</p> <p>Make calls to +1314944XXXX, "send a sip refer", SIP REFER should be received with refer to set to 972-852-XXXX</p>		PASSED	REFER request and the transfer are successful.