Configuration Guide for Google CES Call Recording Using Avaya SBC V10.2.1.1-104-25336

TekVizion

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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 Call Recording** 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 Call Recording with Avaya SBC V10.2.1.1-104-25336 configuration.

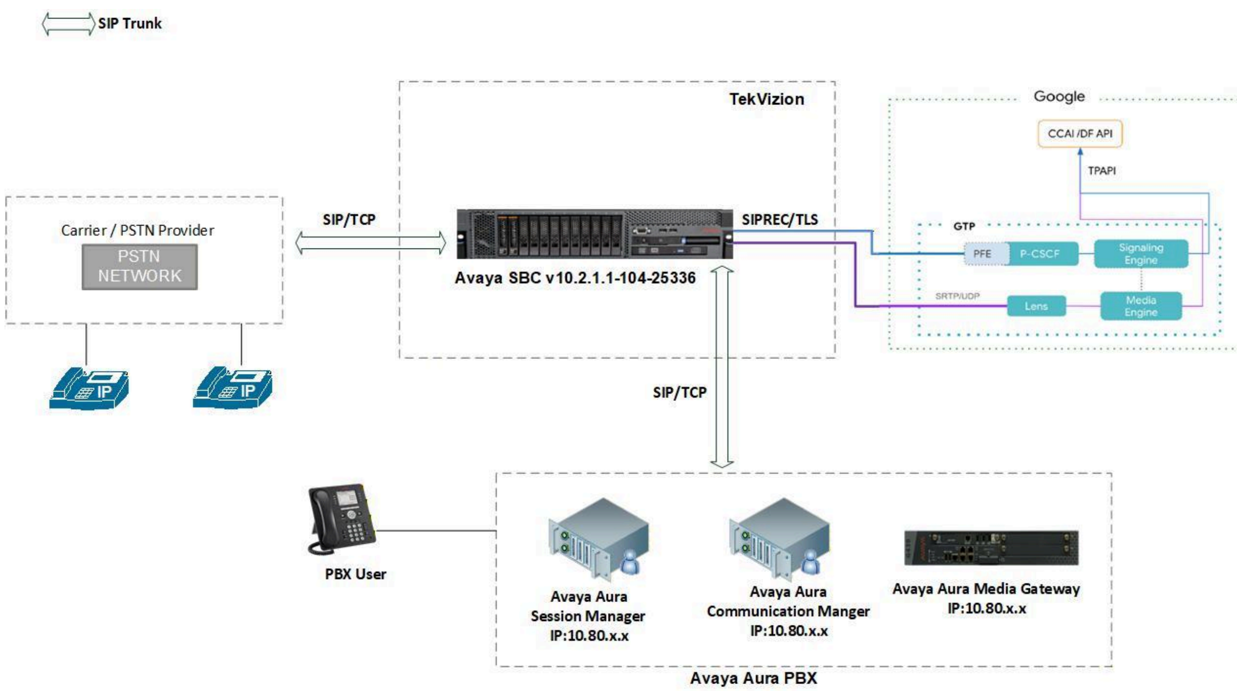


Figure 1: SIP Trunk Lab Reference Network

The lab network consists of the following components:

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

3 Hardware Components

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

4 Software Requirements

- Avaya SBC version: 10.2.1.1-104-25336
- Avaya Aura PBX version: 10.2.1.2.1.229.28376

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
Avaya SBC	10.2.0.0-86-24077
Avaya SBC	8.1.3.2-38-22279

6 Features

6.1 Features Tested for Google CES Call Recording

- Basic Inbound calls
- Call Hold and Resume
- Call Transfer
- Conference

6.2 Features Not Tested for Google CES Call Recording

- None

6.3 Caveats and Limitations

DTLS	Avaya SBC does not support DTLS
Blind Transfer	Avaya PBX does not support blind transfer. This test case is performed by ringing transfer
Long duration call	Avaya SBC does not send refresh session re-INVITE. Google CES sends session refresh every 60 minutes using re-INVITE

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	Recording Profile	Section 7.4.6
Step 7	Session Policies	Section 7.4.7
Step 8	Session Flows	Section 7.4.8
Step 9	Signaling Manipulation	Section 7.4.9
Step 10	Media Rules	Section 7.4.10
Step 11	Signaling Rules	Section 7.4.11
Step 12	End Point Policy Groups	Section 7.4.12
Step 13	Media Interface	Section 7.4.13
Step 14	Network Management	Section 7.4.14
Step 15	Signaling Interface	Section 7.4.15
Step 16	End Point Flow	Section 7.4.16
Step 17	TLS Configuration	Section 7.4.17

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	10.64.X.X, 10.70.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 Call recording.

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 call recording.

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

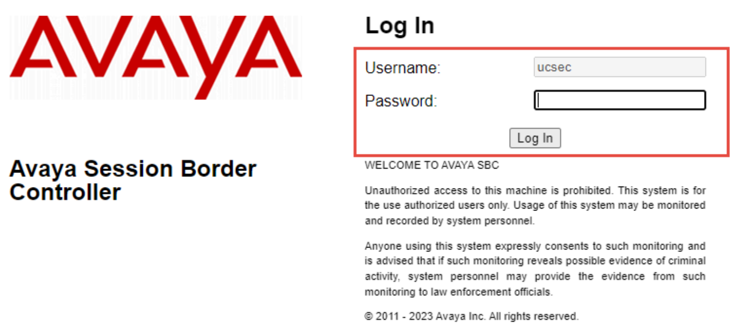


Figure 2: Avaya SBC Login

- 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.

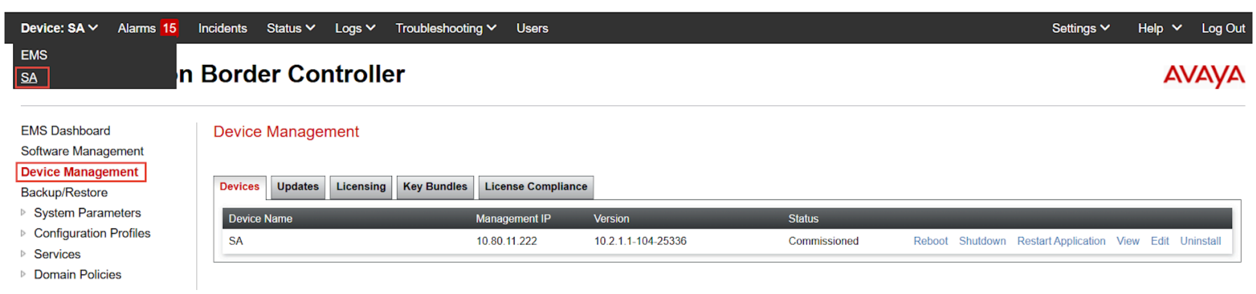


Figure 3: Selection of Avaya SBC Device

7.4.2 Server Interworking

Server Interworking for Avaya Aura Session Manager (SM)

- Navigate: **Configuration Profiles □ Server Interworking**
- Select the default Interworking Profile avaya-ru, click **Clone**
- Set Clone Name: **AvayaSM10.2**
- Click **Finish**

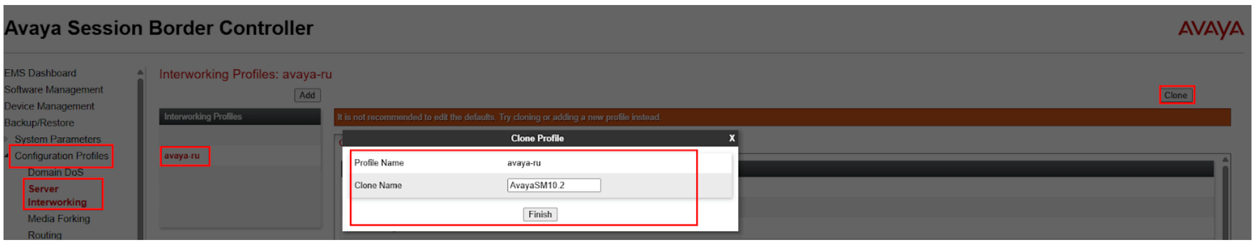


Figure 4: Server Interworking Profile for Avaya Aura SM

- Click **Finish**

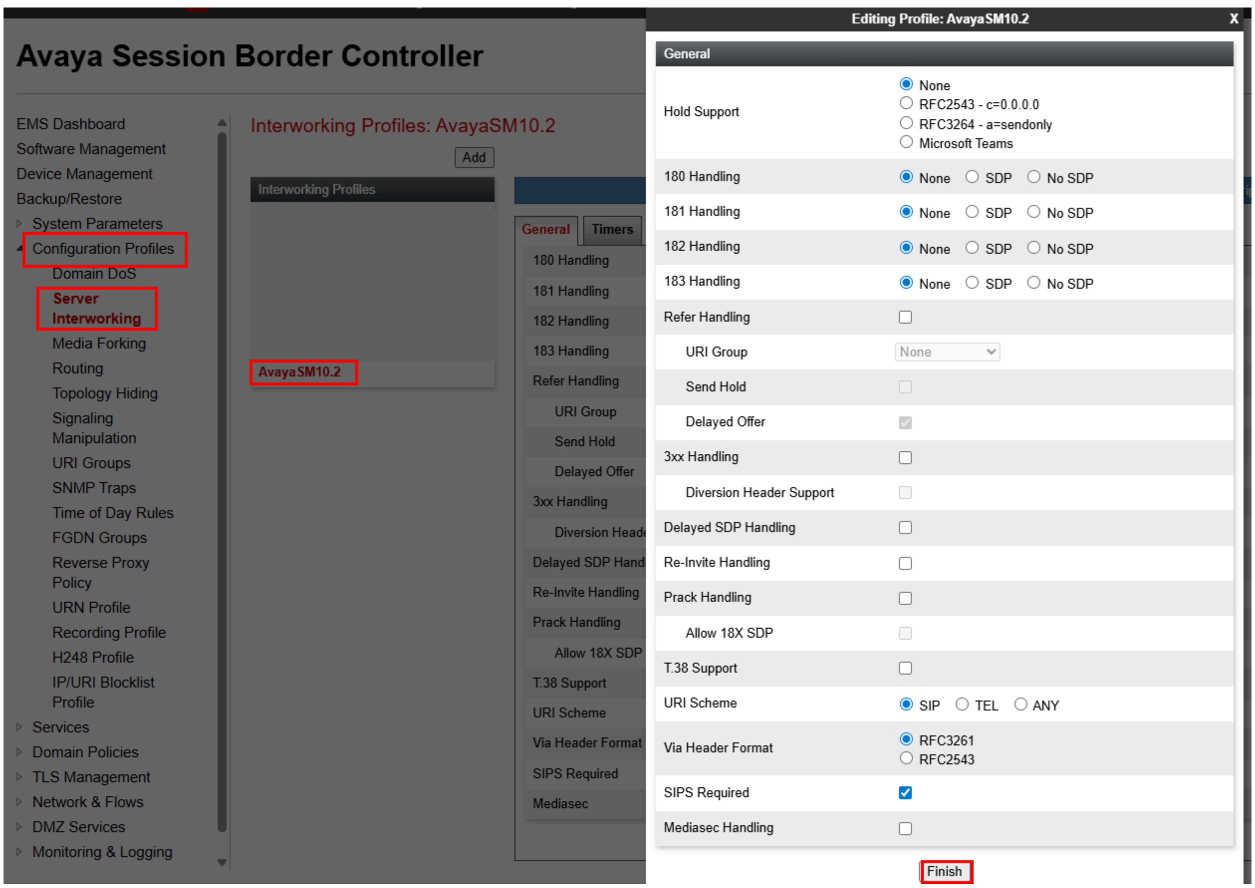


Figure 5: Server Interworking Profile for Avaya Aura SM (Cont.)

- Set Extensions: **Avaya**
- Click **Finish**

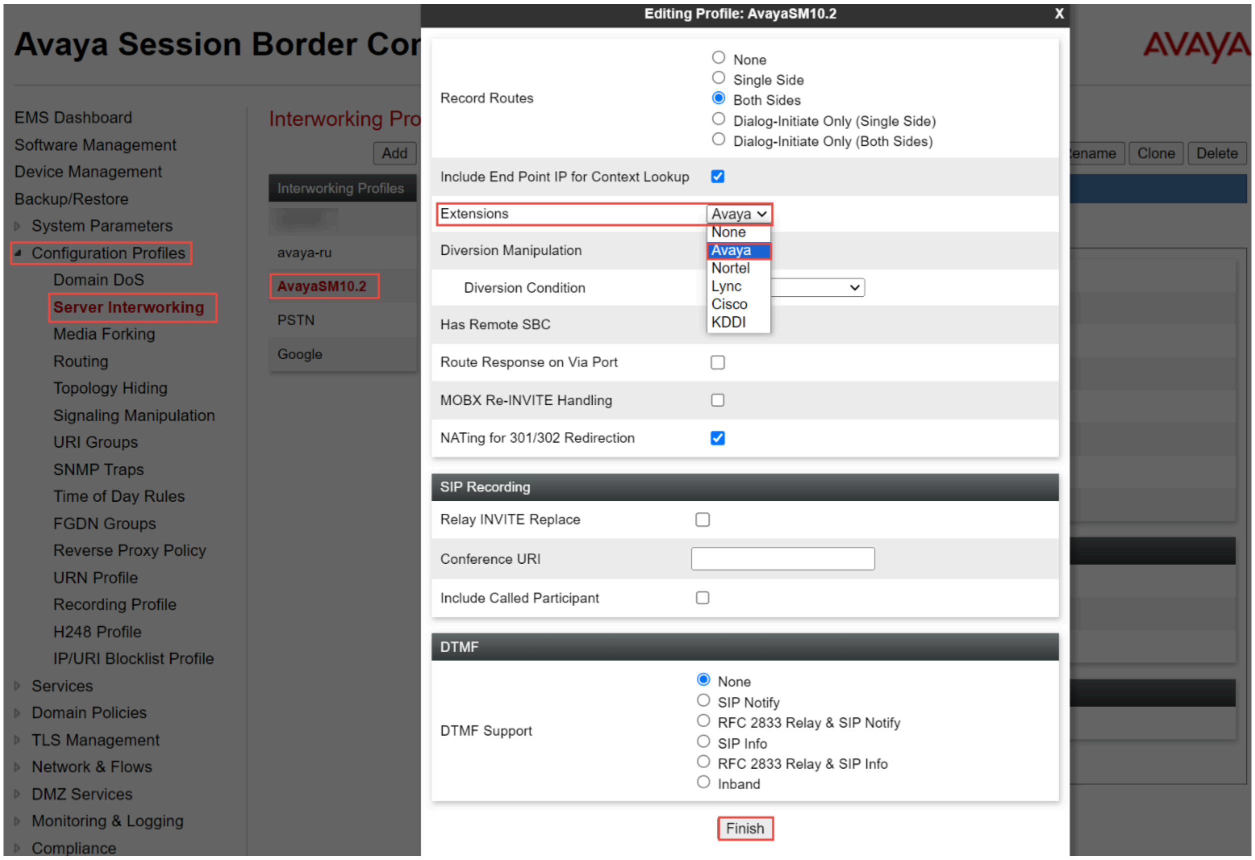


Figure 6: Server Interworking Profile for Avaya Aura SM (Cont.)

Server Interworking for **Google CES**

- Repeat the same procedure to create the Interworking Profile towards Google CES.
- SIPS required: **Unchecked**

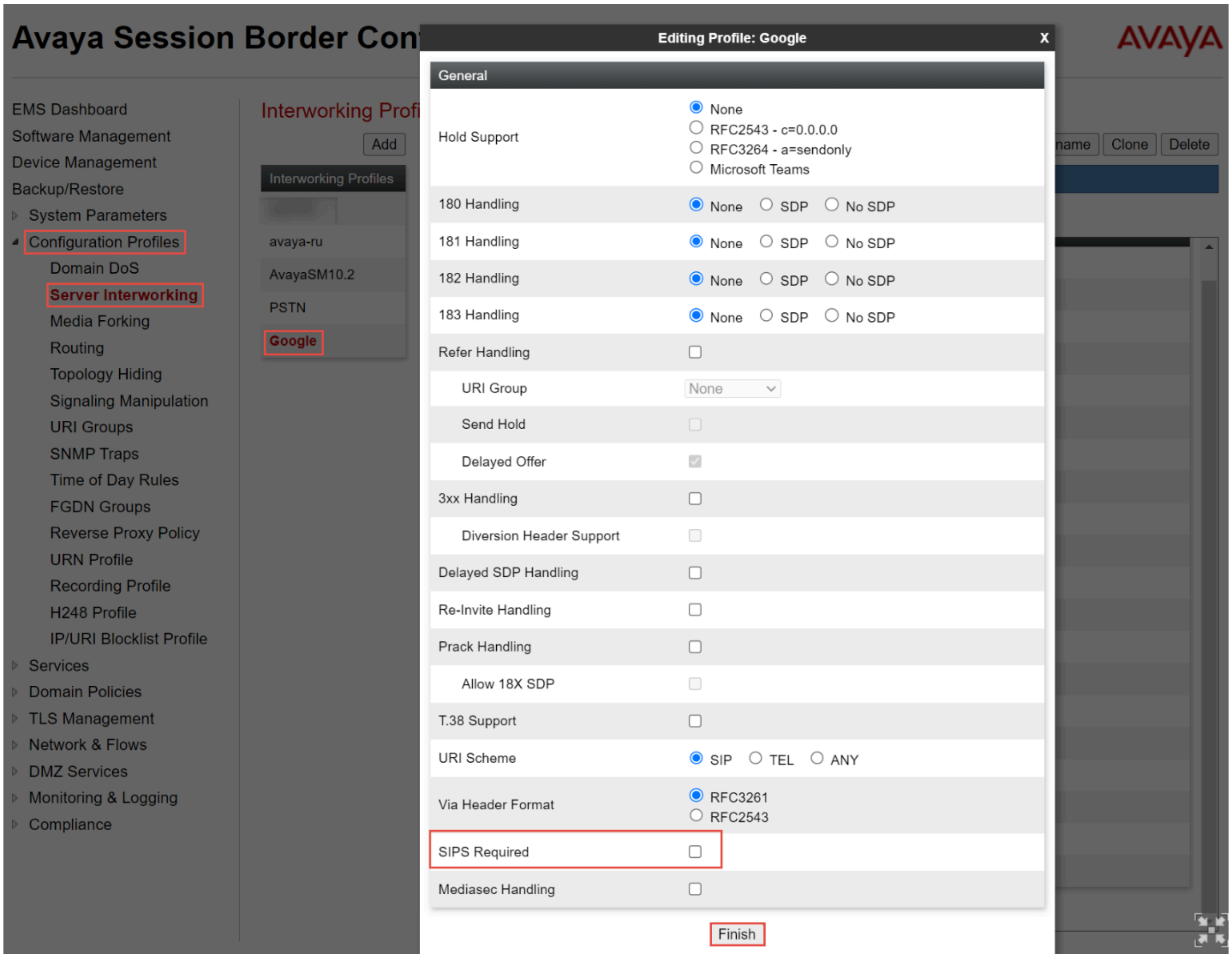


Figure 7: Server Interworking Profile for Google CES

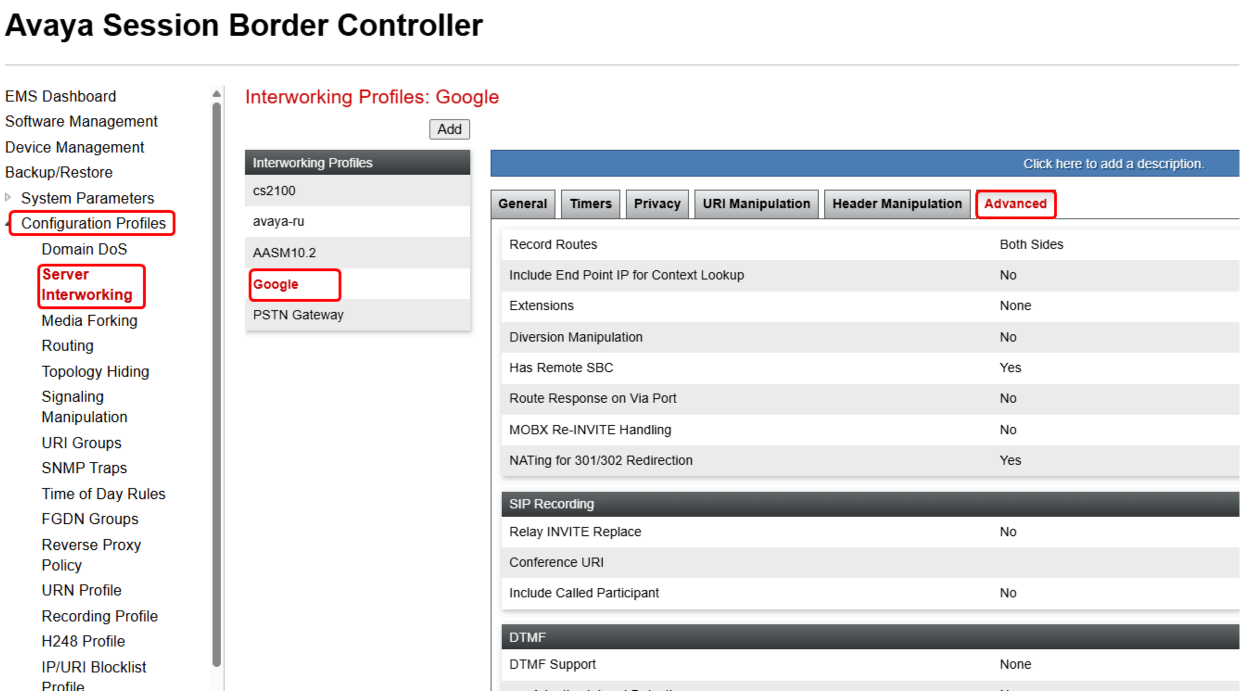


Figure 8: Server Interworking Profile for Google CES (Cont.)

Server Interworking for **PSTN Gateway**

- Repeat the same procedure to create the Interworking Profile towards **PSTN Gateway**

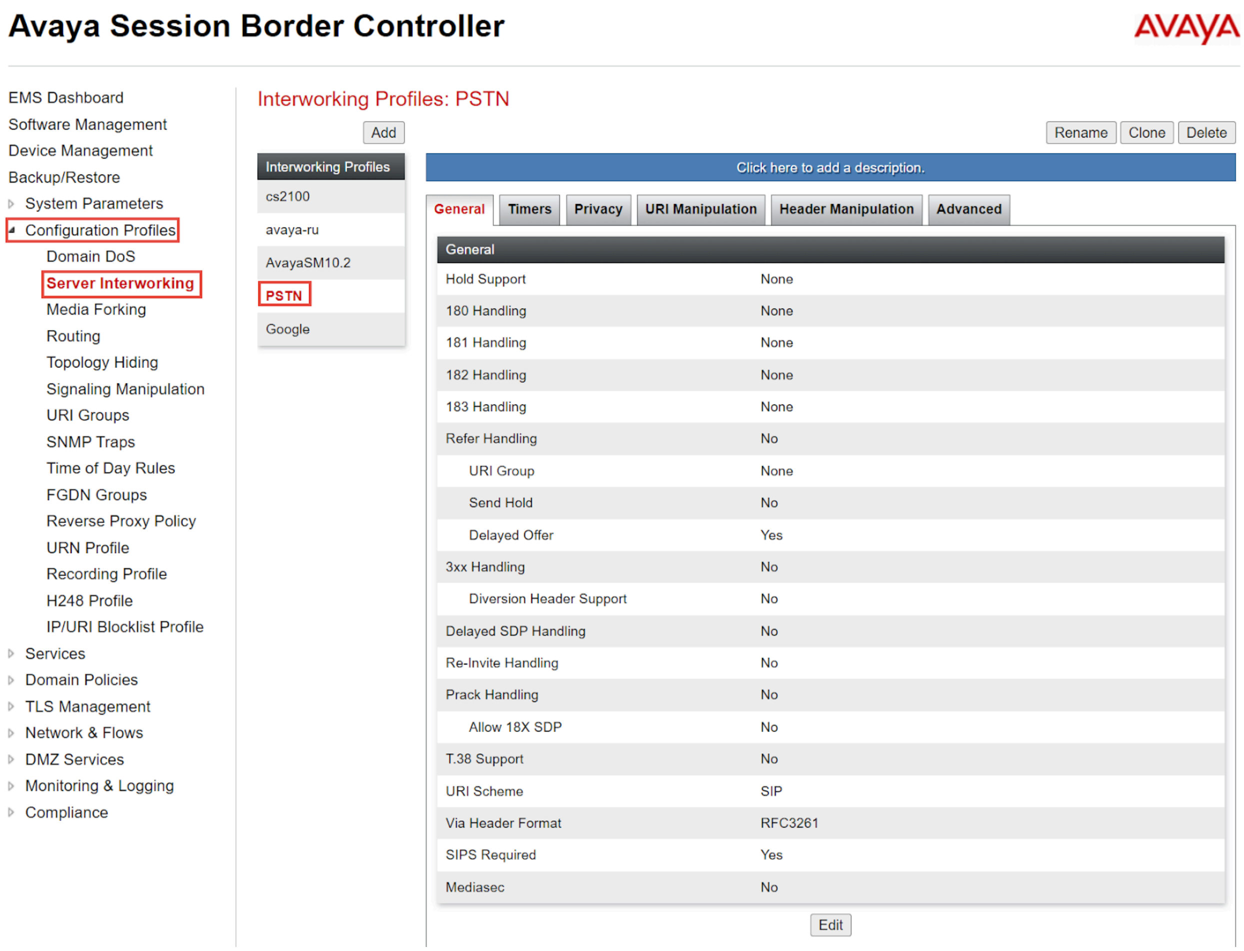


Figure 9: Server Interworking Profile for PSTN Gateway

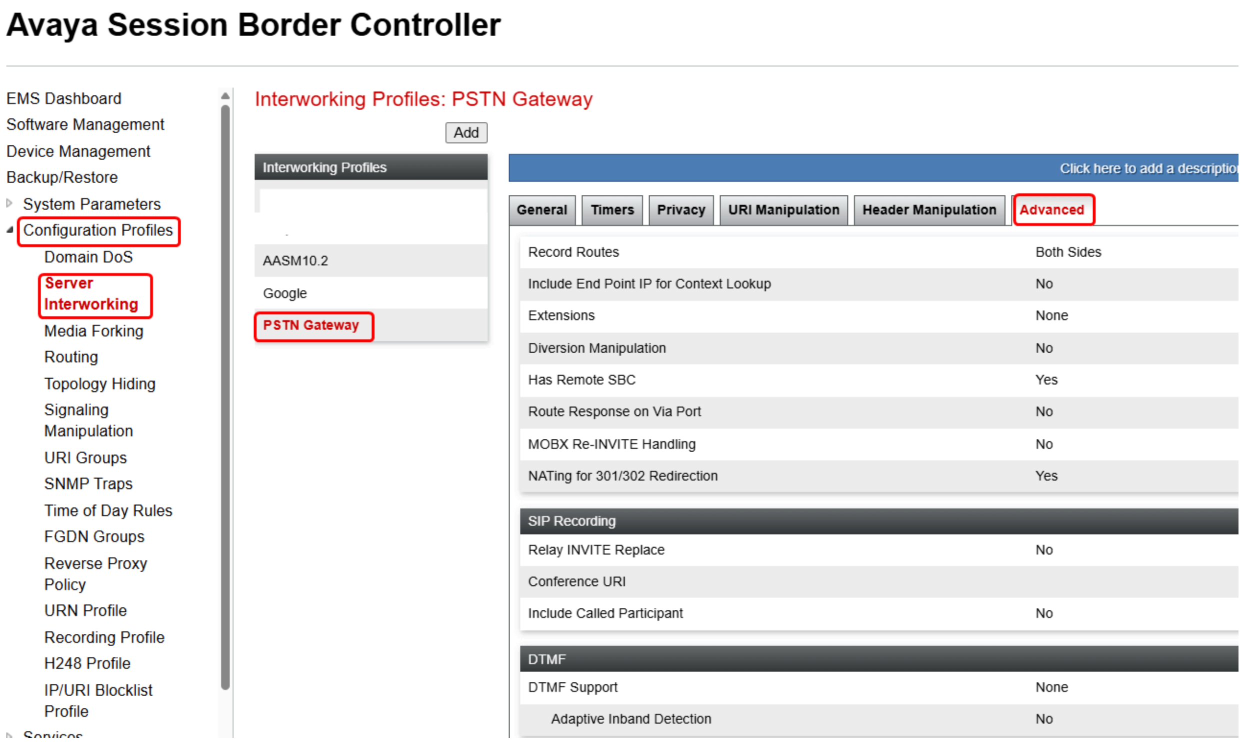


Figure 10: Server Interworking Profile for PSTN Gateway (Cont.)

7.4.3 SIP Servers

SIP Server for **Avaya Aura SM**

- Navigate: **Services** □ **SIP Servers**
- Click **Add**
- Set Profile Name: **AvayaSM10.2**
- Click **Next**

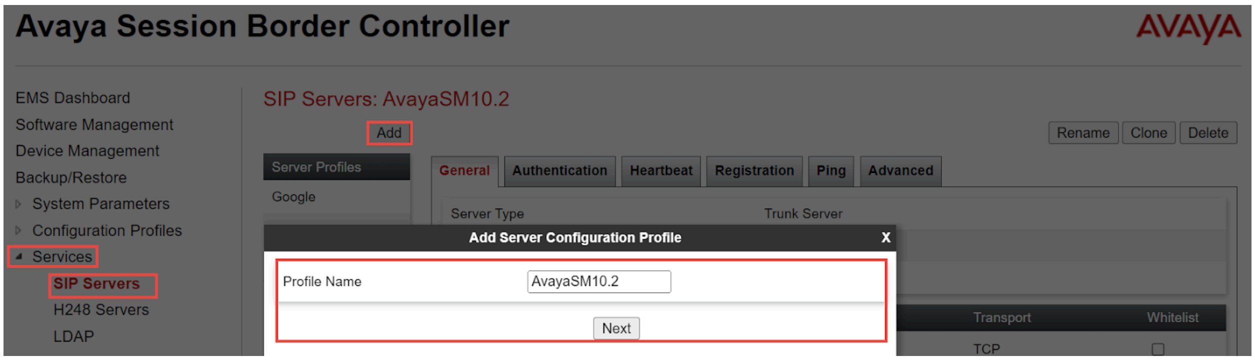


Figure 11: SIP Server for Avaya Aura SM

- 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**

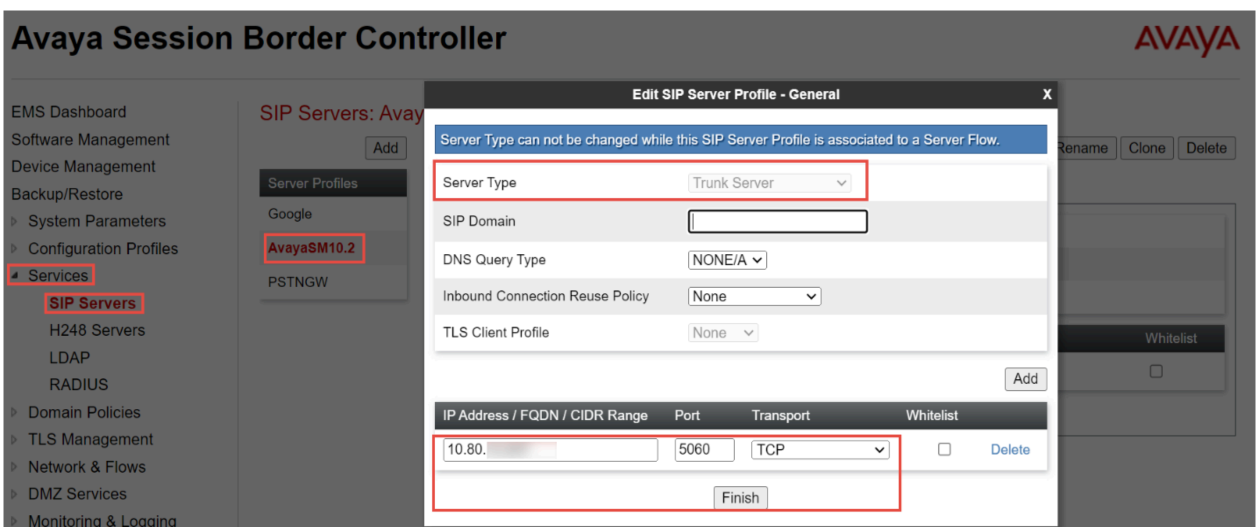


Figure 12: SIP Server for Avaya Aura SM (Cont.)

- 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@10.70.X.X**
- Set To URI: **ping@10.80.X.X**
- Click **Finish**

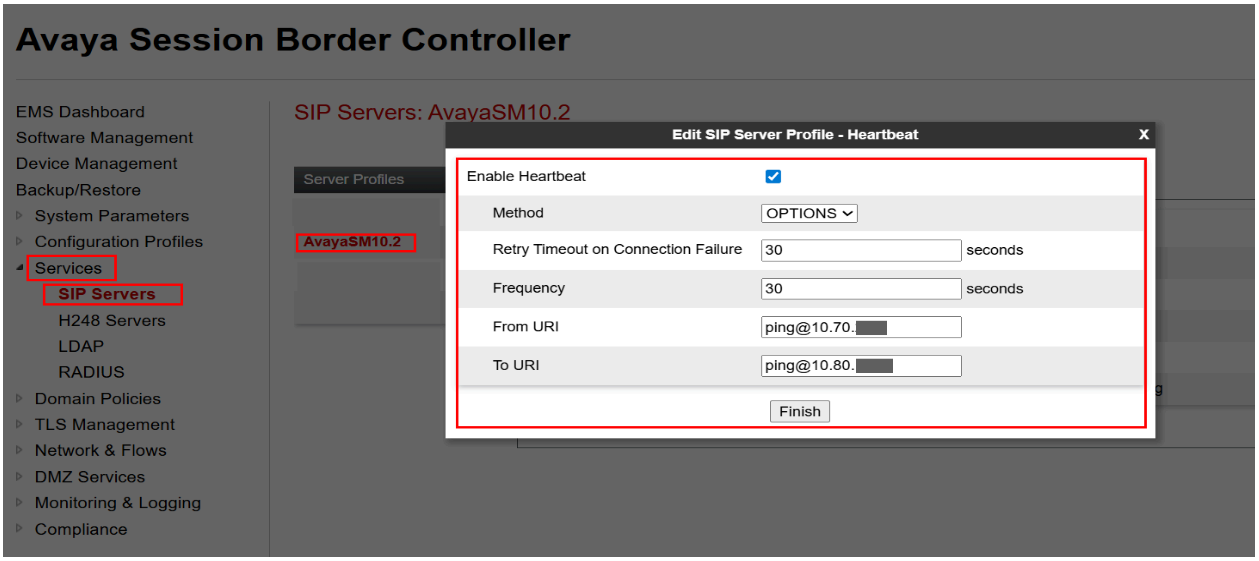


Figure 13: SIP Server for Avaya Aura SM (Cont.)

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

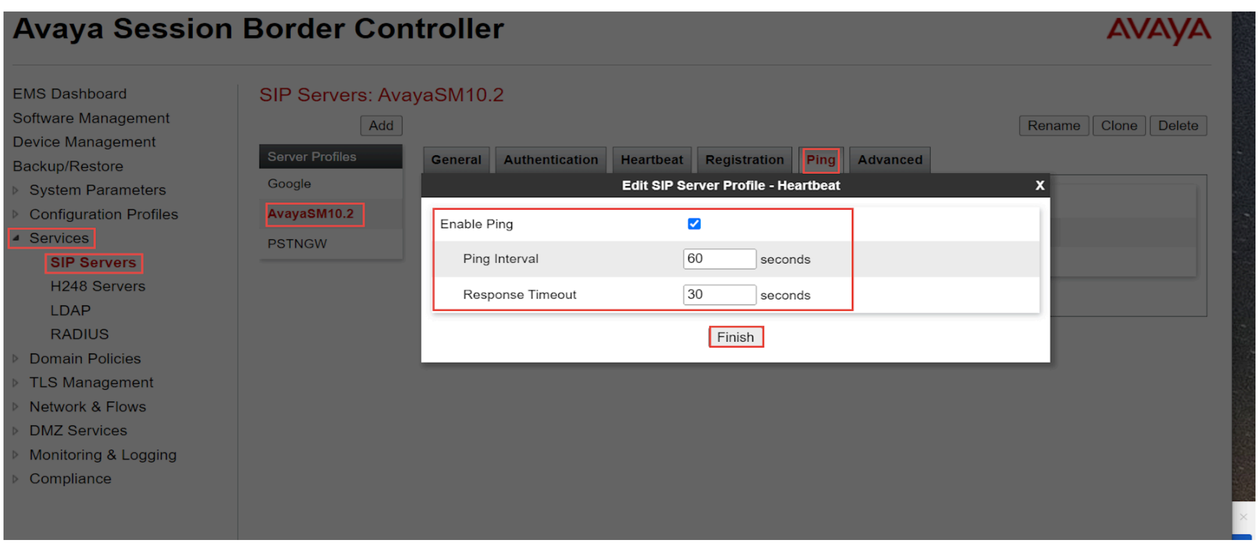


Figure 14: SIP Server for Avaya Aura SM (Cont.)

- Navigate: **Advanced** tab
- Set Enable Grooming: **Checked**
- Set Interworking Profile: Select **AvayaSM10.2,** Refer Section 7.4.2

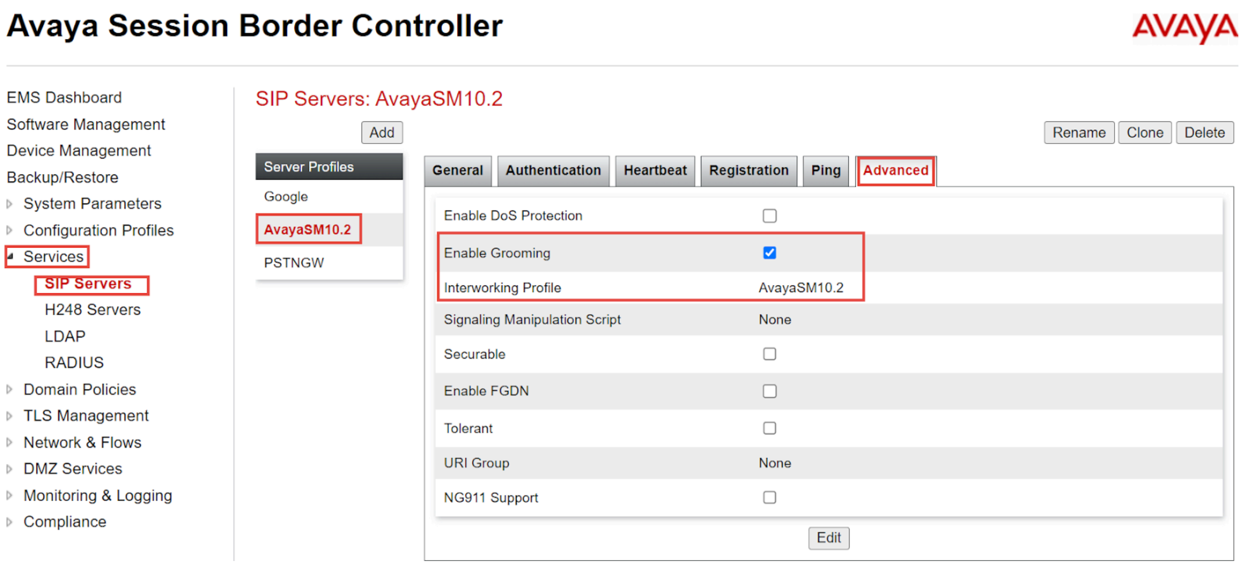


Figure 15: SIP Server for Avaya Aura SM (Cont.)

SIP Server for Google CES

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

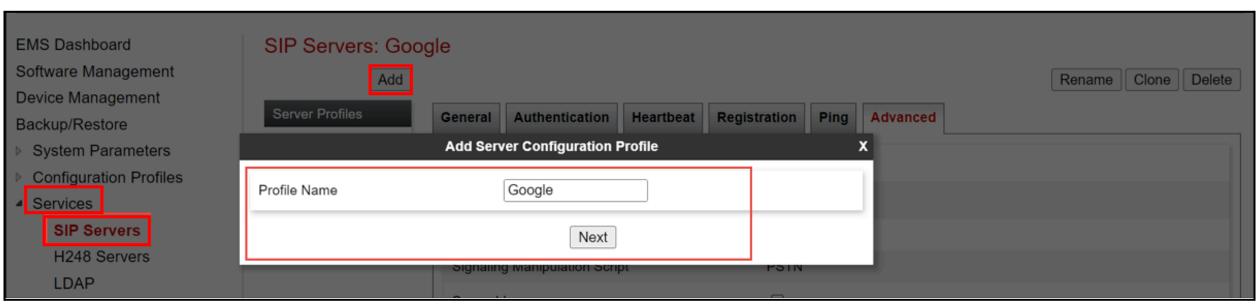


Figure 16: SIP Server for Google CES

- Set Server Type: Select **Recording Server** from the drop down
- Set TLS Client Profile: **Google.** Refer Section 7.4.17
- Set IP Address/FQDN: **us.telephony.goog**
- Set Port: **5672**
- Set Transport: **TLS**
- Click **Finish**

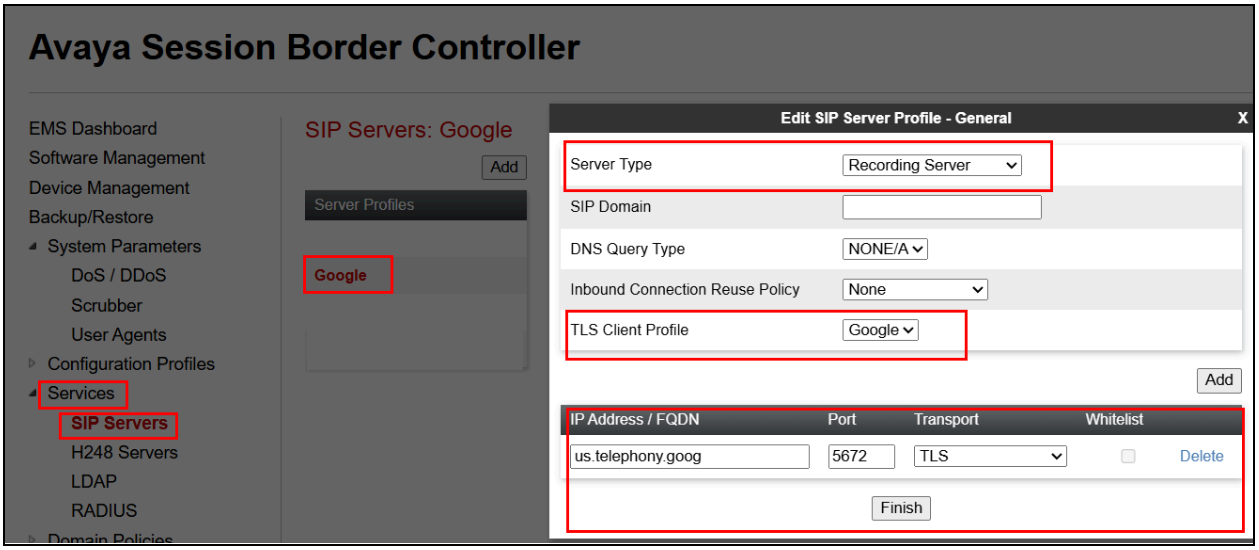


Figure 17: SIP Server for Google CES (Cont.)

- 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**

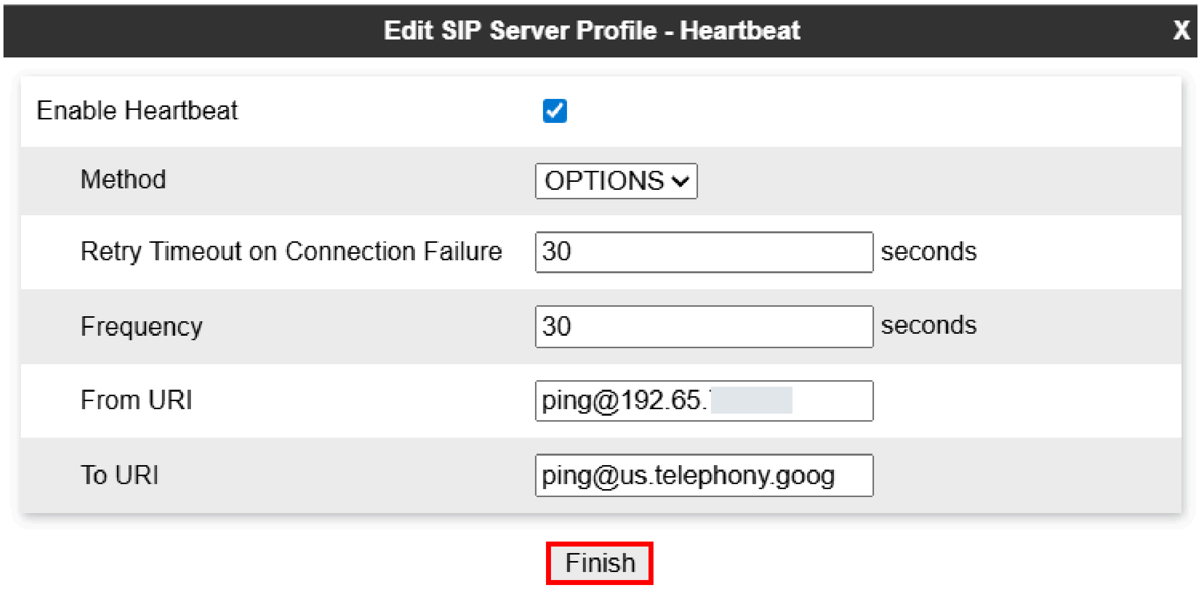


Figure 18: SIP Server for Google CES (Cont.)

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

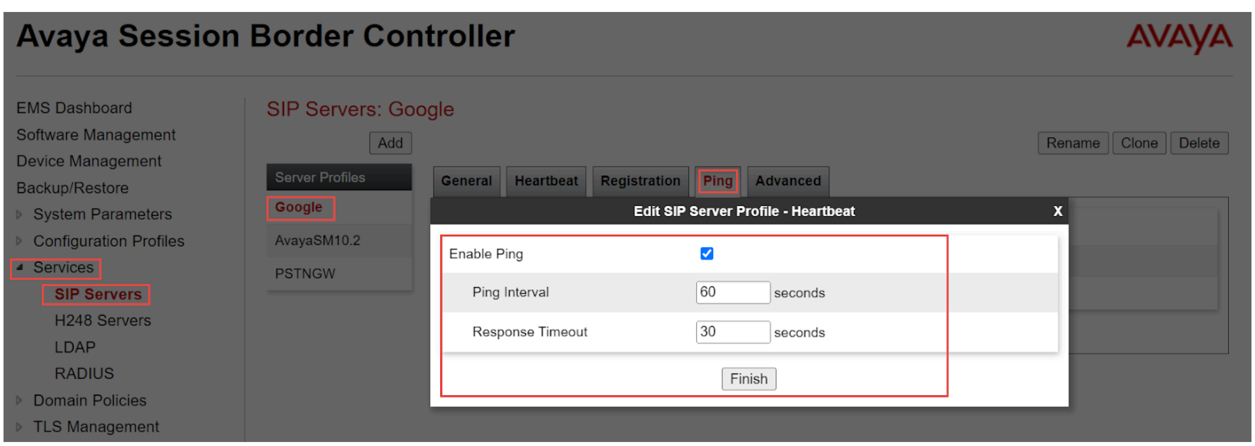


Figure 19: SIP Server for Google CES (Cont.)

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

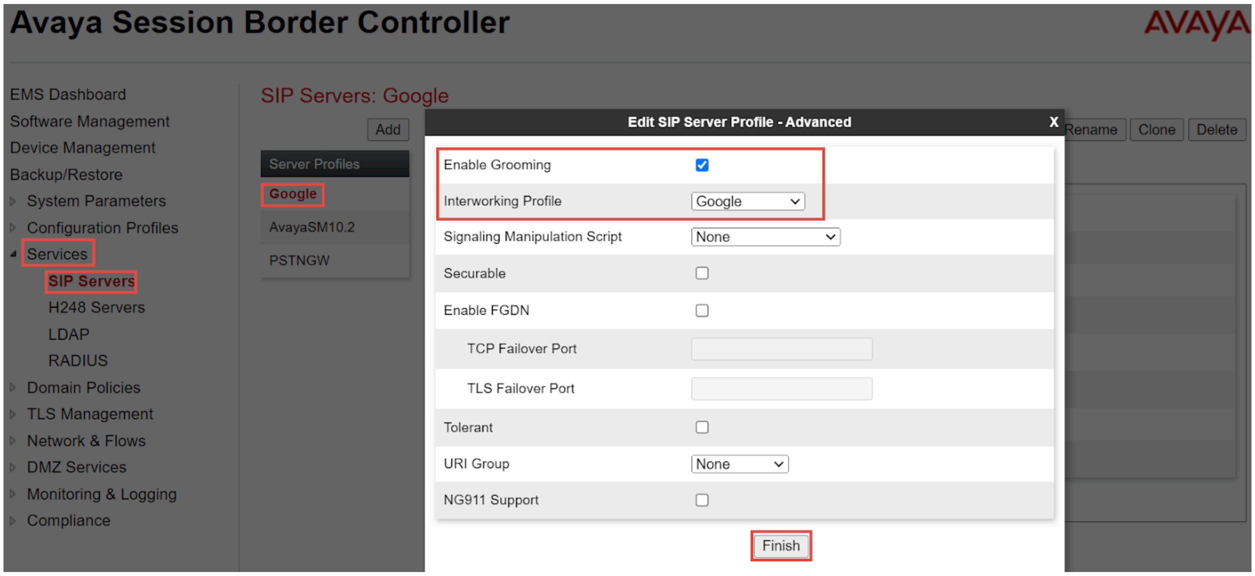


Figure 20: SIP Server for Google CES (Cont.)

SIP Server for **PSTN Gateway**

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

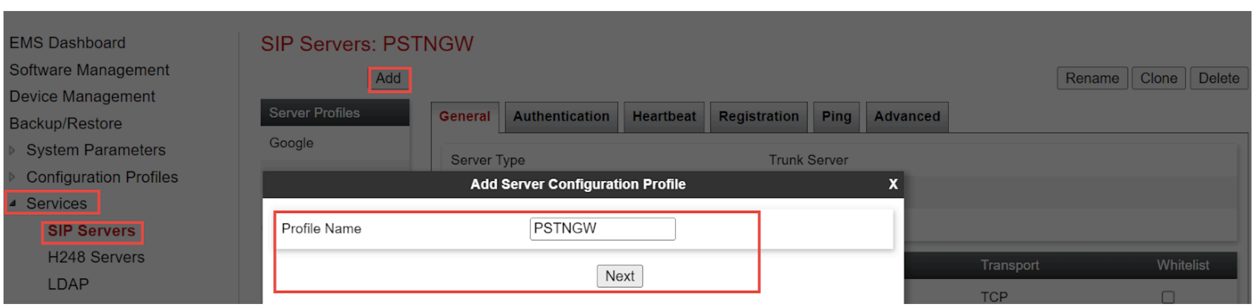


Figure 21: SIP Server for PSTN Gateway

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

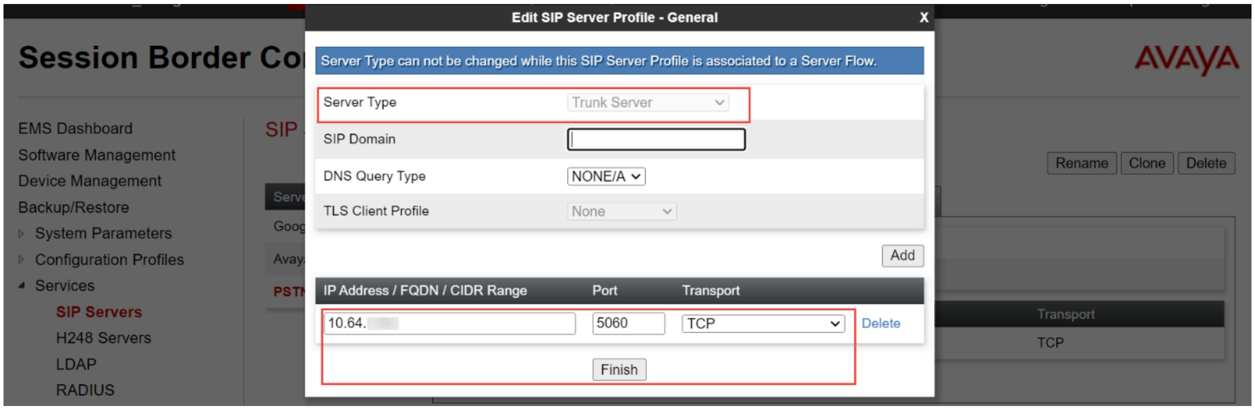


Figure 22: SIP Server for PSTN Gateway (Cont.)

- Navigate: **Heartbeat** tab
- Set Enable Heartbeat: **Checked**
- Set Method: **OPTIONS**
- Set Retry Timeout on Connection Failure: **30 seconds**
- Set Frequency: **60 seconds**
- Set From URI: **ping@10.64.X.X**
- Set To URI: **ping@10.64.X.X**
- Click **Finish**

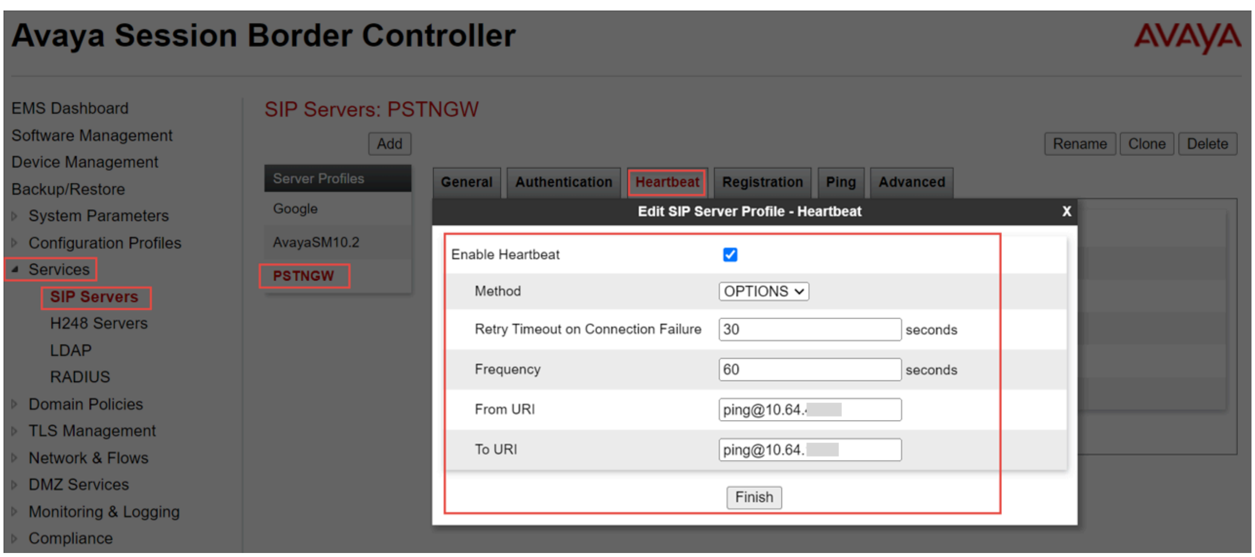


Figure 23: SIP Server for PSTN Gateway (Cont.)

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

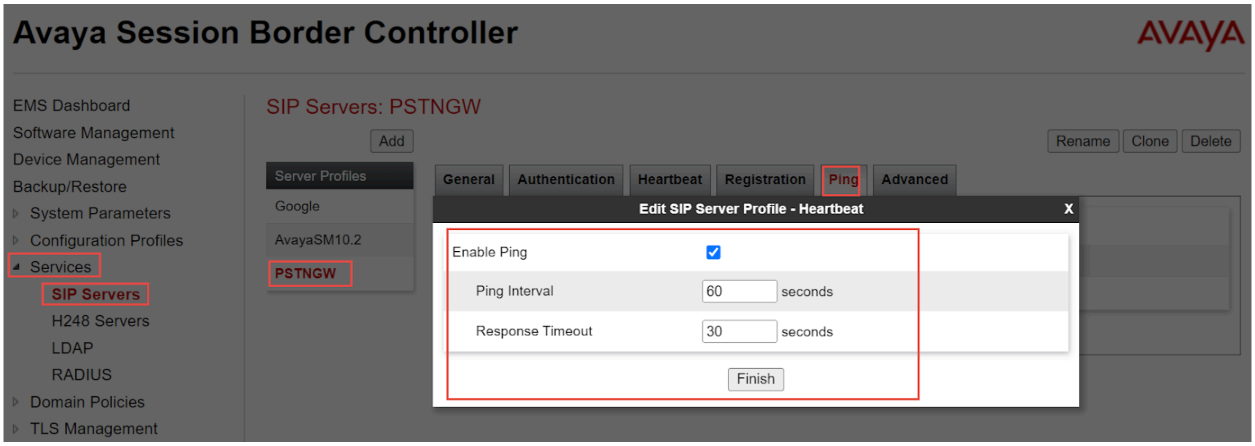


Figure 24: SIP Server for PSTN Gateway (Cont.)

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

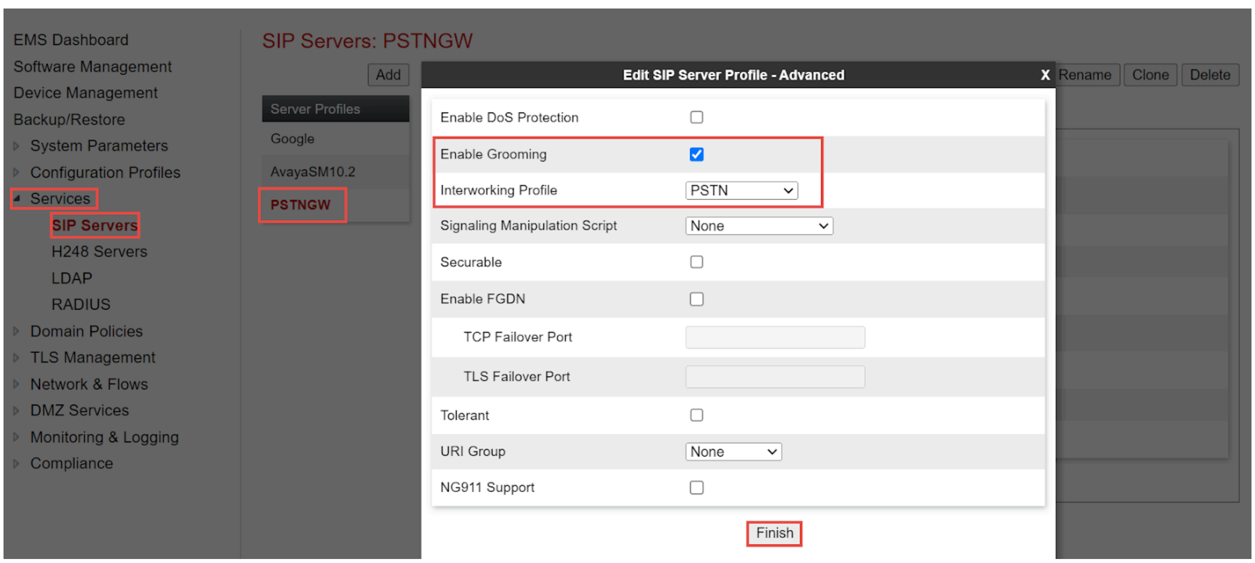


Figure 25: SIP Server for PSTN Gateway (Cont.)

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**

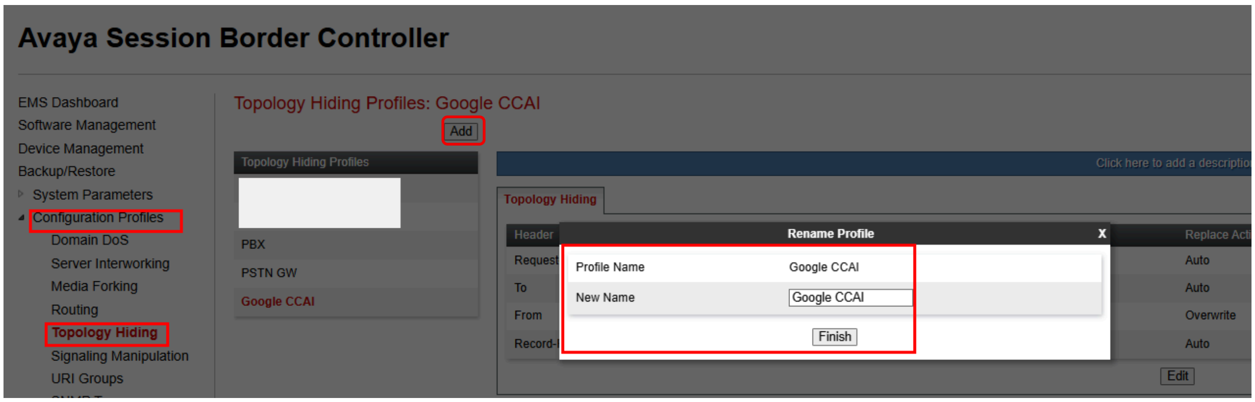


Figure 26: Topology Hiding for Google CES

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

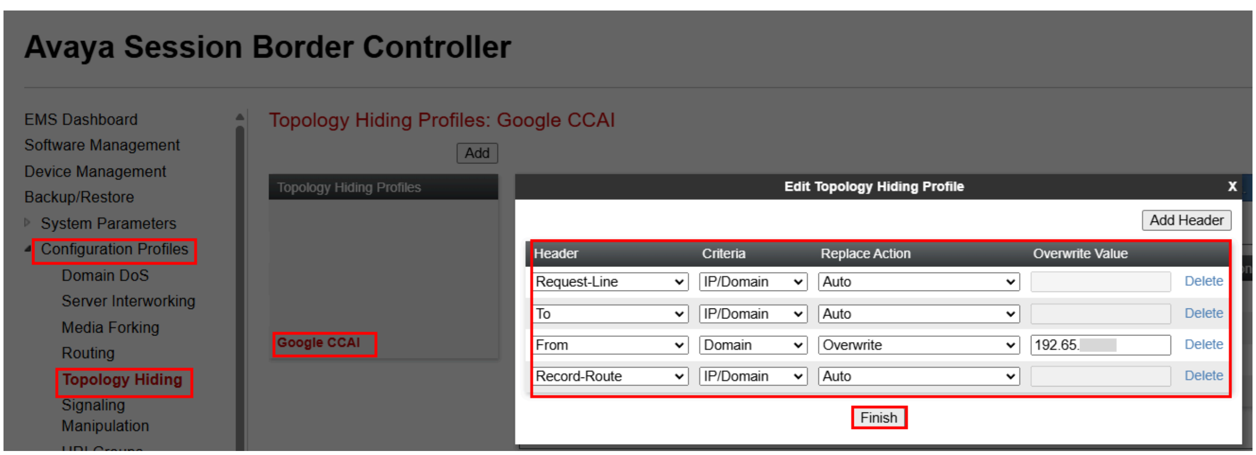


Figure 27: Topology Hiding for Google CES (Cont.)

7.4.5 Routing

Routing for **Avaya Aura SM**

- Navigate: **Configuration Profiles** □ **Routing**
- Click **Add**
- Set Profile Name: **AvayaSM10.2**
- Click **Next**

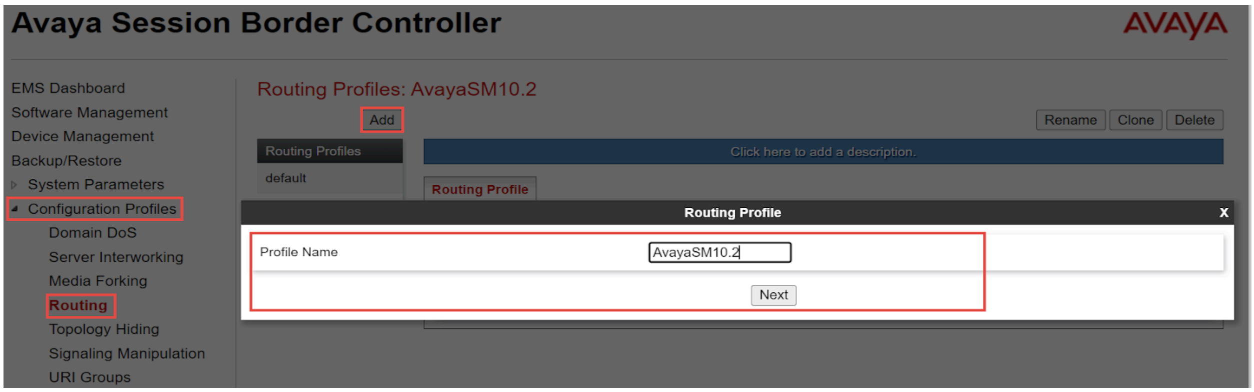


Figure 28: Routing for Avaya Aura SM

- Set URI Group: *
- At Routing Profile Window, Click **Add**
- Set Priority/Weight: **1**
- Select SIP Server Profile: **AvayaSM10.2.** Refer [Section 7.4.3]
- Next Hop Address: **10.80.X.X**
- Click **Finish**

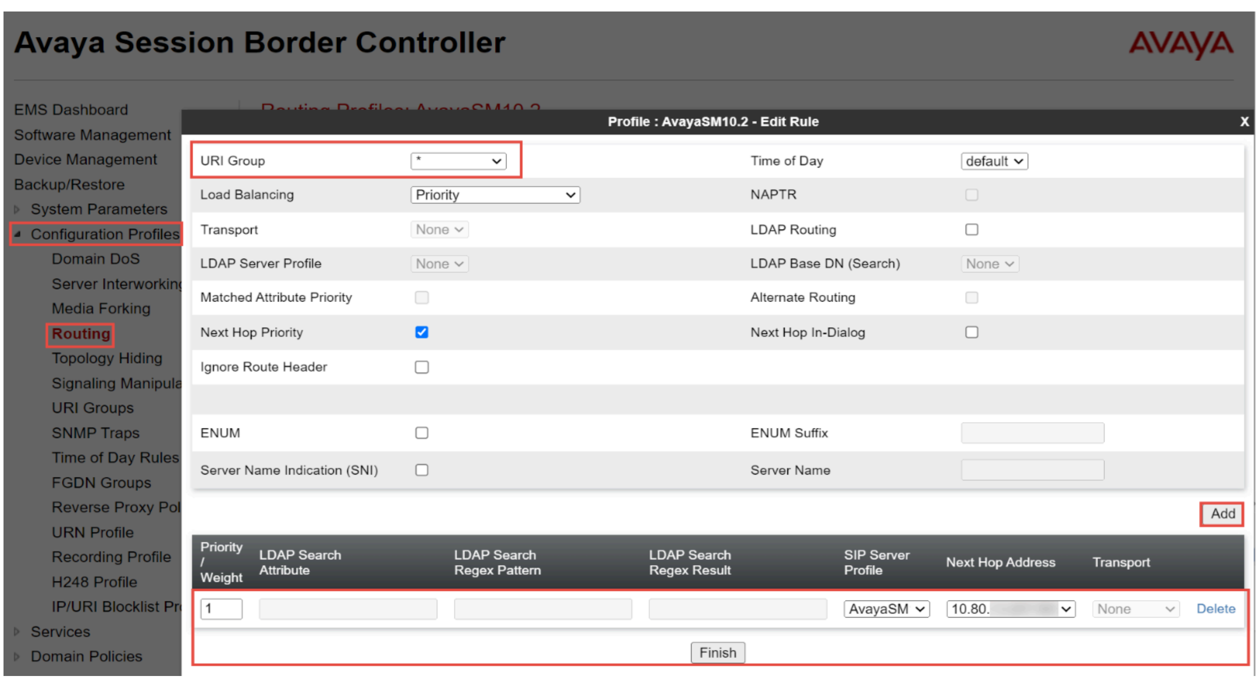


Figure 29: Routing for Avaya Aura SM (Cont.)

Routing for **PSTN Gateway**

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

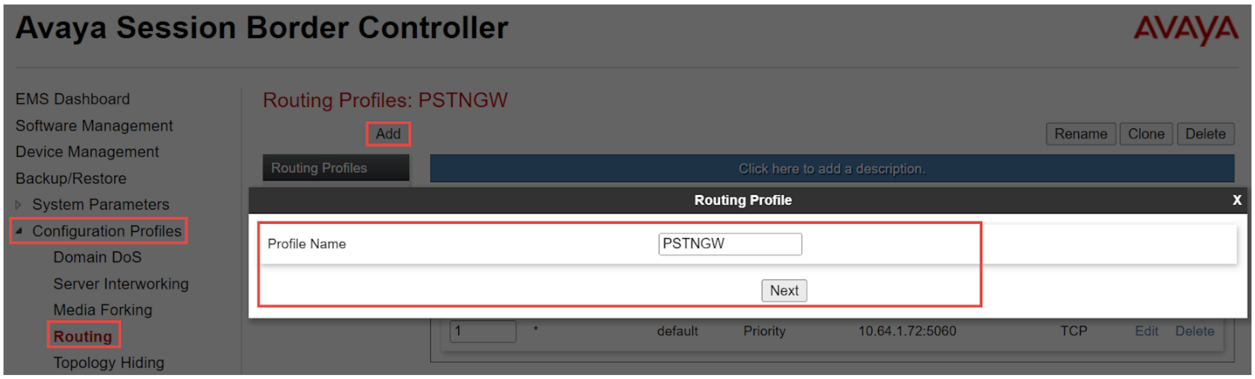


Figure 30: Routing for PSTN Gateway

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

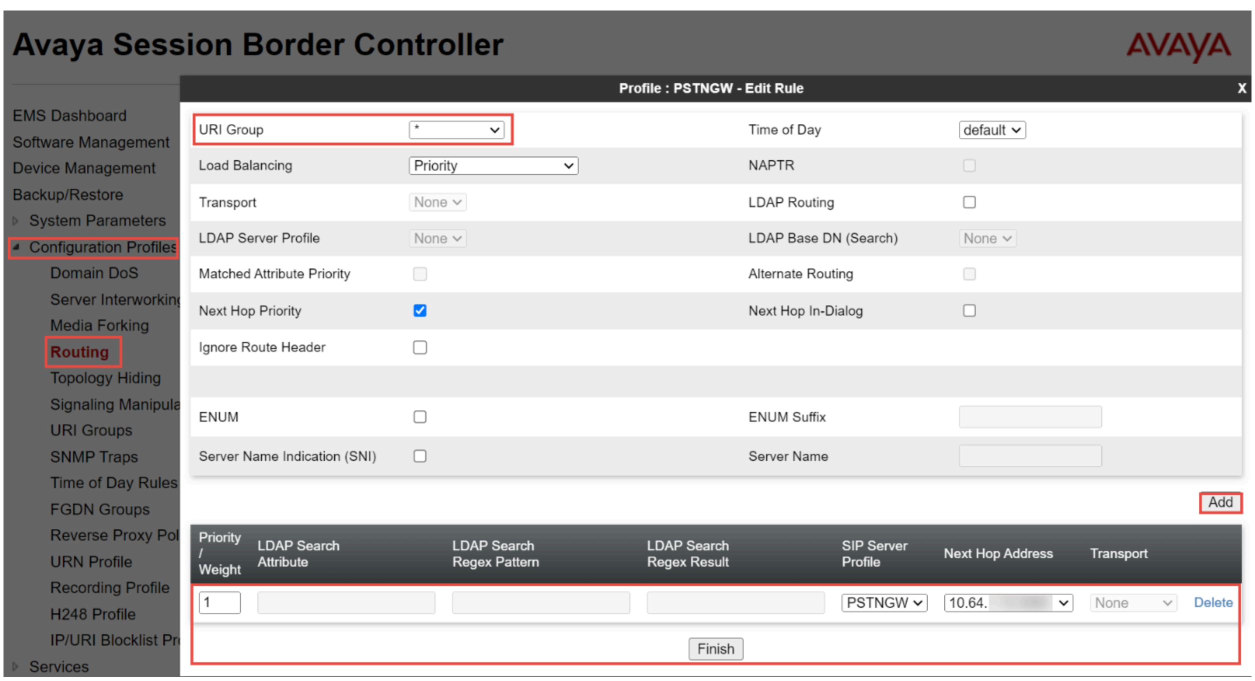


Figure 31: Routing for PSTN Gateway (Cont.)

Routing for **Google CES**

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

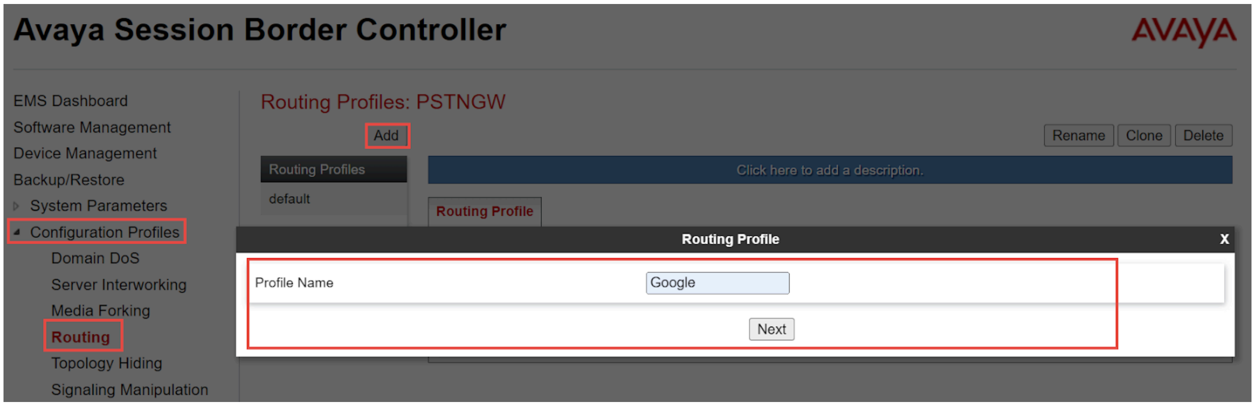


Figure 32: Routing for Google CES

- Set URI Group: *
- 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** from the dropdown
- Click **Finish**

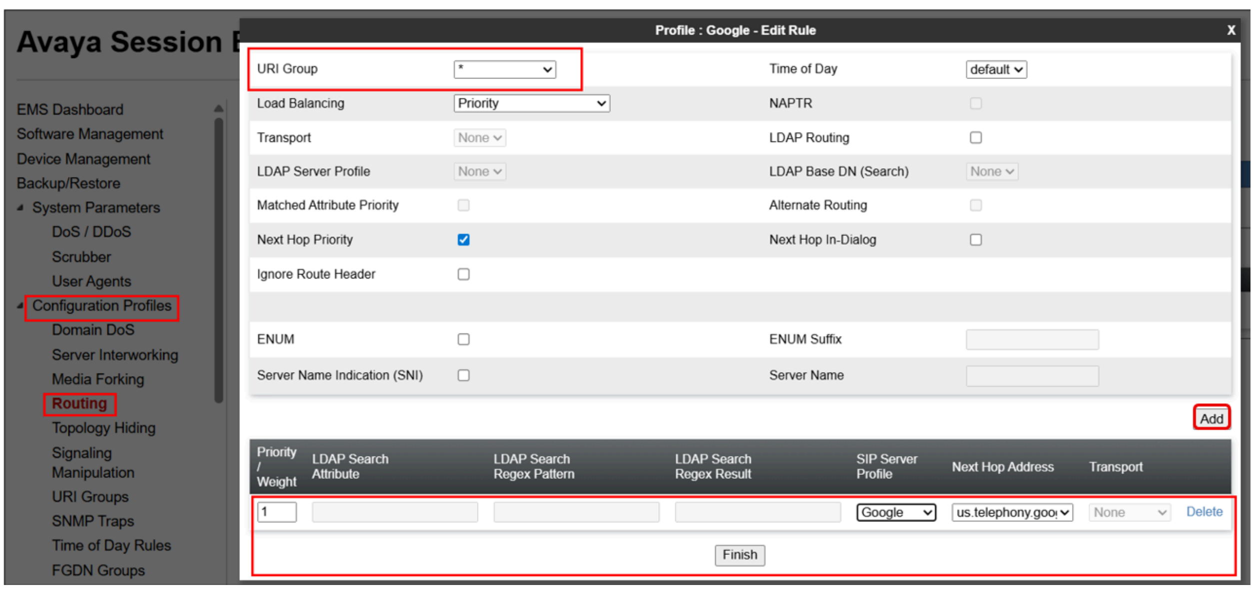


Figure 33: Routing for Google CES (Cont.)

7.4.6 Recording Profile

- Navigate: **Configuration□ Recording Profile**
- Click **Add**
- Set Profile Name: **Google_RP**
- Click **Next**

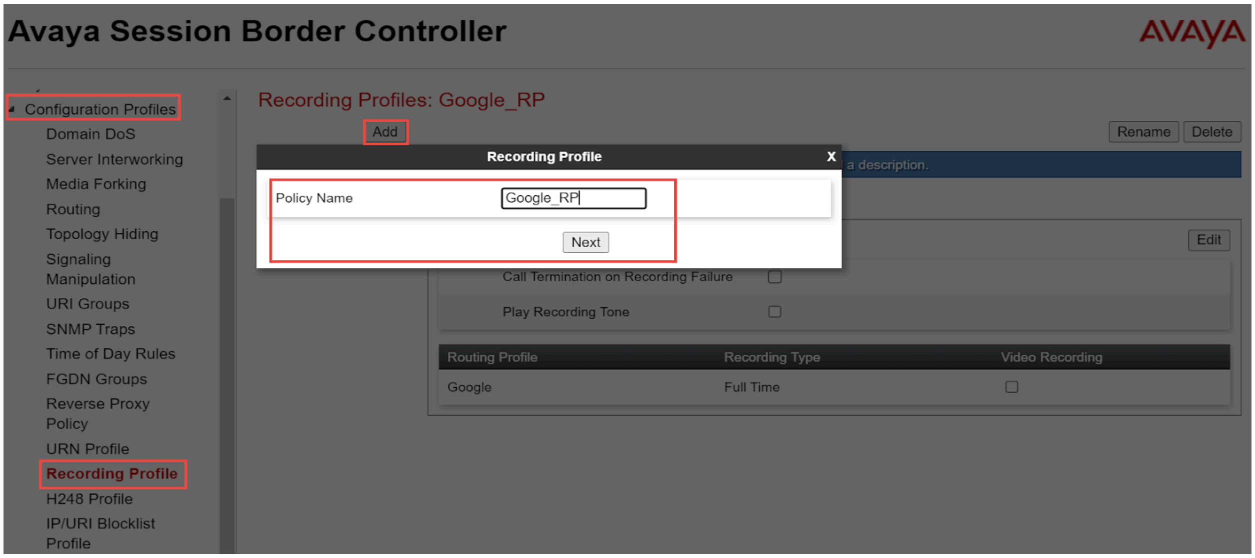


Figure 34: Recording Profile for Google CES

- Set Routing Profile: Select **Google.** Refer [Section 7.4.5](#)
- Set Recording Type: Select **Full Time** from the dropdown
- Click **Finish**

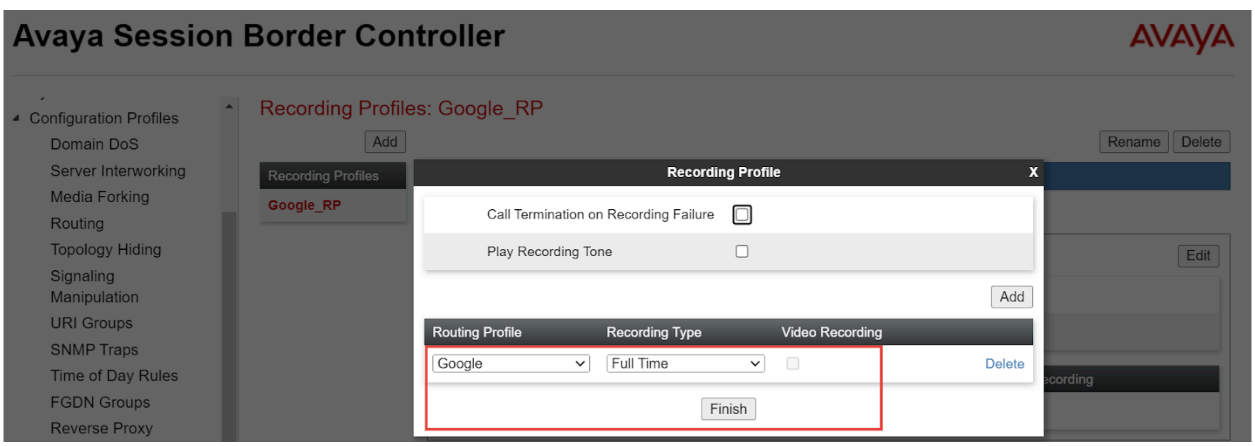


Figure 35: Recording Profile for Google CES (Cont.)

7.4.7 Session Policies

- Navigate: **Domain Policies** ⎕ **Session Policies**
- Select default under Session Policies, Click **Clone**
- Set Profile Name: **Google_SP**
- Click **Next**

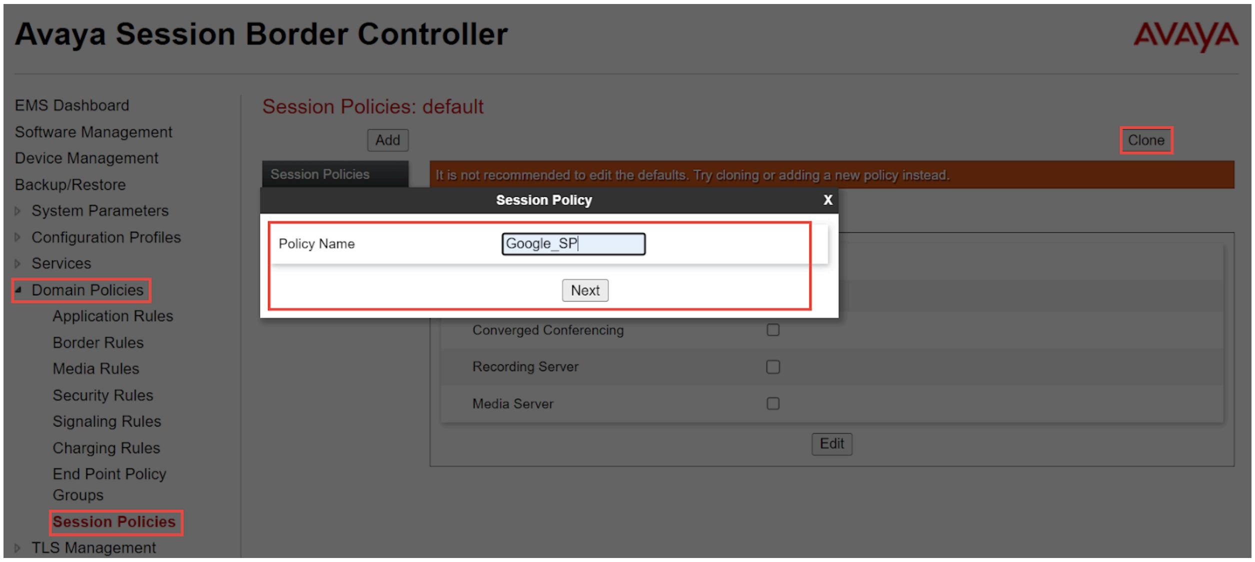


Figure 36: Session Policies for Google CES

- Media Anchoring: **Checked**
- Recording Server: **Checked**
- Set Recording Profile: Select the recording profile **Google_RP.** Refer Section 7.4.6
- Click **Finish**

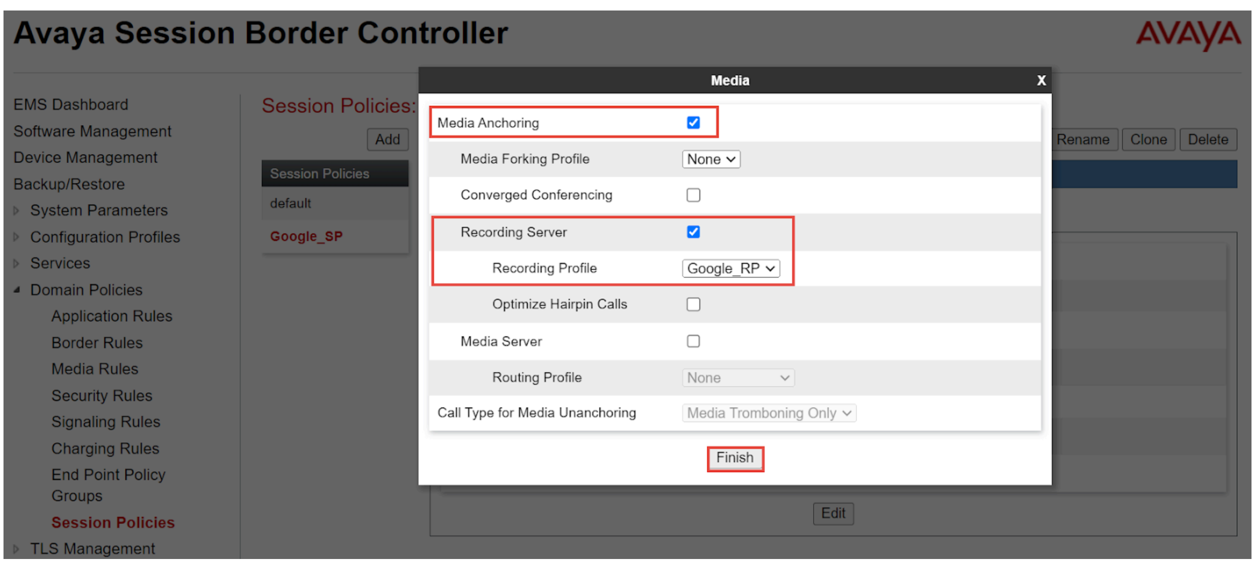


Figure 37: Session Policies for Google CES (Cont.)

7.4.8 Session Flows

- Navigate: **Network & Flows Session Flows**
- Click **Add**

Figure 38: Session Flows

- Set Name: **Google_SF**
- Select Session Policy: **Google_SP.** Refer [Section 7.4.7](#)
- Click **Finish**

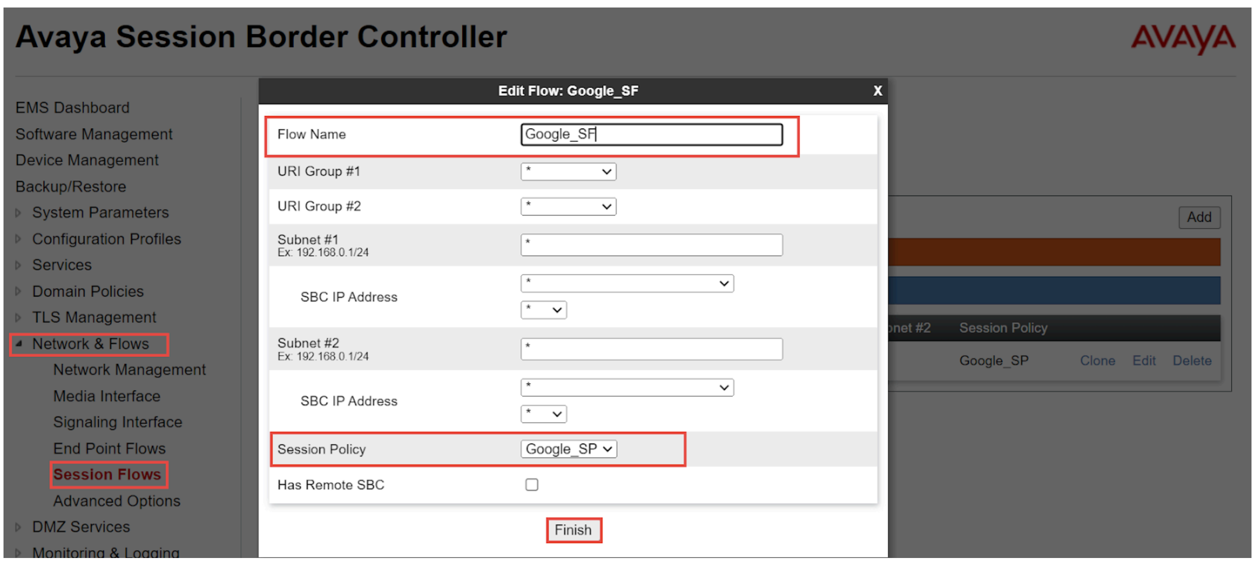


Figure 39: Session flow for Google CES

7.4.9 Signaling Manipulation

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

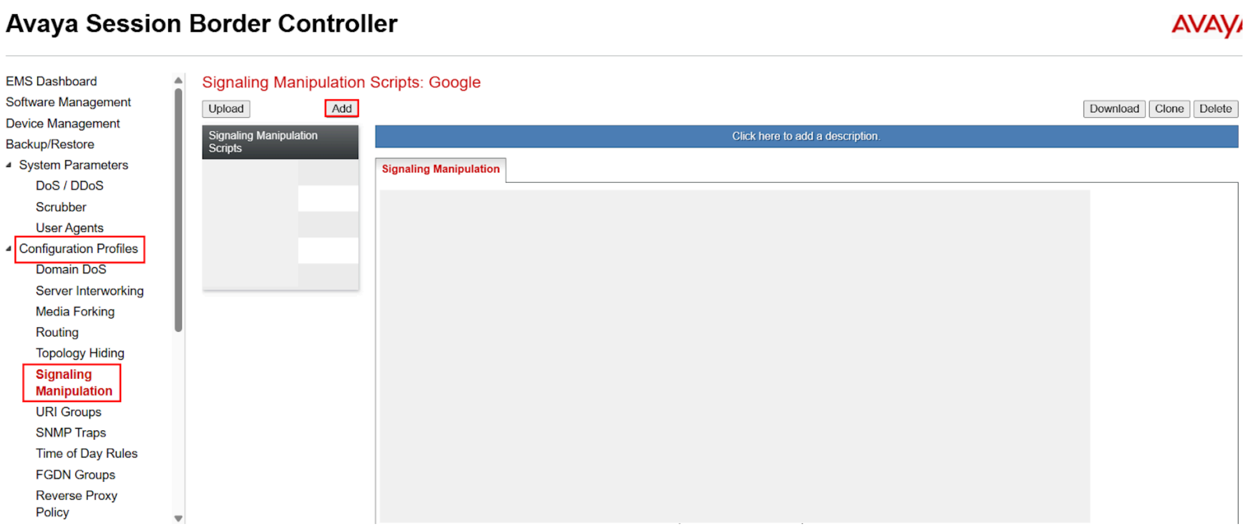


Figure 40: Signaling Manipulation

- Title: **Google**
- 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.
- When the call is answered immediately (i.e. before the first ring) by the PBX user, the Avaya SBC sends an UPDATE message towards Google, which results in Google responding with 491 Request Pending message. Also, when the call is disconnected by the PBX user, Avaya SBC does not send BYE message towards Google. To avoid this, UPDATE method is removed from the Allow header.
- Click **Save**


```
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("(.*)", "+1361400XXXX ");
        %HEADERS["TO"][1].URI.USER.regex_replace("^..........", "+1361400XXXX ");
        %HEADERS["Allow"][1].regex_replace(", UPDATE,", "");
        }
    act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING" and %METHOD="ACK"
    {
        %HEADERS["TO"][1].URI.USER.regex_replace("^..........", "+1361400XXXX ");
        }
    act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING" and %METHOD="UPDATE"
    {
         %HEADERS["TO"][1].URI.USER.regex_replace("^..........", "+1361400XXXX");
        %HEADERS["Request_Line"][1].regex_replace(";transport=udp", "");
        %HEADERS["Content-Type"][1].regex_replace("application/rs-metadata", "application/rs-metadata+xml");

        }
    }
```

Figure 41: Signaling Manipulation - 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("(.*)", "+1361400XXXX ");
    %HEADERS["TO"][1].URI.USER.regex_replace("^..........", "+1361400XXXX ");
    %HEADERS["Allow"][1].regex_replace(", UPDATE,", "");
    }
  act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING" and %METHOD="ACK"
  {
    %HEADERS["TO"][1].URI.USER.regex_replace("^..........", "+1361400XXXX ");
    }
  act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING" and %METHOD="UPDATE"
  {
     %HEADERS["TO"][1].URI.USER.regex_replace("^..........", "+1361400XXXX");
    %HEADERS["Request_Line"][1].regex_replace(";transport=udp", "");
              %HEADERS["Content-Type"][1].regex_replace("application/rs-metadata",
"application/rs-metadata+xml");

    }
}
```

SIP Manipulation for Participation Label:

- The transcript recording files stored in the Google CES bucket include two participant roles "HUMAN_AGENT" and "END_USER".
- To map the participant roles to the transcripts generated, Google uses the participant labels provided in the Call-Info header. Use the below rule only if Participant labels are required.

Call-Info header with participant roles:

Call-Info:
<http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898XX/conversations/Sr_XXXX?**roles=HUMAN_AGENT,END_USER**>;purpose=Goog-ContactCenter-Conversation

```
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 =
"?roles=HUMAN_AGENT,END_USER>;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("^..........", "+214550XXXX");
      %HEADERS["Allow"][1].regex_replace(", UPDATE,", "");
   }
   act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING" and %METHOD="ACK"
   {
      %HEADERS["TO"][1].URI.USER.regex_replace("^..........", "+1314944XXXX");
      }
   act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING" and %METHOD="UPDATE"
   {
      %HEADERS["TO"][1].URI.USER.regex_replace("^..........", "+13149445XXXX");
      %HEADERS["Request_Line"][1].regex_replace(";transport=udp", "");
    }
   }
```

7.4.10 Media Rules

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

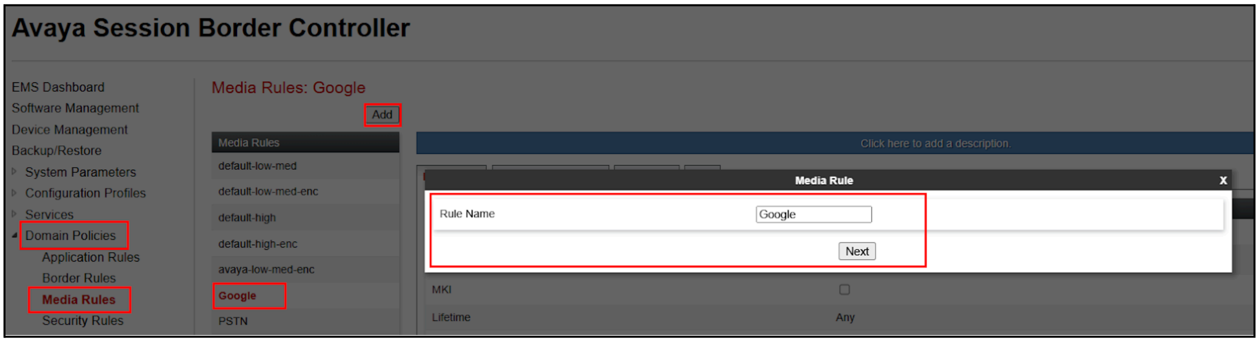


Figure 42: Media Rules

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

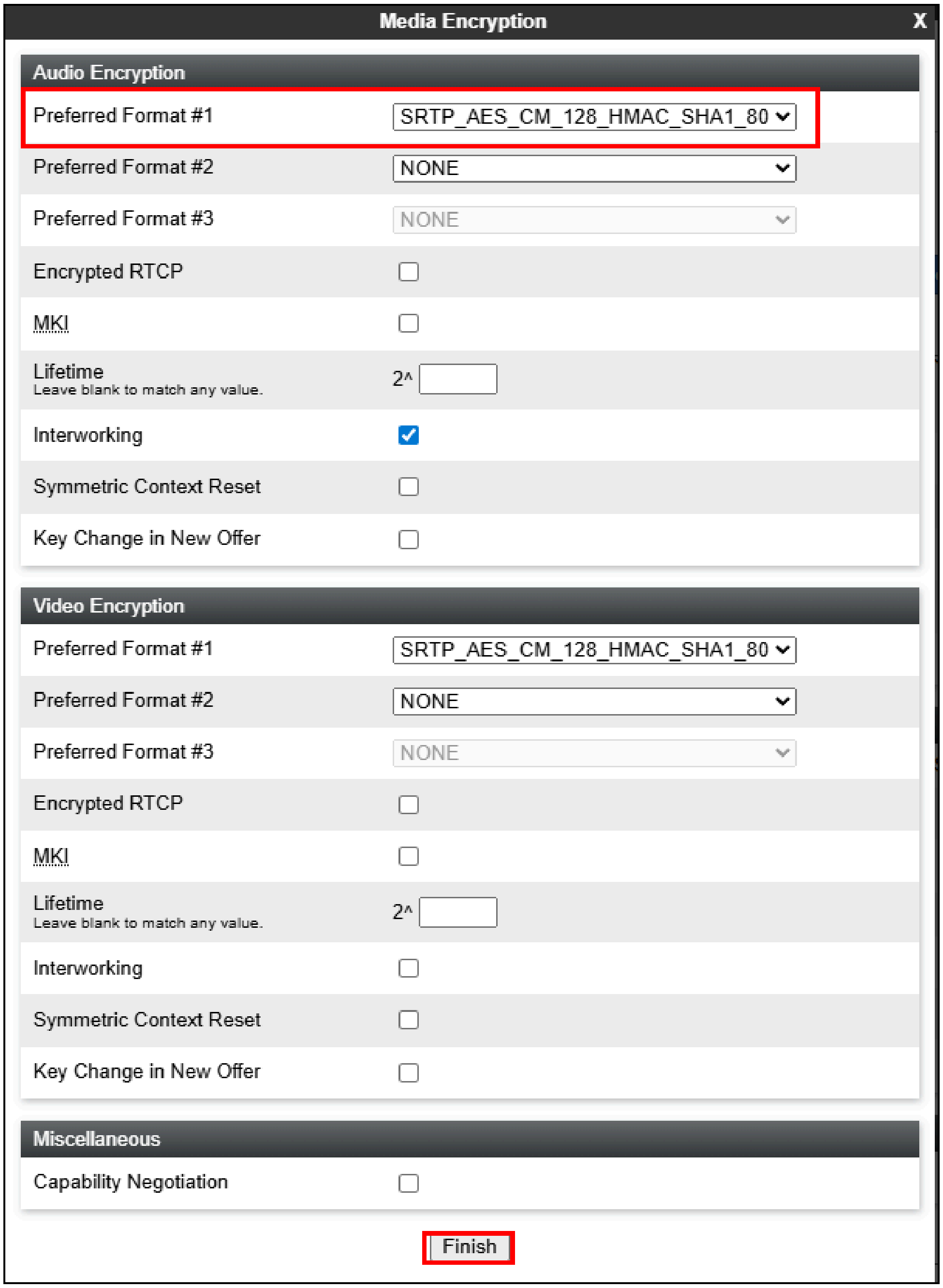


Figure 43: Media Rules (Cont.)

7.4.11 Signaling Rules

- Configure Navigate: **Domain Policies □ Signaling Rules**
- Select default under Signaling Rules, Click **Clone**
- Set Rule Name: **Avaya SM**
- Click **Next**

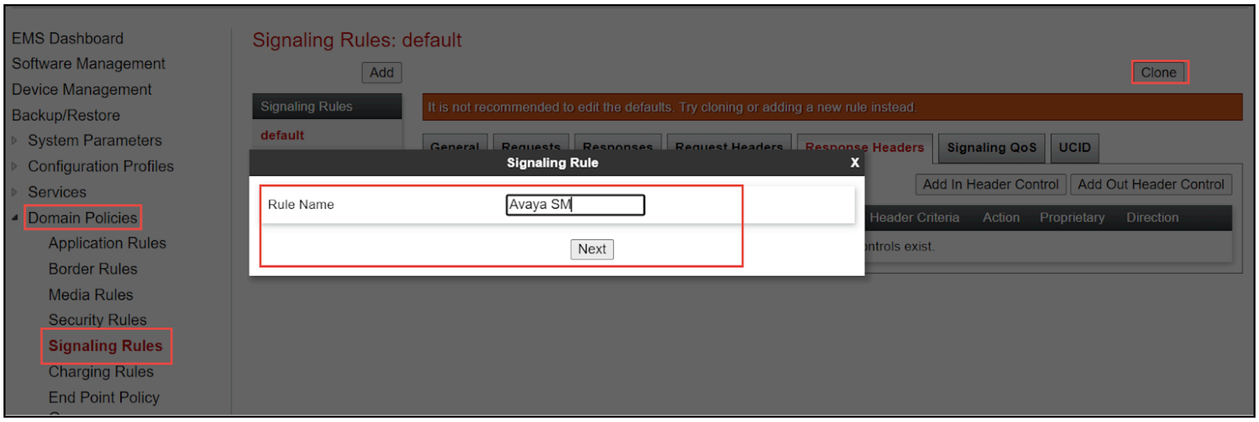


Figure 44: Signaling Rules for Avaya Aura SM

- Select the newly cloned **Signaling Rule Avaya SM**, under tab **Request Headers**.
- Click **Add In Header Control**

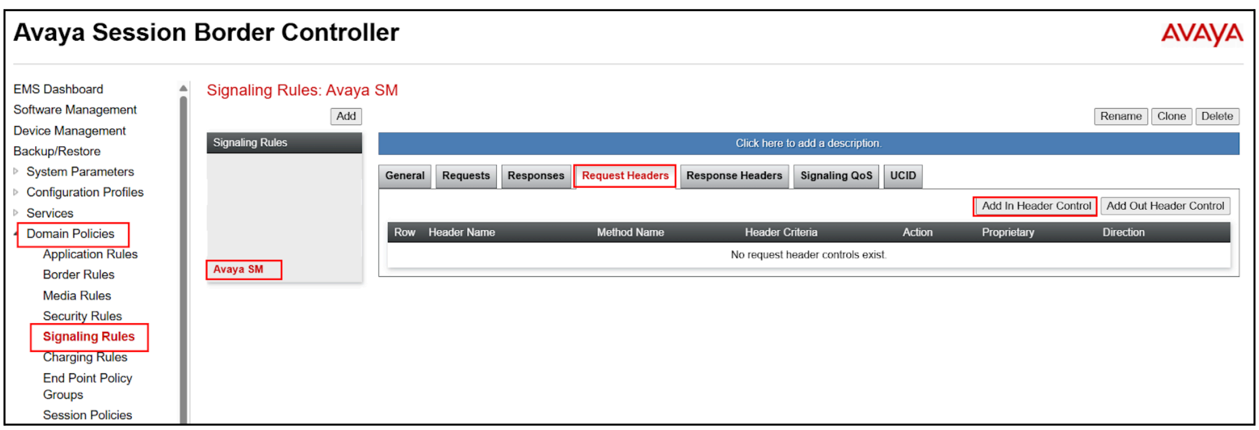


Figure 46: Signaling Rules for Avaya Aura SM (Cont.)

- Set Proprietary Request Header: **Checked**
- Set Header Name: **AV-Global-Session-ID**
- Set Method Name: Select ALL from the drop down
- Set Header Criteria: **Forbidden**
- Set Presence Action: **Remove header** is selected from the drop down
- Click **Finish**

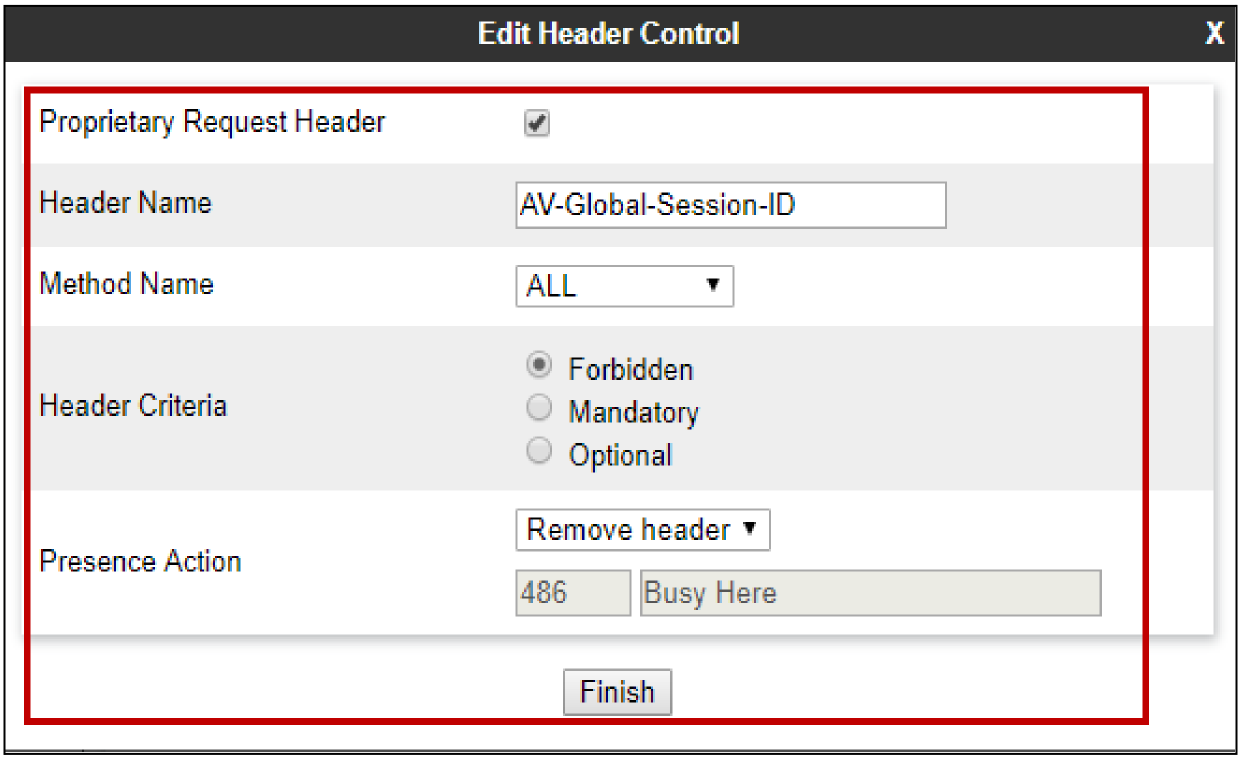


Figure 47: Signaling Rules for Avaya Aura SM (Cont.)

- Repeat the same steps for all other required headers for Request Headers.

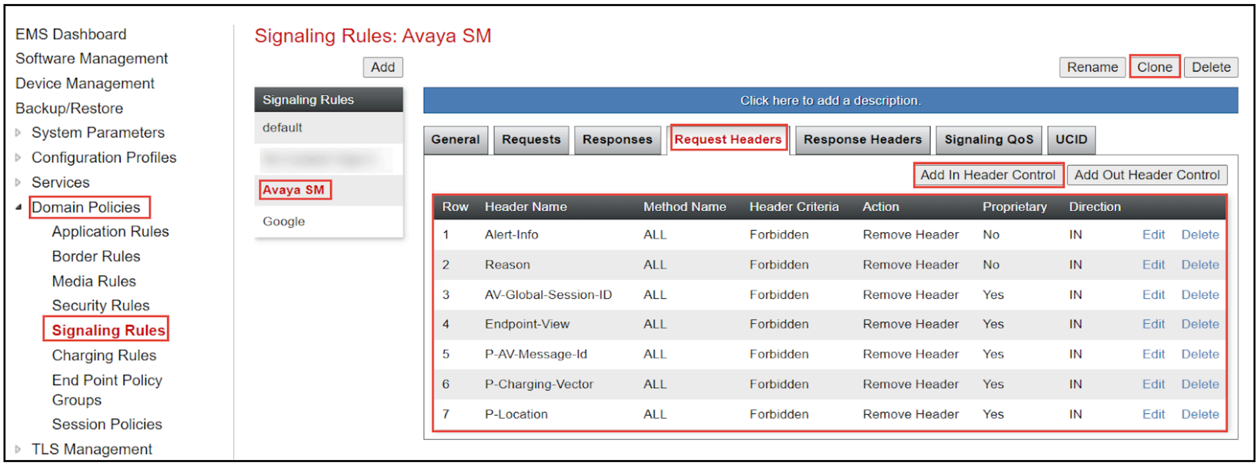


Figure 48: Signaling Rules for Avaya Aura SM (Cont.)

- Click **Add Out Header Control**
- Repeat the same steps for all the required headers for Response Headers.

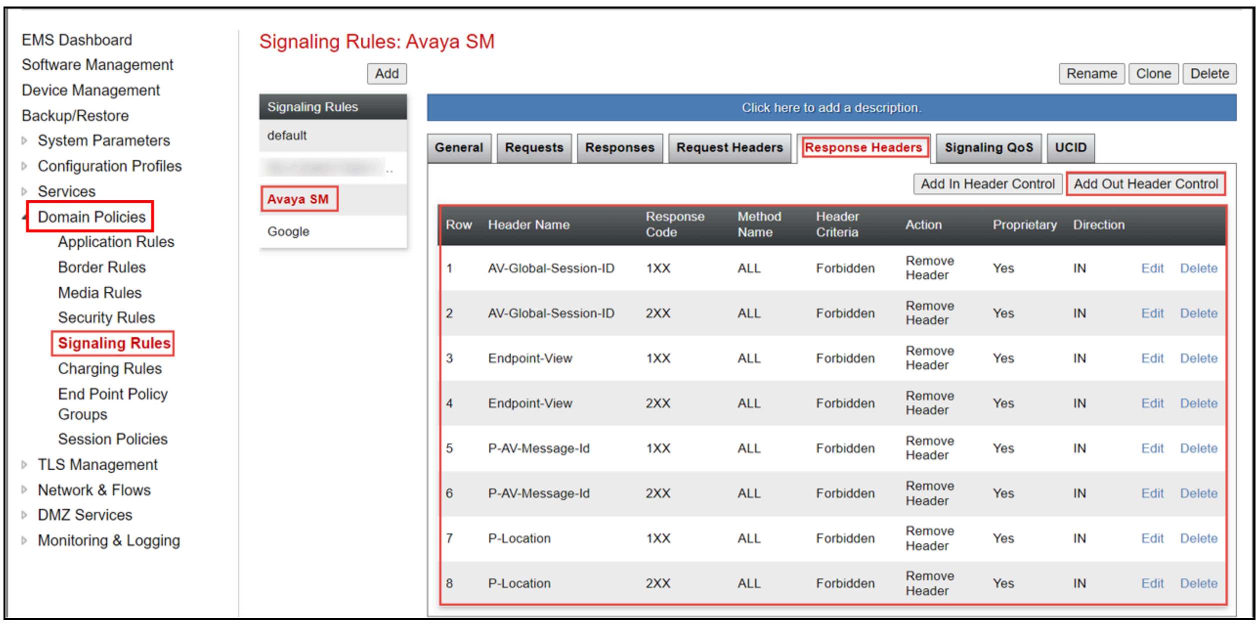


Figure 49: Signaling Rules for Avaya Aura SM (Cont.)

7.4.12 End Point Policy Groups

End Point Policy Group for **Avaya Aura SM**

- A new End Point Policy Group is created for Avaya Aura Session Manager.
- Navigate: **Domain Policies** **End Point Policy Groups**
- Select **default-low** under Policy Groups
- Click **Clone**

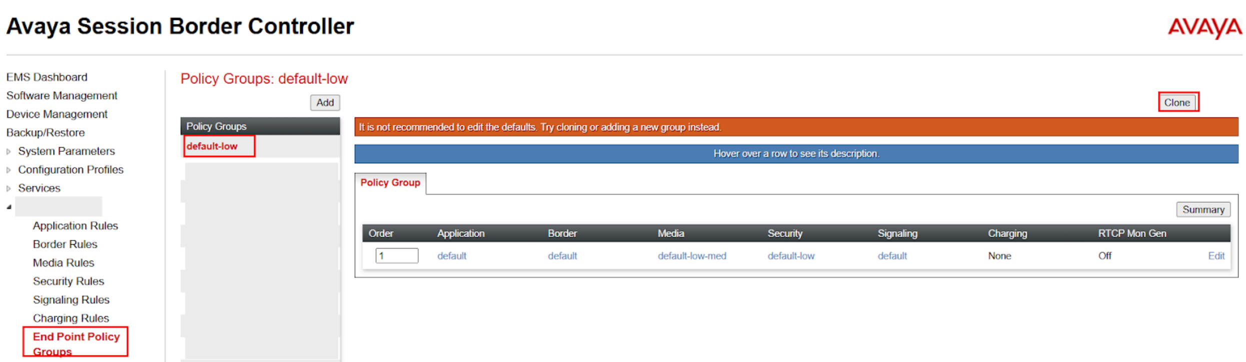


Figure 50: End Point Policy Group

- Set Group Name: **Avaya SM**
- Click **Next**

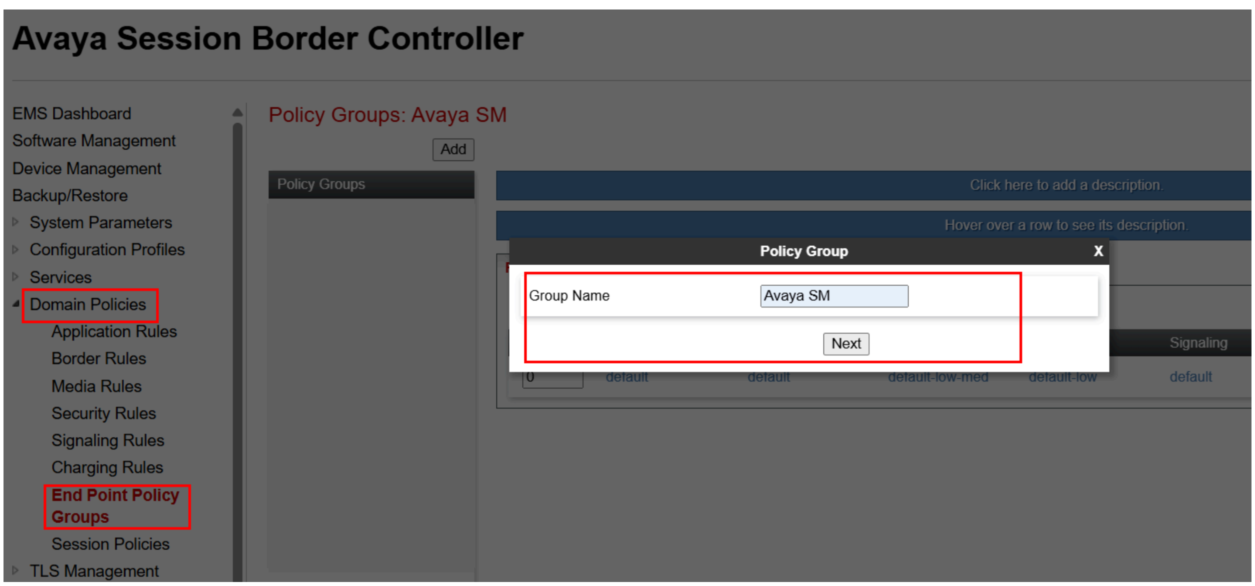


Figure 51: End Point Policy Group for Avaya Aura SM

- Set Signaling Rule: **Avaya SM**. Refer [Section 7.4.11]
- Click **Finish**

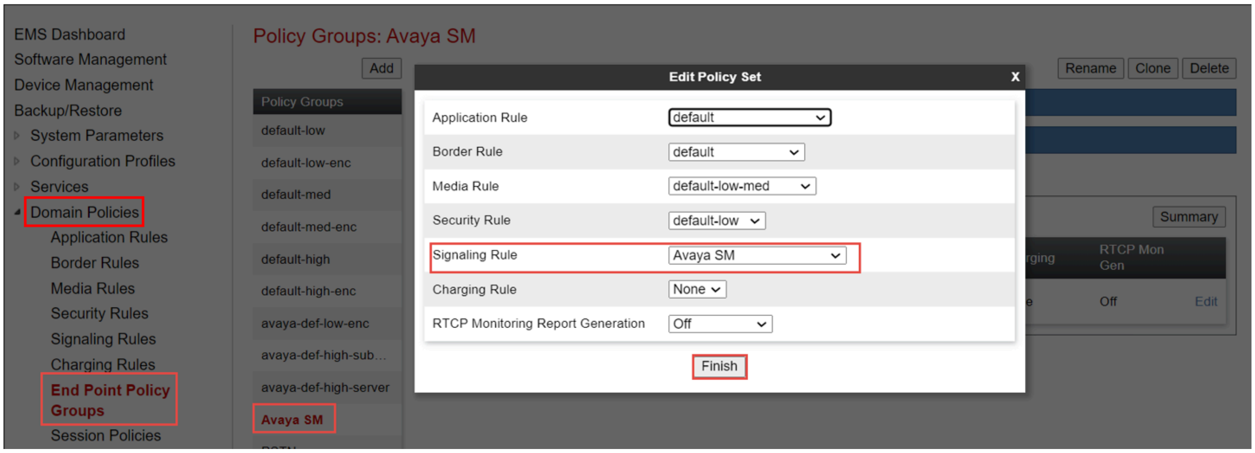


Figure 52: End Point Policy Group for Avaya Aura SM (Cont.)

End Point Policy Group for **Google CES**

- Select Media Rule: **Google.** Refer [Section 7.4.10]
- Click **Finish**

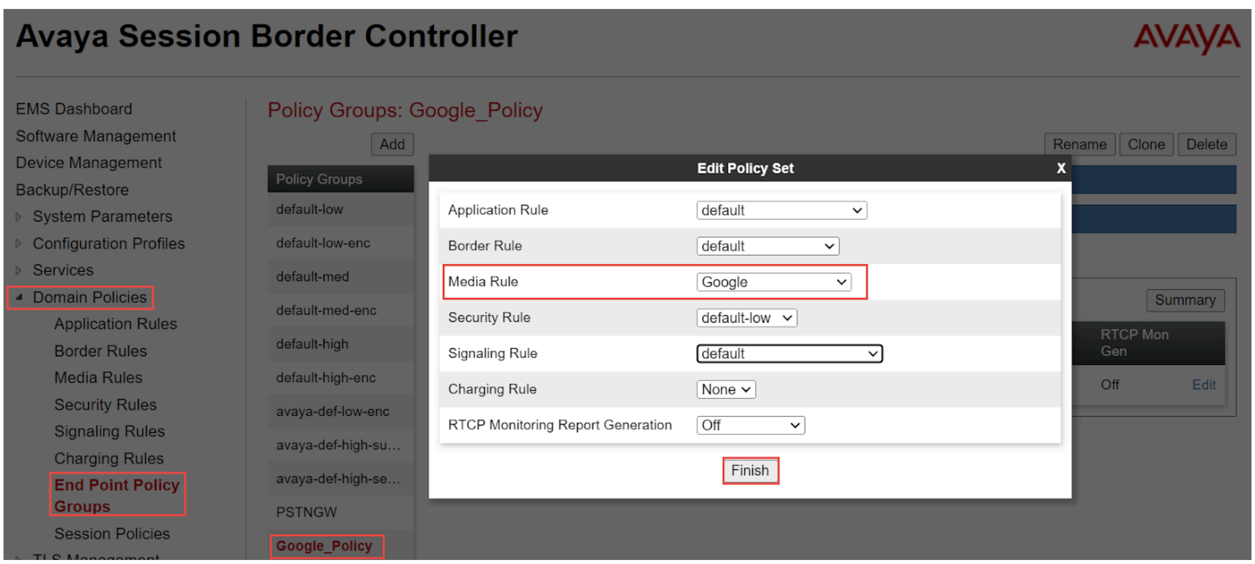


Figure 53: End Point Policy Group for Google CES

End Point Policy Group for **PSTN Gateway**

- Navigate: **Domain Policies** □ **End Point Policy Groups**
- Select **default-low** under Policy Groups
- Click **Clone**

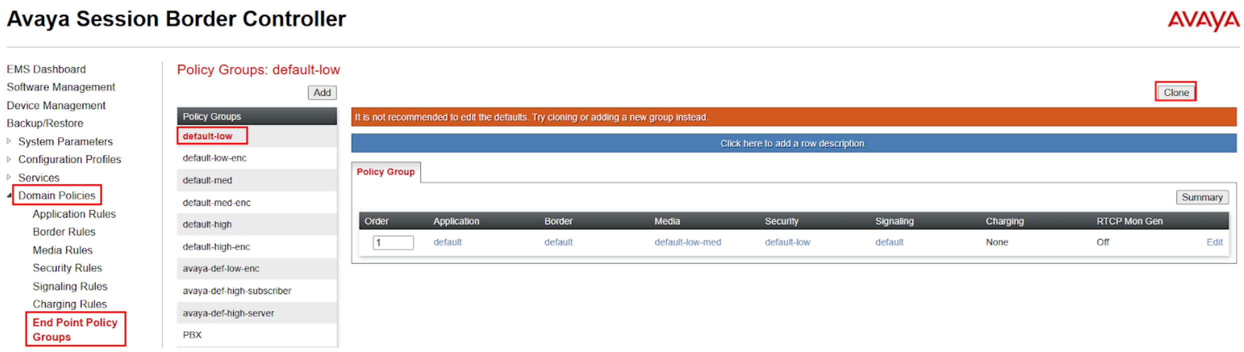


Figure 54: End Point Policy Groups for PSTN Gateway

- Set Group Name: **PSTN**
- Click **Finish**

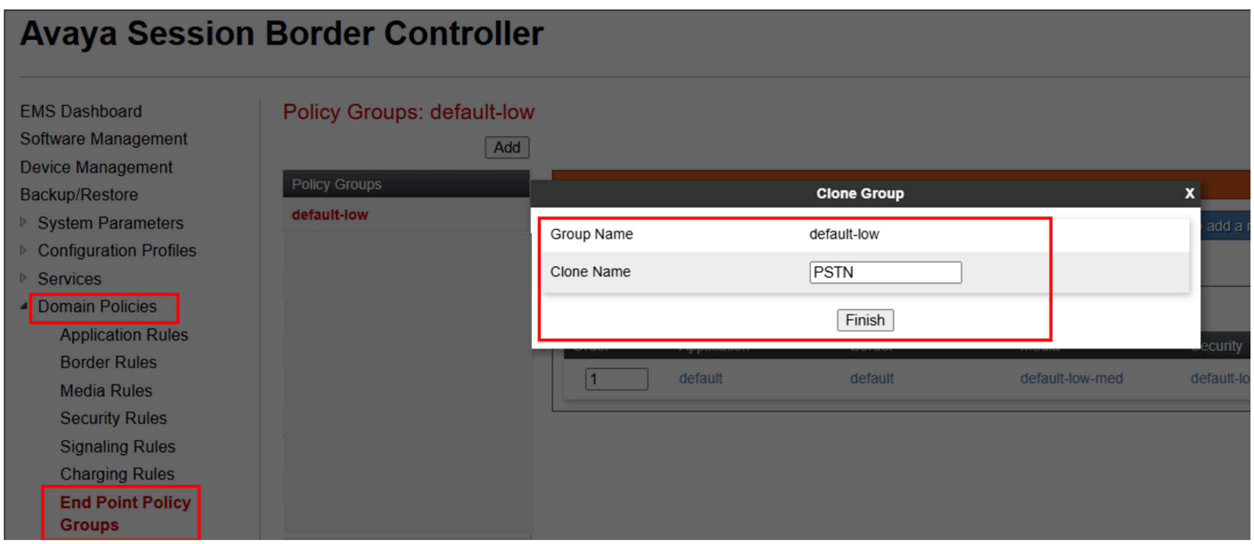


Figure 55: End Point Policy Group for PSTN Gateway (Cont.)

- Click **Finish**

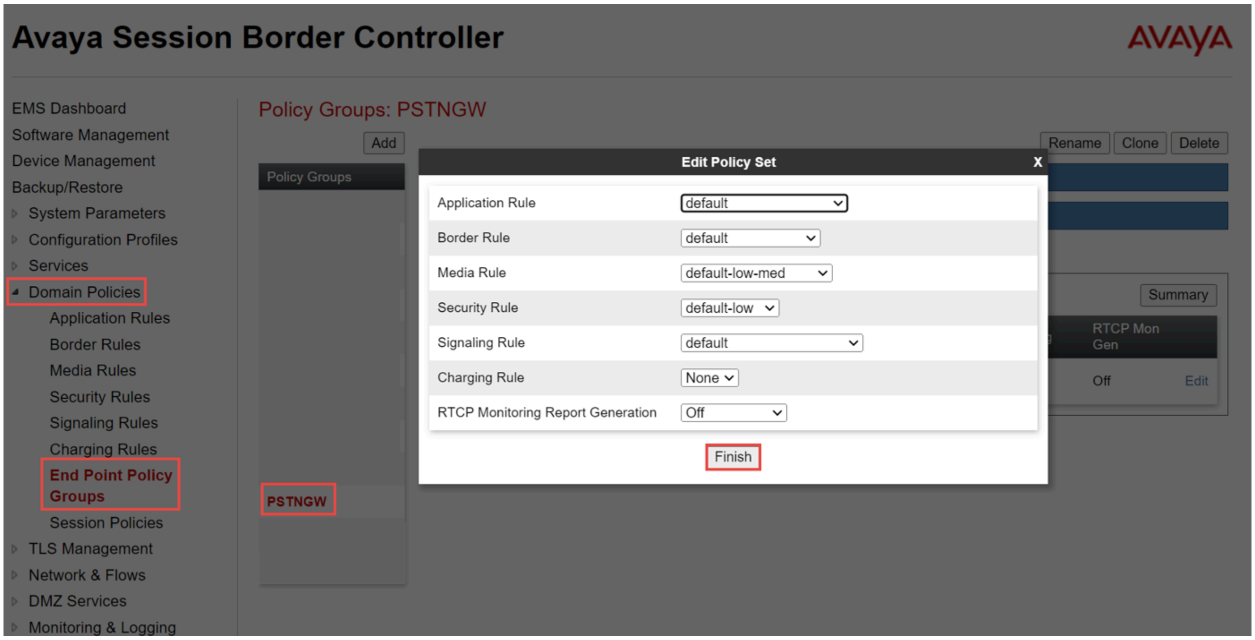


Figure 56: End Point Policy Group for PSTN Gateway (Cont.)

7.4.13 Media Interface

- Navigate: **Network & Flows** □ **Media Interface**. Click **Add**
- Set Name: **AvayaSM10.2**
- Set IP Address: **AvayaSM10.2 (A2, VLAN 0)** from the drop down and the IP address populates automatically. The IP address for Interface facing Avaya Aura SM is **10.70.X.X**
- Set Port Range: **35000-40000**
- Click **Finish**

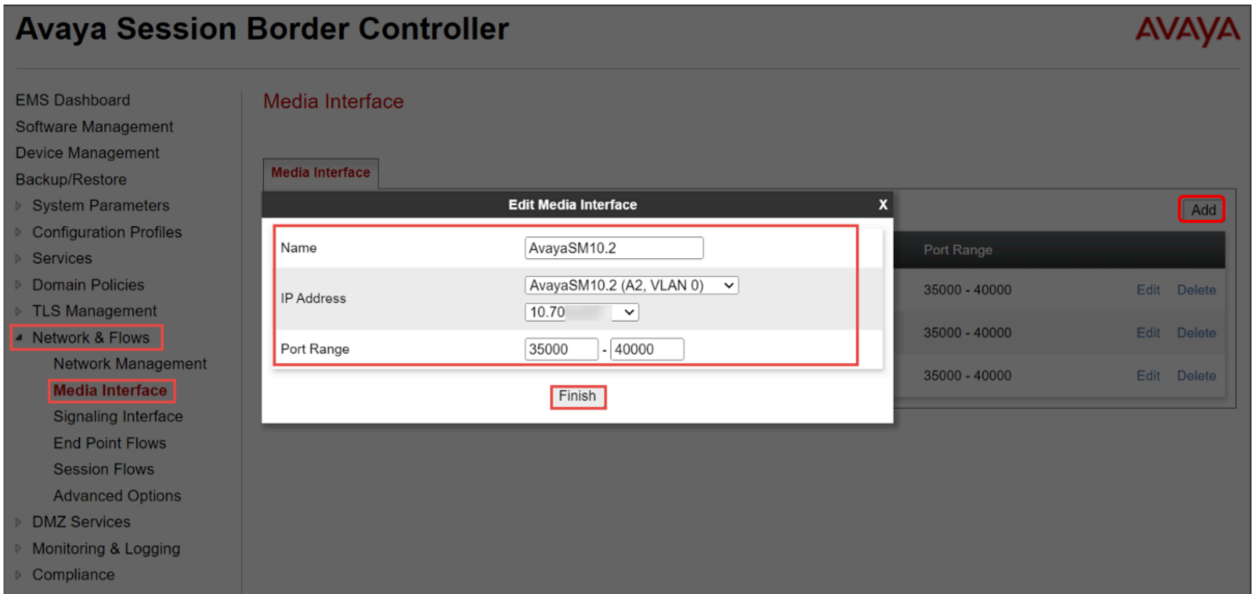


Figure 57: Media Interface Facing Avaya Aura SM

- 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**

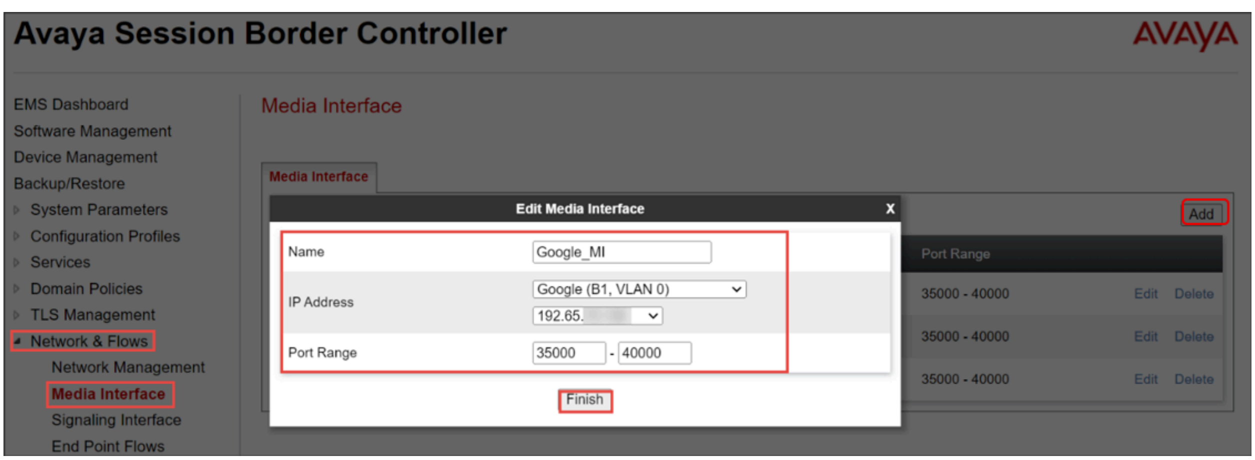


Figure 58: Media Interface Facing Google CES

- Set Name: **PSTNGW**
- Set IP Address: **PSTNGW (B2, VLAN 0)** from the drop down and the IP address populates automatically. The IP address for Interface facing PSTN Gateway is **10.64.X.X**
- Set Port Range: **35000-40000**
- Click **Finish**

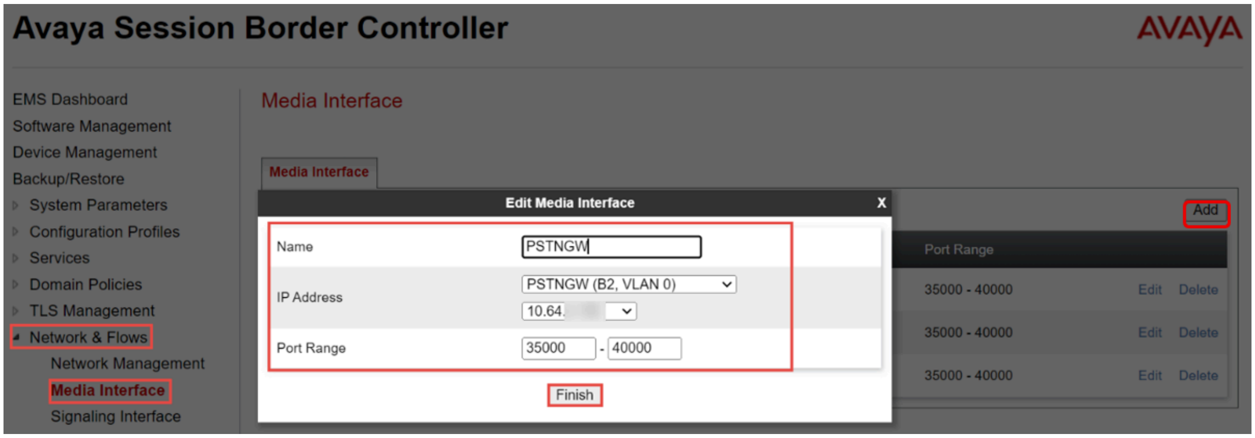


Figure 59: Media Interface Facing PSTN Gateway

7.4.14 Network Management

Network Management for **Avaya Aura SM**

- Navigate: **Network & Flows** ⎕ **Network Management**. Click Add, new Add Network Interface window appears
- Set Name: **AvayaSM10.2** is given for the network facing Avaya Aura SM
- Set Default Gateway IP Address: **10.70.X.X**
- Set Network Prefix or Subnet Mask**: 255.255.X.X**
- Set Interface: **A2**
- Set IP Address: **10.70.X.X**
- Click **Finish**

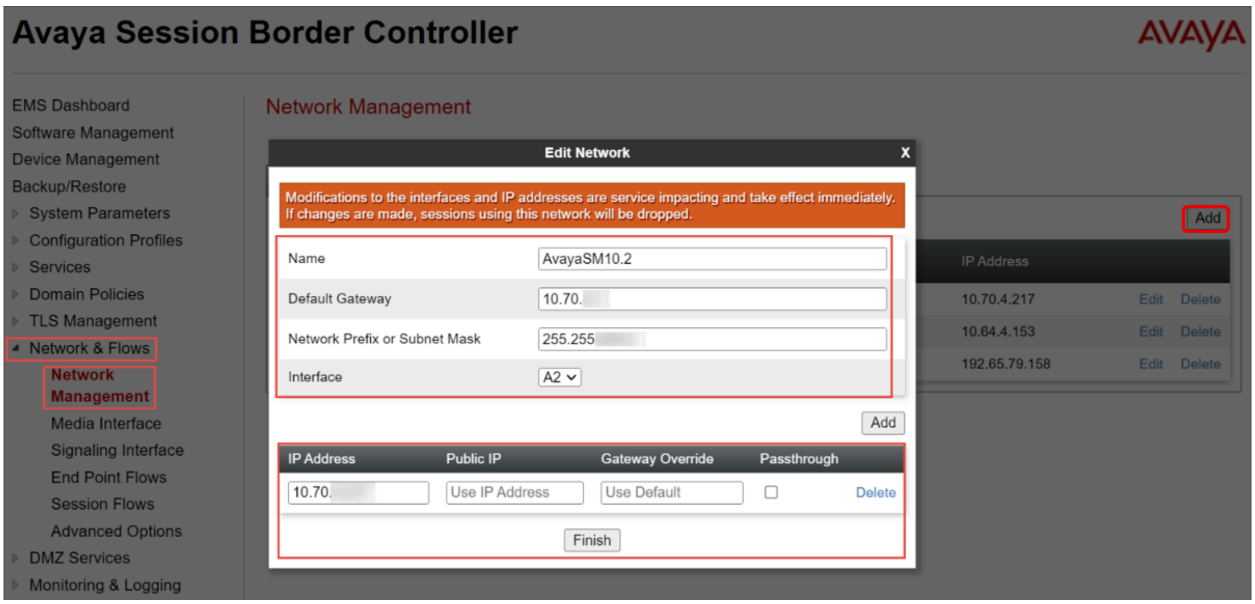


Figure 60: Network Management Facing Avaya Aura SM

Network Interface for **Google CES**

- Set Name: **Google** is given for the network facing Google CES
- 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**

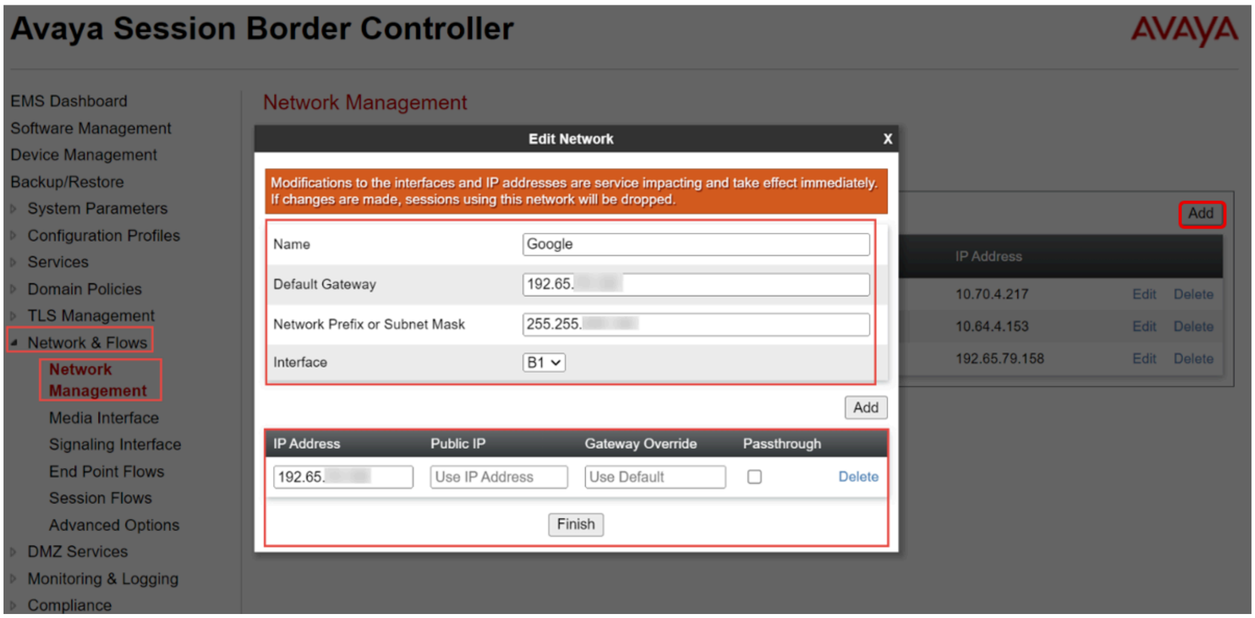


Figure 61: Network Management Facing Google CES

Network Interface for **PSTN Gateway**

- Set Name: **PSTNGW** is given for the network facing PSTN Gateway
- Set Default Gateway IP Address: **10.64.X.X**
- Set Network Prefix or Subnet Mask**: 255.255.X.X**
- Set Interface: **B2**
- Set IP Address: **10.64.X.X**
- Click **Finish**

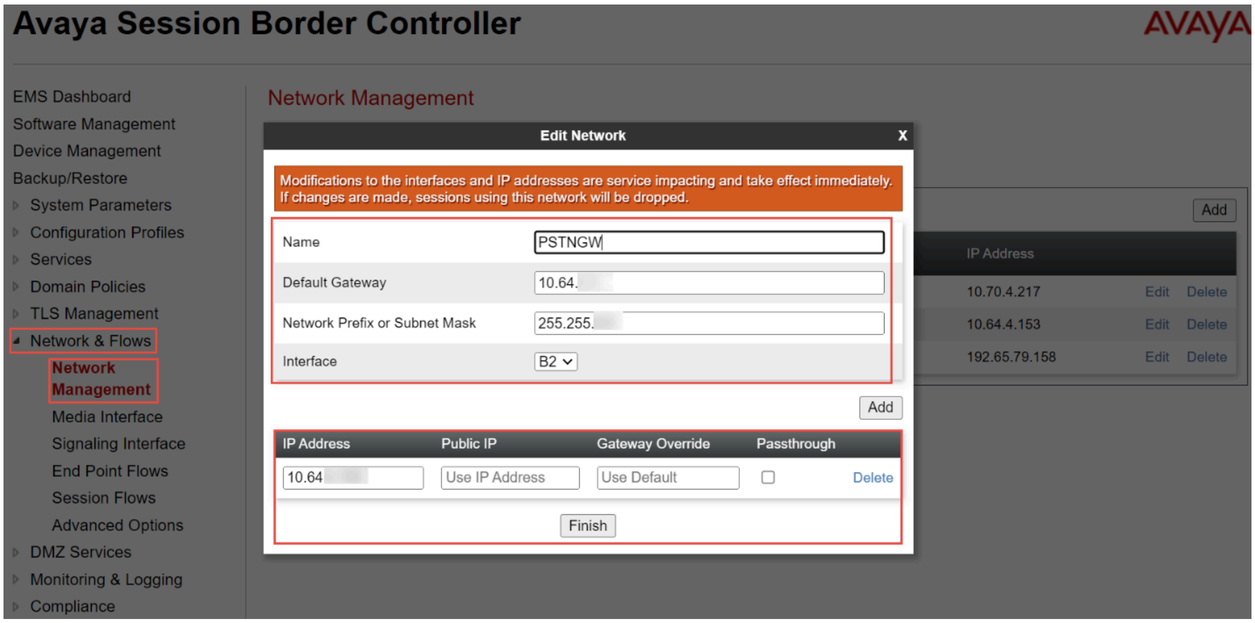


Figure 62: Network Management Facing PSTN Gateway

7.4.15 Signaling Interface

Signaling Interface for **Avaya Aura SM**

- Navigate to: **Network & Flows** ▯ **Signaling Interface**. Click **Add**, new Add Signaling Interface window appears
- Set Name: **AvayaSM10.2** is given for the interface facing **Avaya Aura SM**
- Set IP Address: **AvayaSM10.2 (A2, VLAN 0),** with IP address**: 10.70.X.X**
- Set TCP Port: **5060**
- Click **Finish**

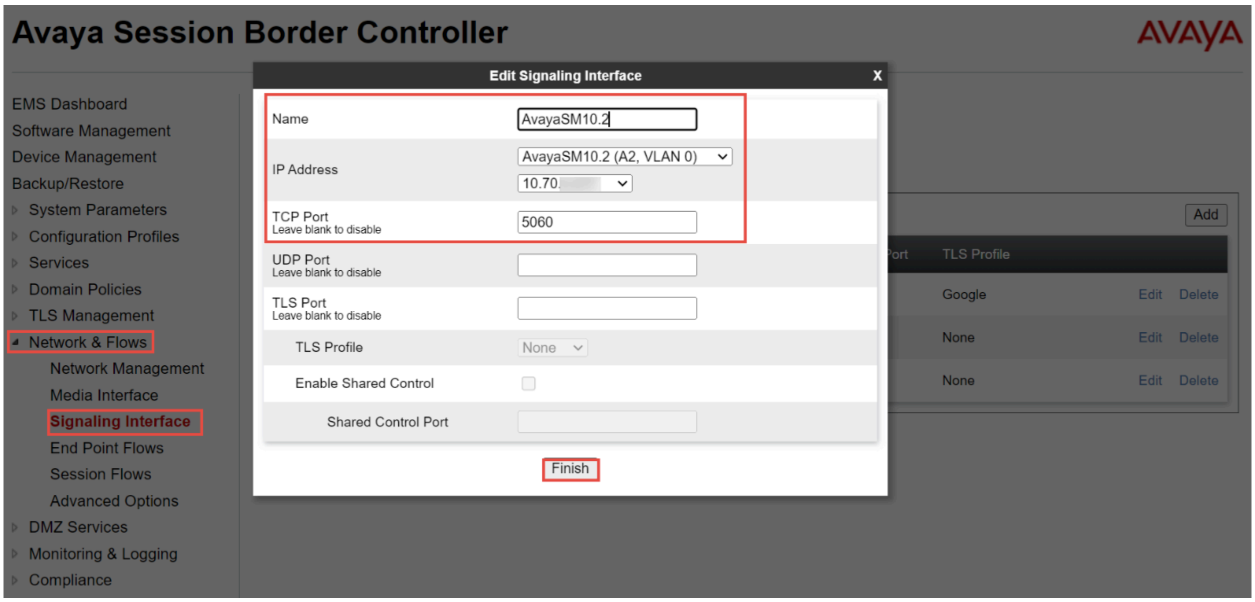


Figure 63: Signaling Interface Facing Avaya Aura SM

Signaling Interface for **Google CES**

- Set Name: **Google_SI** is given for the interface facing **Google CES**
- Set IP Address: **Google (B1, VLAN 0),** with IP address**: 192.65.X.X**
- Set TLS Port: **5061**
- Select TLS profile: **Google.** Refer Section 7.4.17
- Click **Finish**

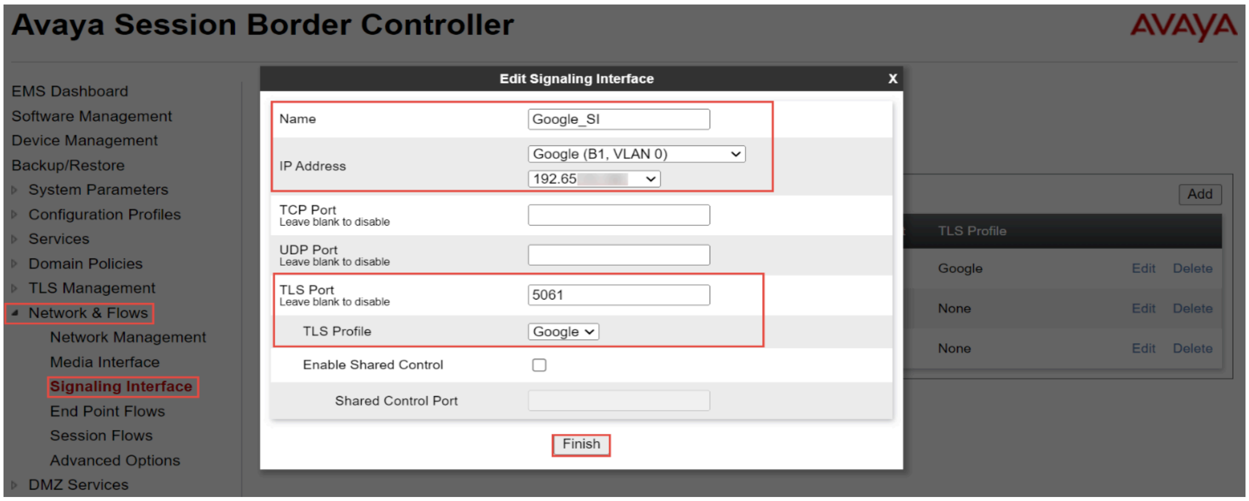


Figure 64: Signaling Interface Facing Google CES

Signaling Interface for **PSTN Gateway**

- Set Name: **PSTNGW** is given for the interface facing **Avaya Aura SM**
- Set IP Address: **PSTNGW (B2, VLAN 0)** with IP address: **10.64.X.X**
- Set TCP Port: **5060**
- Click **Finish**

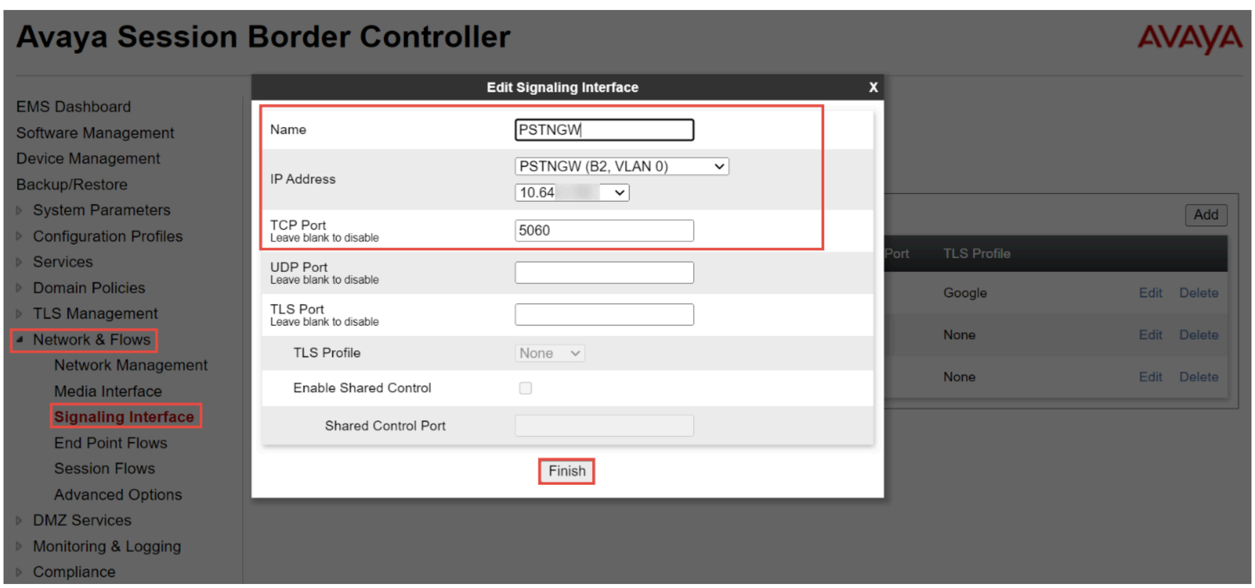


Figure 65: Signaling Interface Facing PSTN Gateway

7.4.16 End Point Flow

End Point Flow for **PSTN Gateway**

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

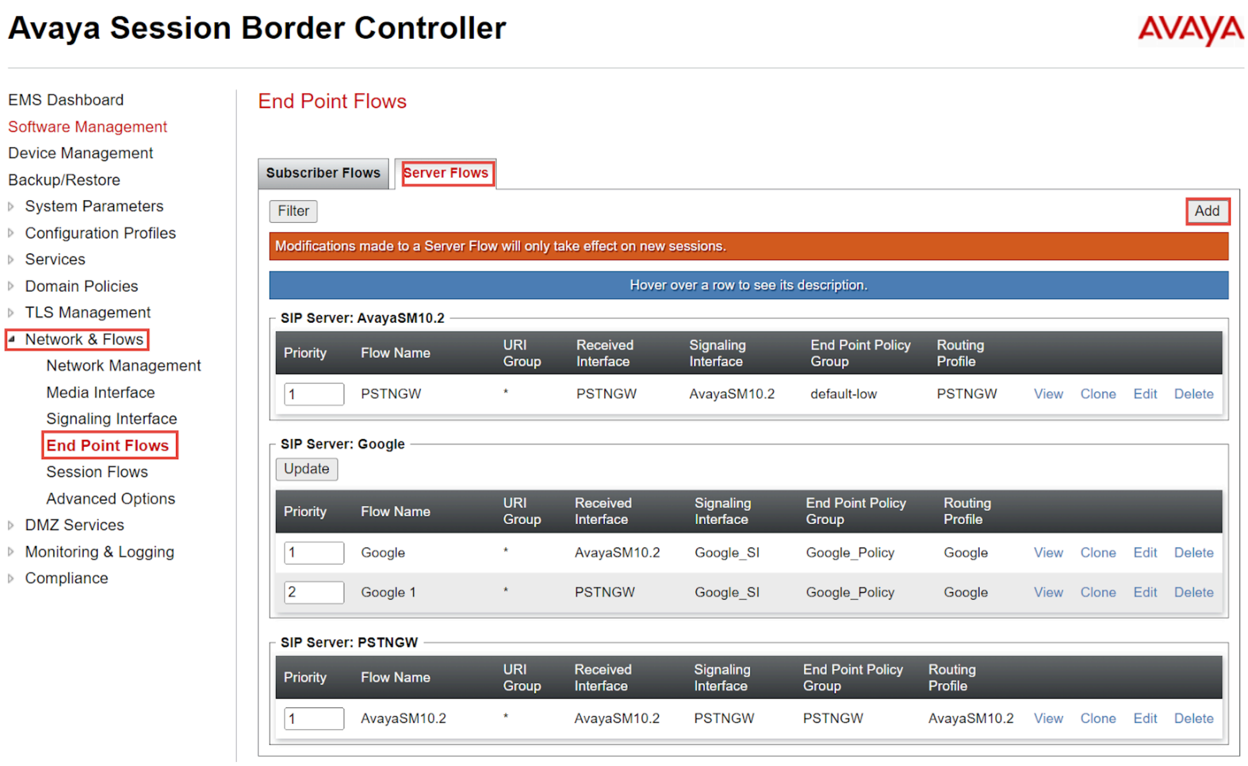


Figure 66:Server Flows

- Set Flow Name: **PSTNGW**
- Select SIP Server Profile: **AvayaSM10.2**
- Select Received Interface: **PSTNGW**
- Select Signaling Interface: **AvayaSM10.2**
- Select Media Interface: **AvayaSM10.2**
- Select Routing Profile: **PSTNGW**
- Select Topology Hiding Profile: **AvayaSM10.2**
- Click **Finish**

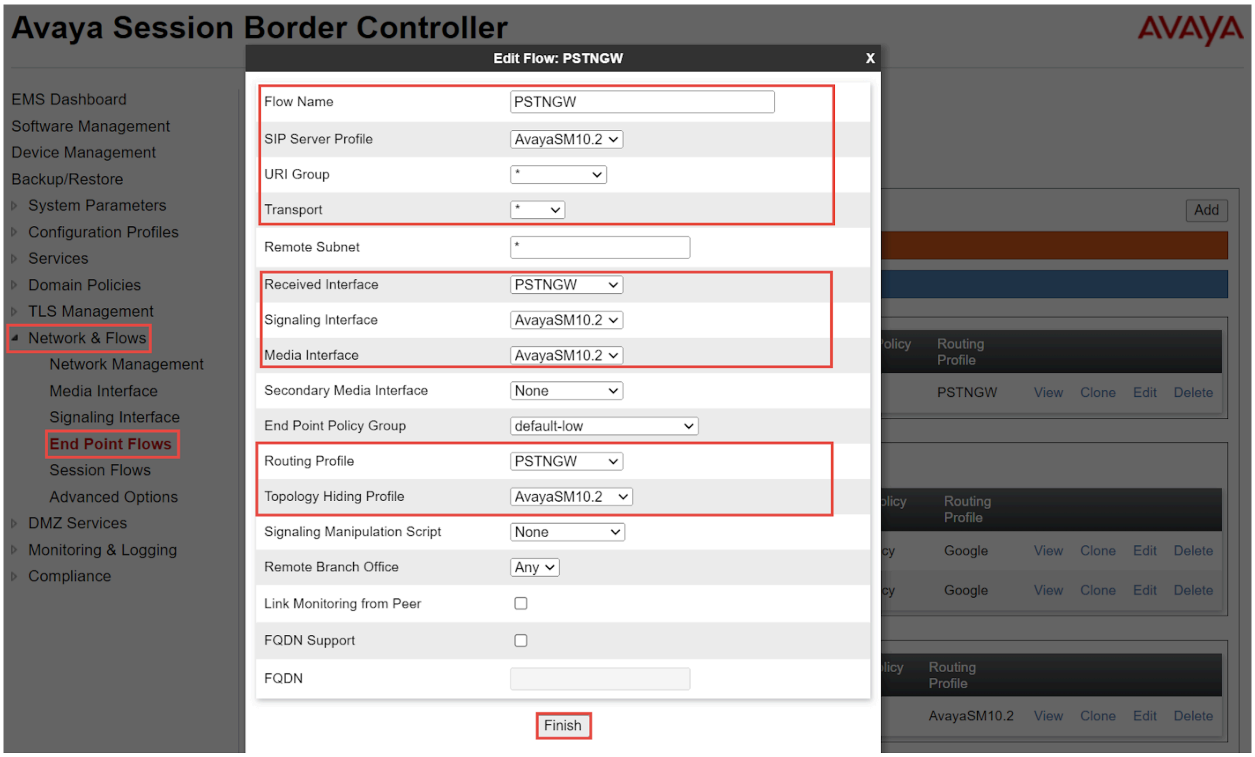
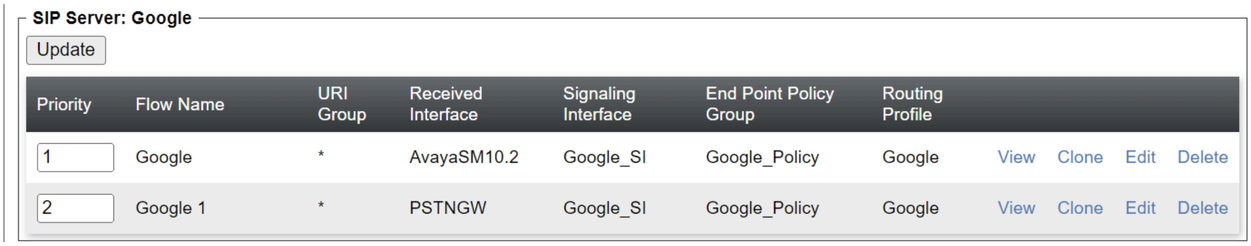


Figure 67: Server Flow for PSTN Gateway

End point flow for **Google CES**



SIP Server: Google

Update

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Google	*	AvayaSM10.2	Google_SI	Google_Policy	Google	View	Clone	Edit	Delete
2	Google 1	*	PSTNGW	Google_SI	Google_Policy	Google	View	Clone	Edit	Delete

Figure 68: Server Flow for Google CES

- Set Flow Name: **Google**
- Select SIP Server Profile: **Google**
- Select Received Interface: **AvayaSM10.2**
- Select Signaling Interface: **Google_SI**
- Select Media Interface: **Google_MI**
- Select End Point Policy Group: **Google_Policy**
- Select Routing Profile: **Google**
- Select Topology Hiding Profile: **Google**
- Select Signaling Manipulation script: **Google**
- Click **Finish**

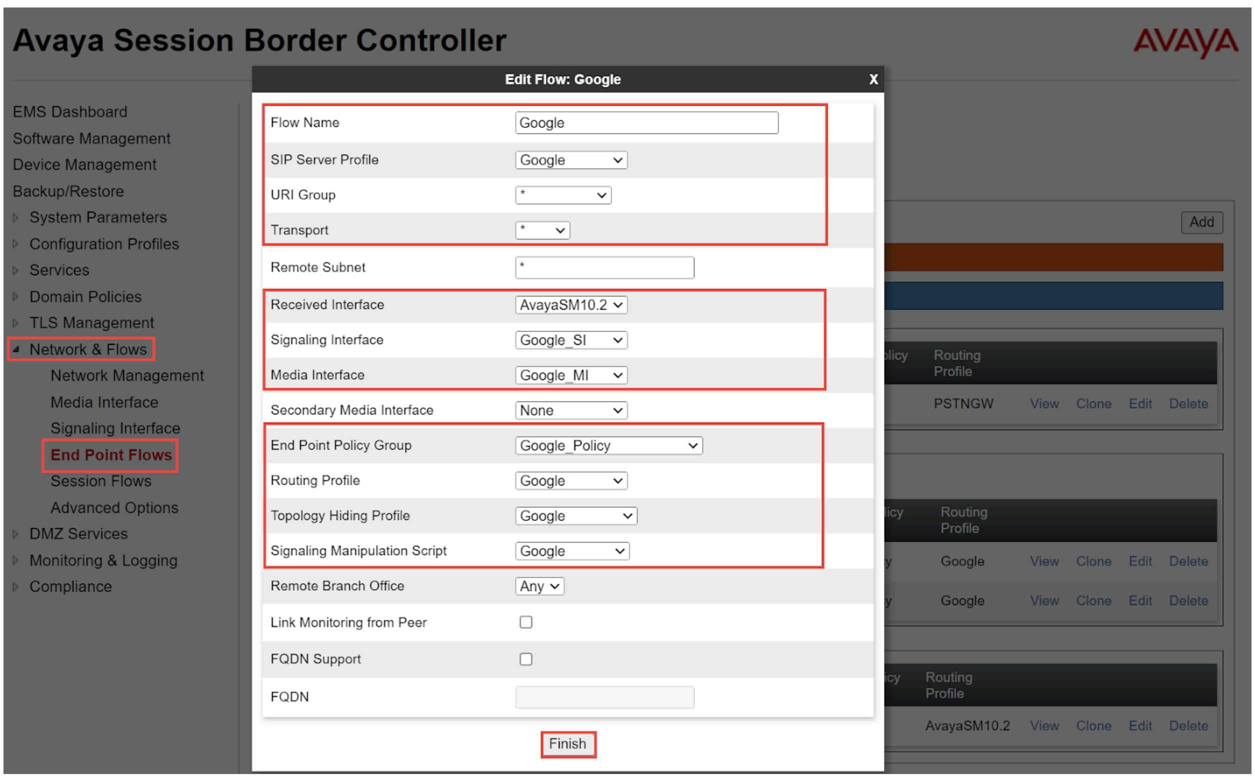


Figure 69: Server Flow for Google CES (Cont.)

- Set Flow Name: **Google 1**
- Select SIP Server Profile: **Google**
- Select Received Interface: **PSTNGW**
- Select Signaling Interface: **Google_SI**
- Select Media Interface: **Google_MI**
- Select End Point Policy Group: **Google_Policy**
- Select Routing Profile: **Google**
- Select Topology Hiding Profile: **Google**
- Select Signaling Manipulation script: **Google**
- Click **Finish**

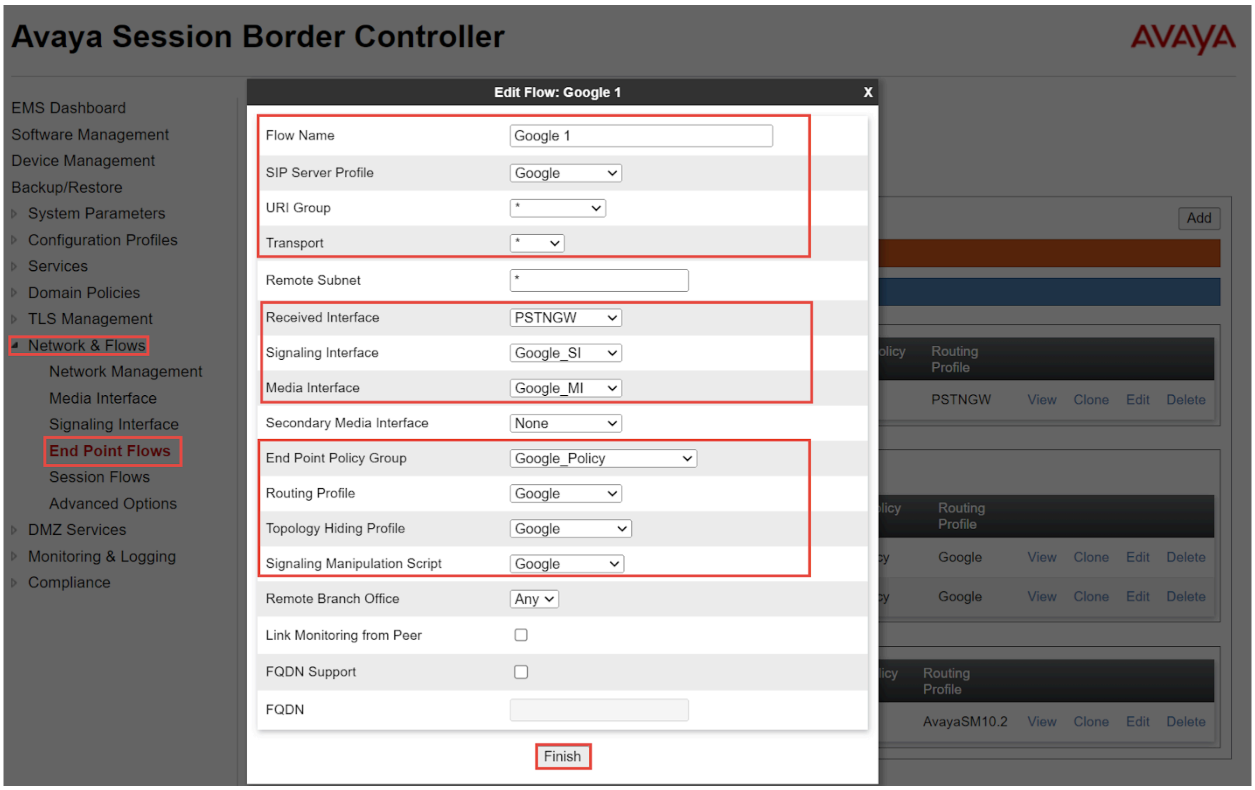


Figure 70: Server Flow for Google CES (Cont.)

End point flow for **Avaya Aura SM**

SIP Server: PSTNGW

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	AvayaSM10.2	*	AvayaSM10.2	PSTNGW	PSTNGW	AvayaSM10.2	View	Clone	Edit	Delete

Figure 71: Server Flow for Avaya Aura SM

- Set Flow Name: **AvayaSM10.2**
- Select SIP Server Profile: **PSTNGW**
- Select Received Interface: **AvayaSM10.2**
- Select Signaling Interface: **PSTNGW**
- Select Media Interface: **PSTNGW**
- Select End Point Policy Group: **PSTNGW**
- Select Routing Profile: **AvayaSM10.2**
- Select Topology Hiding Profile: **PSTNGW**
- Click **Finish**

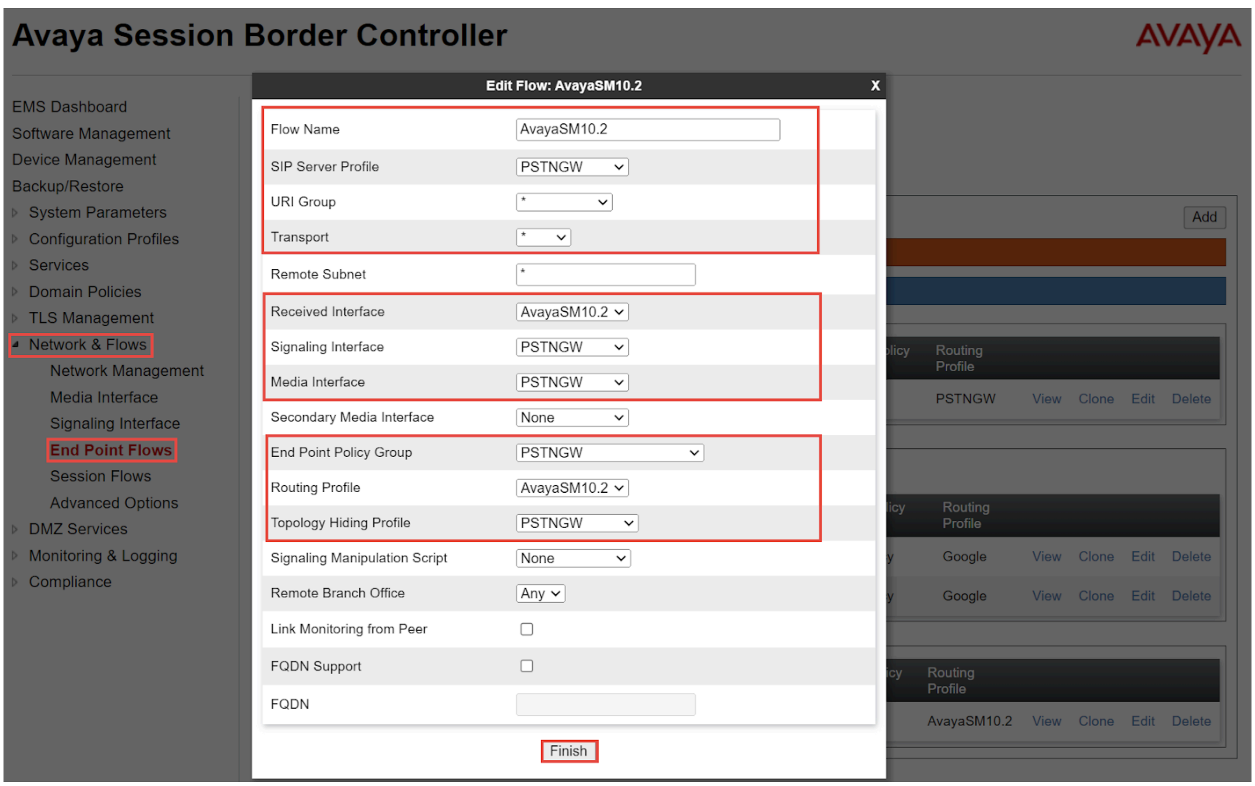


Figure 72: Server Flow for Avaya Aura SM (Cont.)

7.4.17 TLS Configuration

Configure TLS management for Google CES

- Navigate: **TLS Management** **Certificates**.
- Click Generate **CSR**

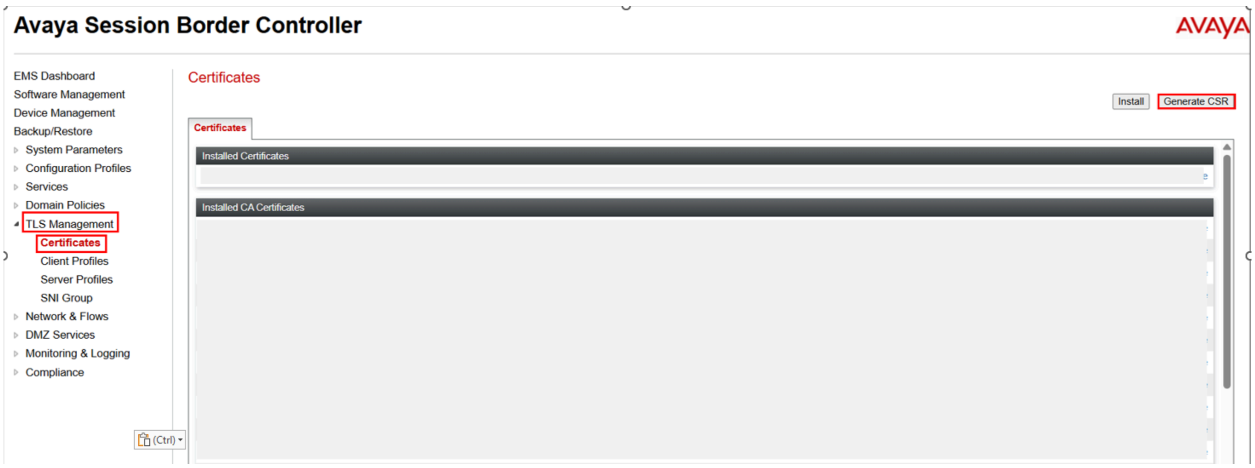


Figure 73: Generate CSR

- 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**
- Click **Generate CSR**

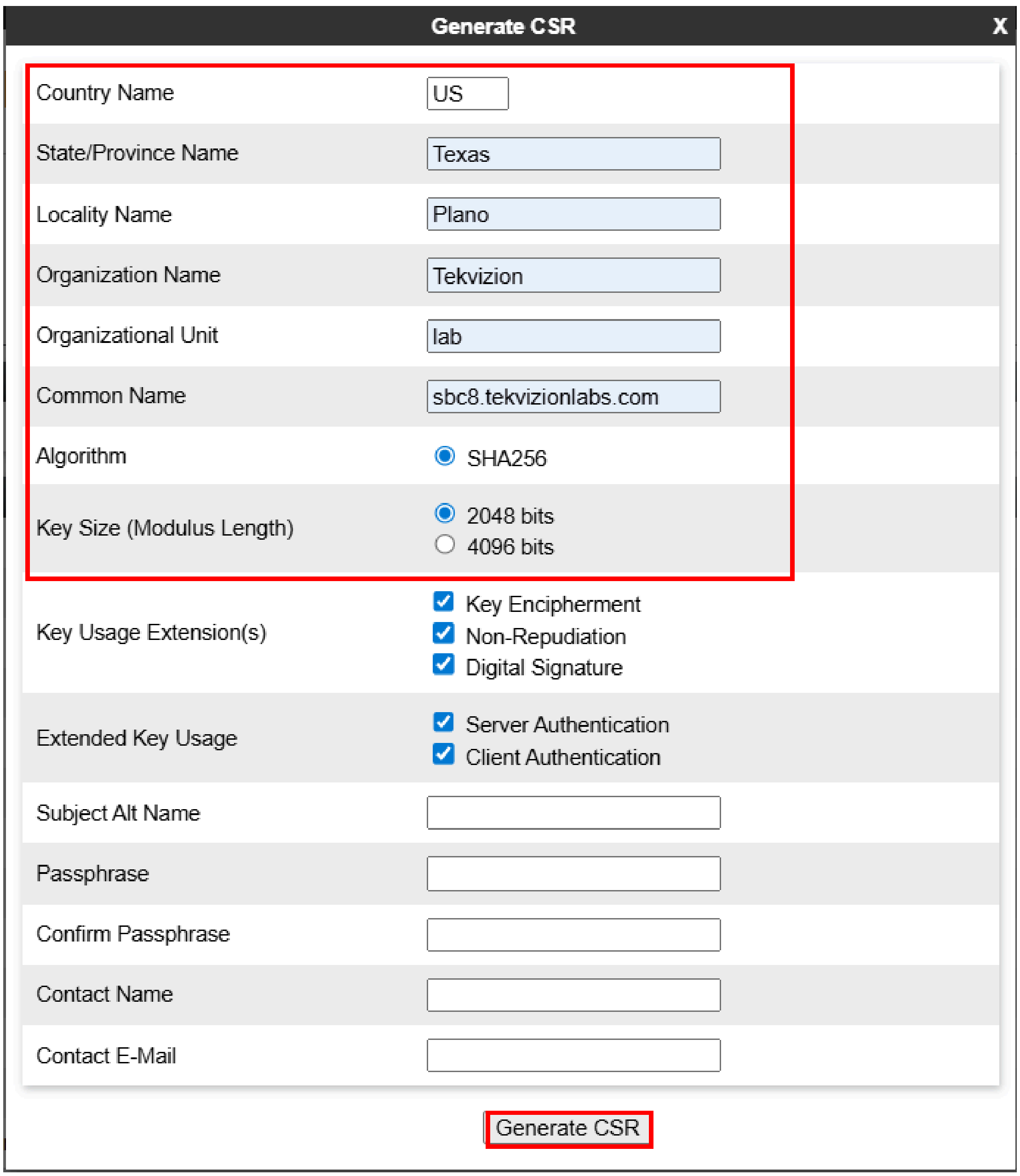


Figure 74: Generate CSR (Cont.)

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**

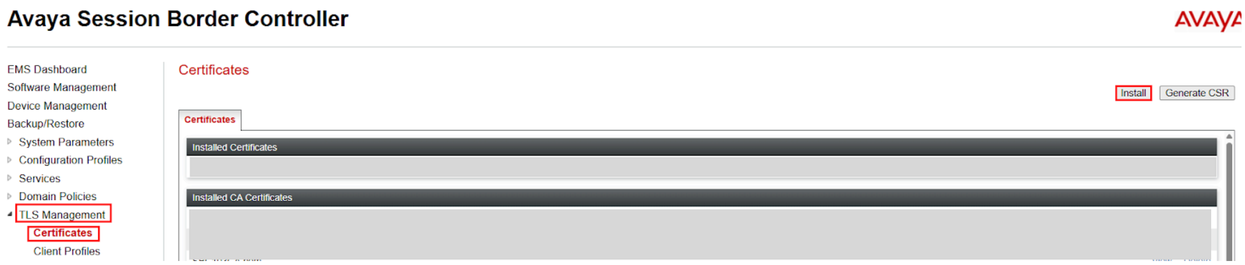


Figure 75: Certificate installation

- 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**

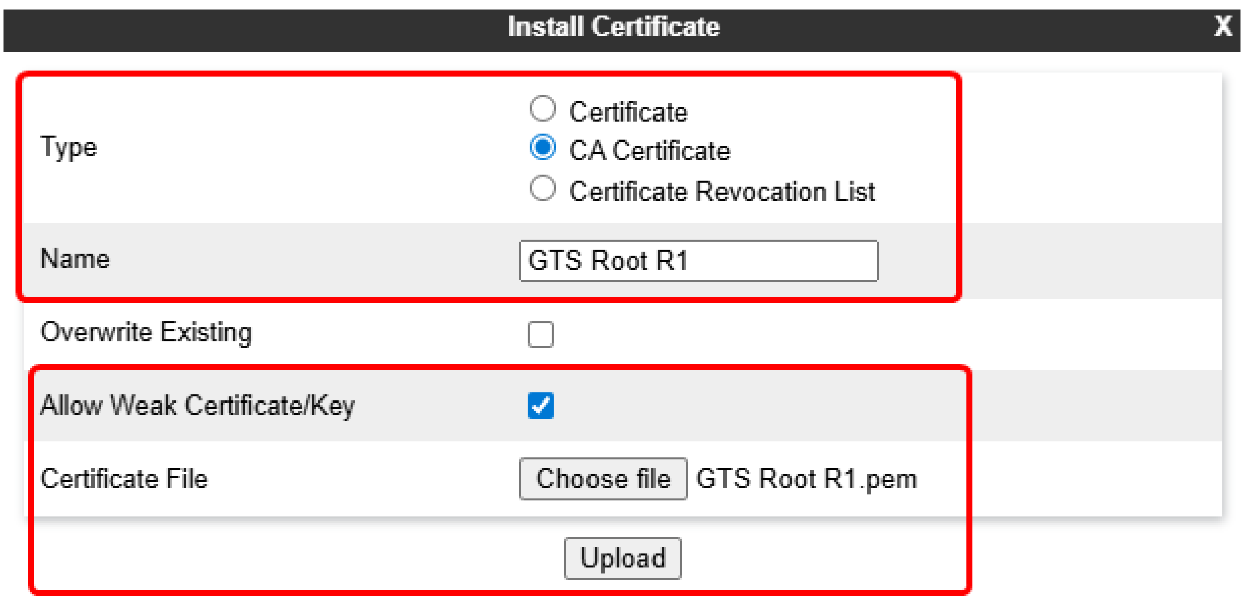


Figure 76: GTS Root R1

Upload SBC intermediate certificates:

- 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**

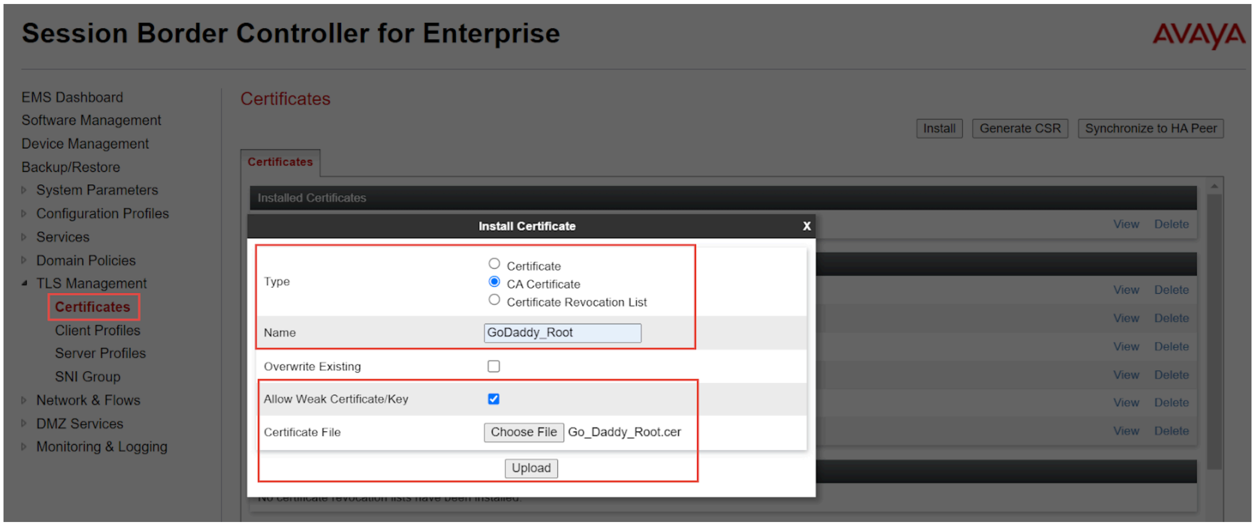


Figure 77: Upload GoDaddy Root CA

- 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**

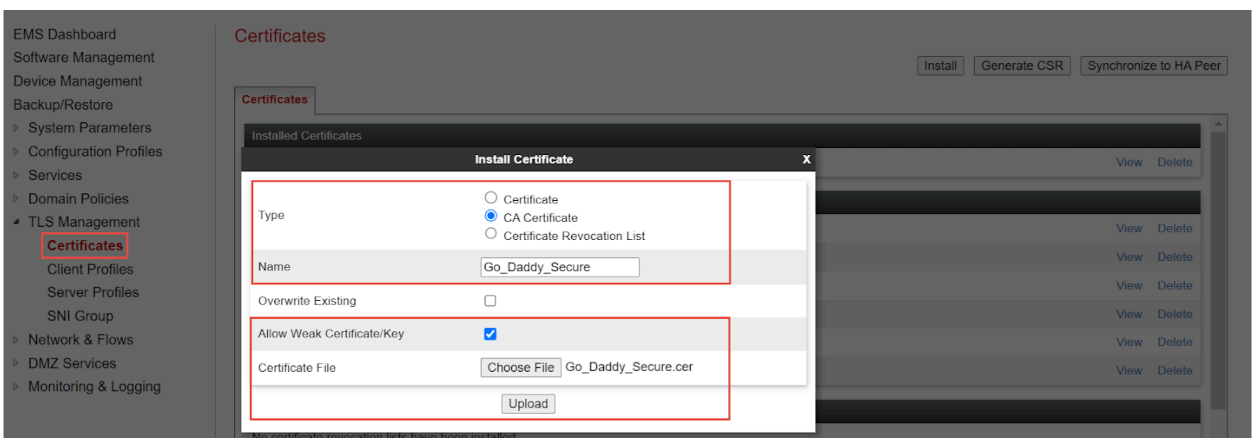


Figure 78: Upload GoDaddy Secure CA

- Navigate: **TLS management** □ **Certificates**. Click **Install**
- Set Type: Select **Certificate**
- Set Name: **sbc8**
- Set Allow Weak Certificate/Key: **Checked**
- Set Certificate File: Click Choose File to select **23xxxx.pem**
- Select Key File: **sbc8.key** from drop down
- Click **Upload**

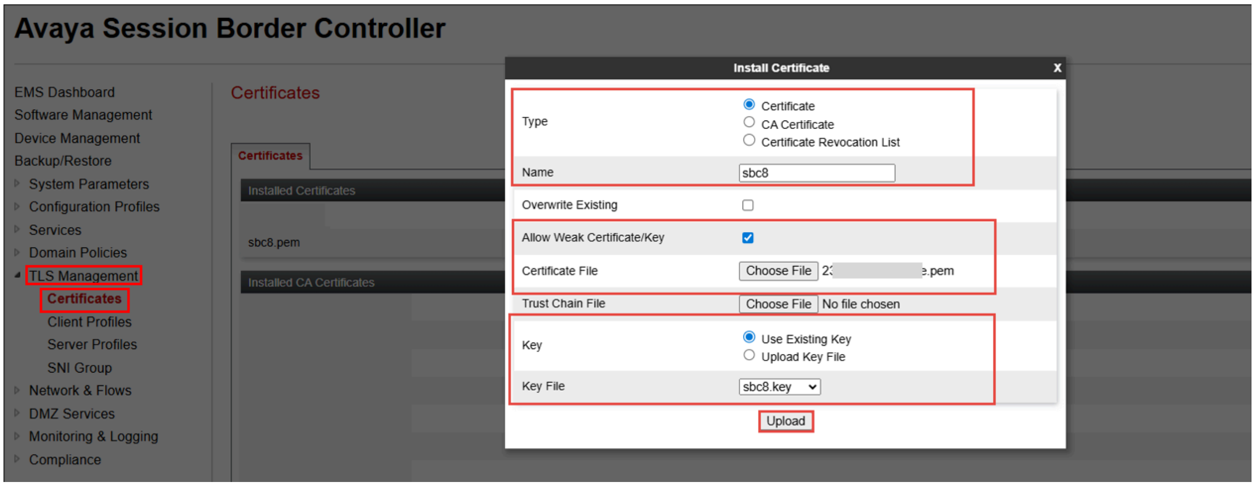


Figure 79: Upload Avaya SBC server Certificate

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**

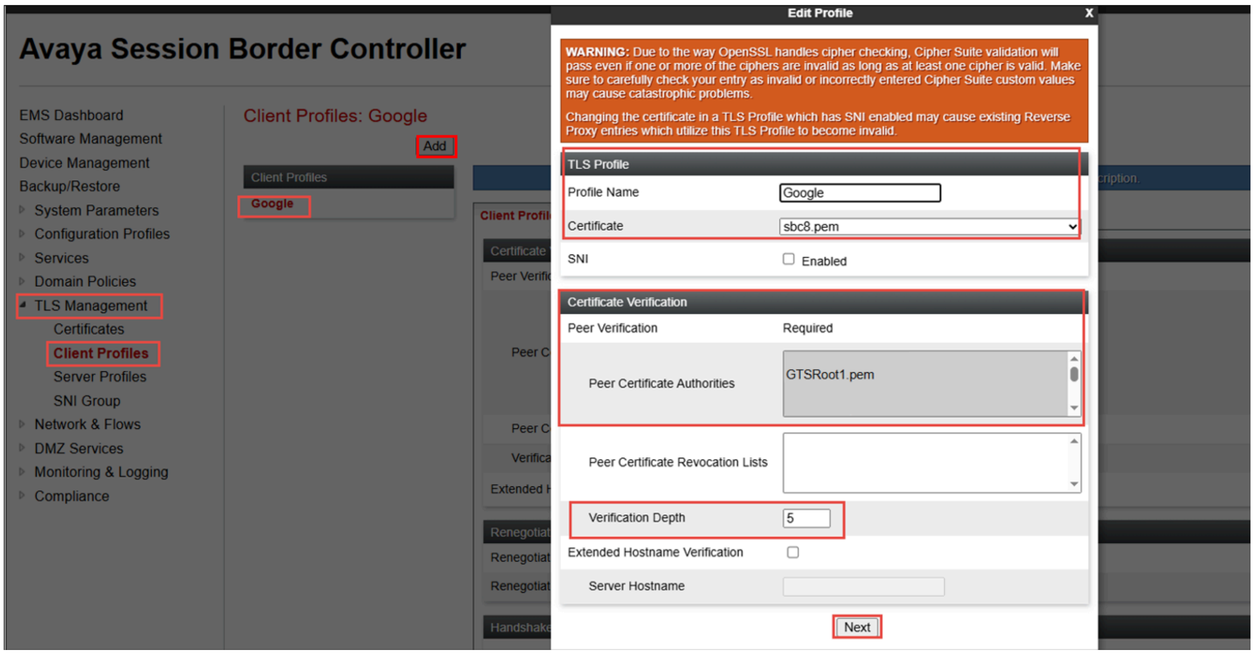


Figure 80: Client Profile Google CES

- Set Version: Select **TLS 1.2** version

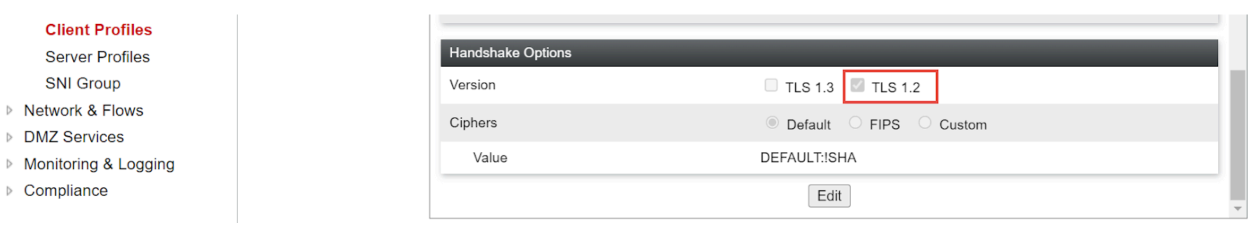


Figure 81: Client Profile Google CES (Cont.)

Server Profile for **Google CES**

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

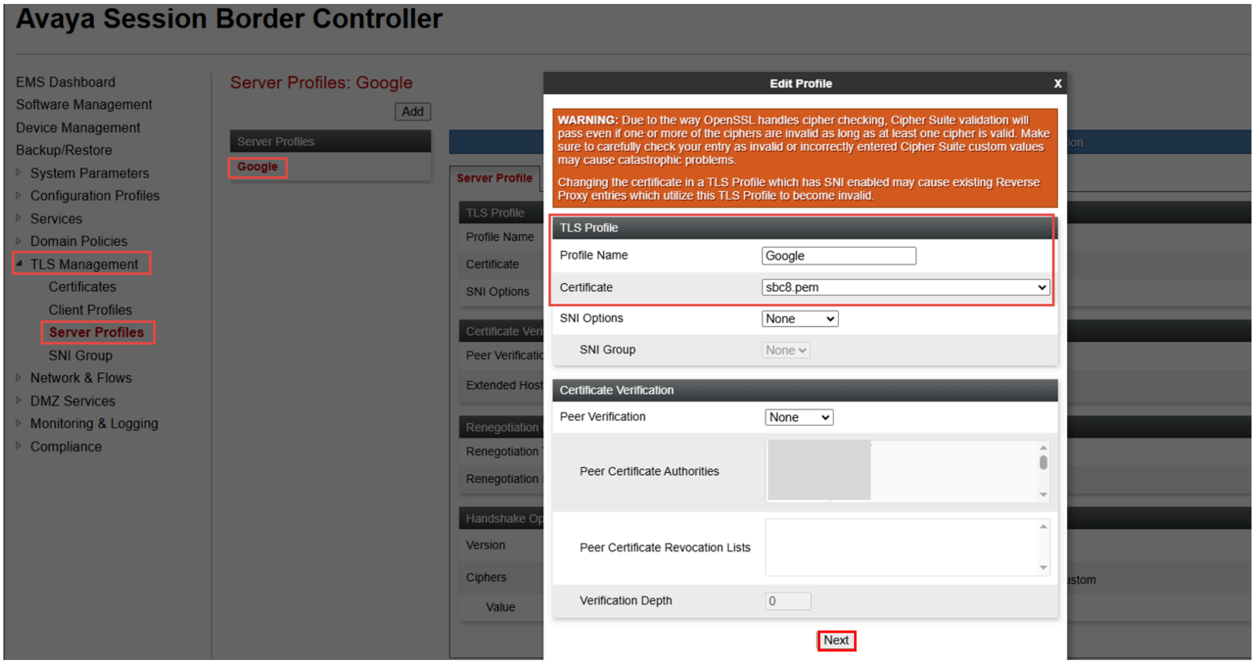


Figure 82: Server Profile towards Google CES

- Set Version: Select **TLS 1.2** versions

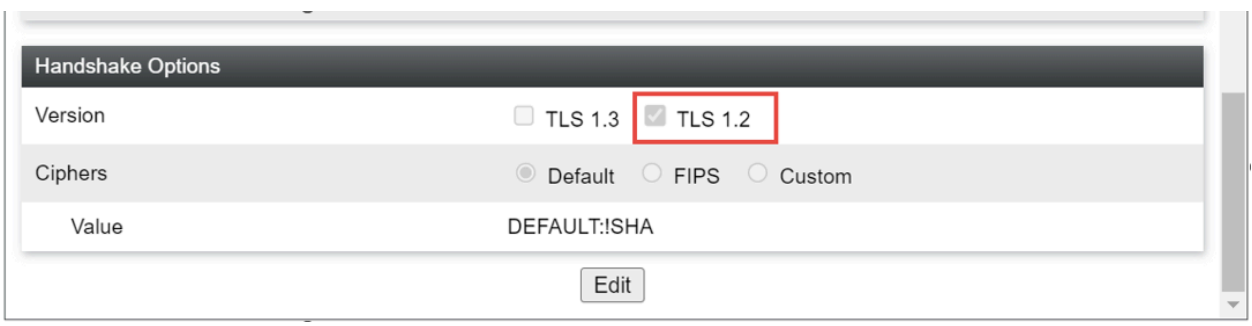


Figure 83: Server Profile towards Google CES (Cont.)

8 SIP INVITE To Google CES

8.1 SIP INVITE for SIPREC call

```
INVITE sip:+13614[redacted]@us.telephony.goog:5672;transport=tls SIP/2.0
From: "Pradeep Gopal" <sip:2145[redacted]@192.65.[redacted]>;tag=981EFC88-230A
To: <sip:+13614[redacted]@us.telephony.goog:5672;transport=tls>
CSeq: 4360 INVITE
Call-ID: a95d66761d308966e2bbc50433215103
Contact: <sip:192.65.[redacted]:5061;transport=tls>;+sip.src
Record-Route: <sip:192.65.[redacted]: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.[redacted]:5061;branch=z9hG4bK-s1632-000655365869-1--s1632-
Expires: 180
Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898[redacted]/conversations/
Sr_a95[redacted]03>;purpose=Goog-ContactCenter-Conversation
Require: siprec
Timestamp: 1758101718
Allow-Events: telephone-event
P-Asserted-Identity: "Pradeep Gopal" <sip:2145[redacted]@192.65.[redacted]>
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.[redacted]
s=SIP
c=IN IP4 192.65.[redacted]
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=senrecv*
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 84: SIPREC call

8.2 SIP INVITE for GTP call

```
INVITE sip:+131494[redacted]@us.telephony.goog:5672 SIP/2.0
From: "Kanitkar" <sip:+121455[redacted]@192.65.[redacted]>;tag=BC91BAE8-1784
To: <sip:+131494[redacted]@us.telephony.goog:5672>
CSeq: 101 INVITE
Call-ID: 2891181e459655649e5e9861e89f3a44
Contact: <sip:2145509018@192.65.[redacted]:5061;transport=tls>
Record-Route: <sip:192.65.[redacted]: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.[redacted]:5061;branch=z9hG4bK-s1632-001787678602-1--s
Via: SIP/2.0/TCP 10.64.[redacted]:5060;branch=z9hG4bK6BDD237C
Expires: 180
Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/ccai-389[redacted]/conversations/
Sr_28[redacted]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[redacted]@192.65.[redacted]>
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.[redacted]
s=SIP
c=IN IP4 192.65.[redacted]
t=0 0
m=audio 35168 RTP/SAVP 101 0 8 19
c=IN IP4 192.65.[redacted]
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 85: GTP call

9 Summary of Tests and Results

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
SBC Configuration Verification					
1	SBC Configuration Verification	TLS connection setup. SBC initiates TLS connection with CES	Successful 4way handshake with Google CES. Validate the right certificates are being negotiated. SBC should be loaded with GTSR1 cert for Google. SBC should also send the certificate chain when sending its cert.	PASSED	
2	SBC Configuration Verification	TCP Keep Alive. SBC will perform monitoring checks by attempting TCP Keep Alive to ensure Network Connectivity	Successful 3way handshake and thereafter termination	PASSED	TCP Keep-alive packets are sent to the SIPREC Trunk
3	SBC Configuration Verification	TCP link is persistent. Establish calls, send multiple calls that should all use the same TCP transport connection	Persistent TCP connection, we should establish a single connection and multiplex all calls over that connection.	PASSED	
4	SBC Configuration Verification	Session Timer support. SBC should be initiator for the Session Refresh timer using Update or re-INVITE	Every 900 secs the SBC should refresh the SIP session.	PASSED	Avaya SBC does not send session refresh re-INVITE. However, Google sends refresh sessions every 60 minutes

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
					using re-INVITE
5	SBC Configuration Verification	SIP Header Manipulation (Call-Info header)	Validate if the Google requested header manipulation is present in the SIP INVITE. Ensure every SDP media has a label.	PASSED	
6	SBC Configuration Verification	*SBCs may need further Header manipulations based on SIP stack constraints. Verify required manipulation are added in SBC to support Google CES Example: FROM, TO header manipulations HOST part change in headers etc.	All signaling in e.164 format	PASSED	
7	SBC Configuration Verification	SDES for SRTP. Configure the SDES parameters for crypto negotiation for the BYOT trunk	Validate the crypto is successfully negotiated and media is encrypted. All SBCs should support SDES for media encryption.	PASSED	
8	SBC Configuration Verification	DTLS for Media Encryption. Configure the DTLS parameters for crypto negotiation for the BYOT trunk, certificate for DTLS	Validate the crypto is successfully negotiated and media is encrypted.	NOT SUPPORTED	Avaya SBC does not support DTLS

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		must be self-signed by the SBC.			
Inbound					
9	SIP OPTIONS	SBC send SIP options every 60 seconds	Verify SBC sends SIP OPTIONS every 60 seconds and responds with 200 OK	PASSED	
10	Inbound	Inbound call: Calling Party disconnects the call. Inbound siprec call, ensure recording are present, disconnect call from calling party and confirm proper disconnect	Verify Call is established with audio and transcripts from both participants Verify call is disconnected properly	PASSED	
11	Inbound	Inbound call: Called Party disconnects the call. Inbound siprec call, ensure recording are present, disconnect call from called party and confirm proper disconnect	Verify Call is established with audio and transcripts from both participants Verify call is disconnected properly	PASSED	
12	Inbound	Long duration call-Outbound Call-1 hour max. Long duration siprec call	Ensure siprec calls stay up for an hour, confirm transcripts are present for entire duration	PASSED	Avaya SBC does not send session refresh re-INVITE. However, Google sends session refresh every 60 minutes using re-INVITE

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
13	Inbound	Long duration hold and resume (wait until session audit\session refresh occurs from DUT). Long duration siprec call, have the call placed on hold by agent, have call resume. Have customer place on hold then have call resume.	Call is connected, we have two active streams, confirm once a stream goes on hold, we receive corresponding signaling events, and that we no longer record transcripts for the participant on hold.	PASSED	Avaya SBC does not send session refresh re-INVITE. However, Google sends session refresh every 60 minutes using re-INVITE
14	Inbound	Handling Error codes 603 decline. User A Calls PSTN A PSTN A rejects the incoming call	Verify SBC handles Call rejected properly	PASSED	
15	Inbound	Inbound call hold scenarios. Call starts out inactive for both participants, session moves to active	Validate if media is present when expected, confirm signaling events modify sdp properly, once call is move to active validate media and transcripts	PASSED	
16	Inbound	Inbound call hold scenarios. call starts out as active for both participants, session move to inactive, and transitions back to active	Validate if media is present when expected, confirm signaling events modify sdp properly, once call is moved to active validate media and transcripts	PASSED	
17	Inbound	Update. Validate that update sent prior to call establishment do not contain SDP	Validate that update prior to call establishment do	PASSED	Avaya SBC does not support UPDATE with SDP

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
			not contain SDP as expected		
18	Inbound	Update. Validate that updates post call establishment contain SDP to modify session	If SBC uses update to modify session, ensure SDP is included	NOT SUPPORTED	
19	Inbound	re-INVITES. Ensure re- INVITES that modify session include SDP	Ensure re-INVITES that modify session include SDP	PASSED	re-INVITE from Avaya SBC is sent to Google CES as part of session refresh, hold scenarios
20	Inbound	Codec negotiation. Ensure that g711 u-law is preferred codec	Ensure we can prioritize g711 as preferred codec, note where SBC configures preferred codec	PASSED	
21	Inbound	3 way conference. Determine requirements, record all leg.	Determine requirements, record all legs	PASSED	
22	Inbound	CES cloud project setup. Establish CES cloud project, provision the project with a GTP phone number for access (Create conversations/participants on the fly through SIP headers)	Verify project is setup, functional test to confirm you can connect to the GTP access phone number	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
23	Inbound	CES cloud project setup. Establish CES cloud project, provision the project with a GTP phone number for access (Pre-creation of conversations/participants)	Verify project is setup, functional test to confirm you can connect to the GTP access phone number	NOT APPLICABLE	This test case is not applicable for call recording
24	Inbound	Consultative transfer. Consultative transfer from 1. PSTN > User1 > User2 2. PSTN > User1 > PSTN user2		PASSED	
25	Inbound	Blind transfer. Blind transfer from 1. PSTN > User1 > User2 2. PSTN > User1 > PSTN user2		PASSED	Avaya PBX does not support blind transfer. This test case is done by ringing transfer
26	Validate Provisioning of trunk using self service	Validate Provisioning of trunk using self service	Use documentation to build trunk using self-service model	PASSED	
27	Inbound	Inbound call hold scenarios using a-law	Validate if media is present when expected, confirm Signaling events modify sdp properly, once call is move to hold active validate media and transcripts	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
28	Inbound	Inbound call: Called Party disconnects the call. using a a-law codec	"Verify Call is established with audio and transcripts from both participants Verify call is disconnected properly Validate media stays in region"	PASSED	
29	Inbound	Long duration call-Outbound Call- 1 hour max using a-law codec	Ensure siprec calls stay up for an hour, confirm transcripts are present for entire duration.	PASSED	Avaya SBC does not send refresh re-INVITE. However, Google sends session refresh every 60 minutes using re-INVITE
30	Inbound	Inbound call: Configure trunk in non default region,	Verify Call is established with audio and transcripts from both participants Verify call is disconnected properly Validate media stays in region	PASSED	Testing is conducted in the US region

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
31	Outbound	Participant Labels test	Configure call info header to specify roles, ensure the media streams align, Frist media stream HUMAN_AGENT role and Second is END_USER.	PASSED	When the roles are set to "HUMAN AGENT" and "END USER," (Call-Info<http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898XX/conversations/Sr_XXX?roles=HUMAN_AGENT,END_USER>;purpose=Goog-ContactCenter-Conversation) the transcript shows the first media stream with the participation role as "HUMAN AGENT," followed by "END USER." The transcript indicates that HUMAN AGENT was listed first, followed by the END USER, in 6 out of 10 attempts.
32	Inbound	DTLS test		NOT SUPPORTED	
33	Inbound	Conference TEST	Determine requirements, record all legs	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
34	Inbound	Validate Call recording	Verify call recording is recorded throughout the call	PASSED	