Configuration Guide for Google CES Agent Handoff Using AudioCodes VE SBC 7.60A.100.022

Table of Contents

1 Audience

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

1.1 Introduction

This configuration guide describes configuration steps for **Google CES Agent Handoff** using **AudioCodes VE SBC 7.60A.100.022**.

1.1.1 TekVizionLabs

TekVizionLabs™ 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 Centre, 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 AudioCodes VE SBC 7.60A.100.022.

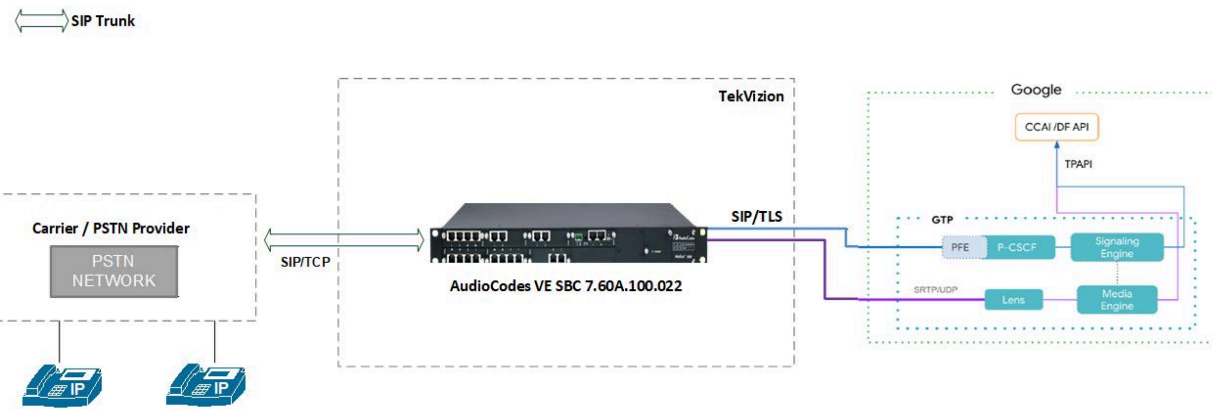


Figure 1: SIP Trunk Lab Reference Network

The lab network consists of the following components:

- Google CES cloud Environment
- AudioCodes VE SBC 7.60A.100.022
- PSTN Gateway

3 Hardware Components

- AudioCodes VE SBC running on ESXi host version 6.7.0

4 Software Requirements

- AudioCodes VE SBC version: 7.60A.100.022

5 Certified AudioCodes VE SBC Version

Table 1 – AudioCodes VE SBC Versions

Google CES - Verified version	
AudioCodes VE SBC	7.60A.100.022

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 AudioCodes VE SBC.

Table 2 – AudioCodes VE SBC Configuration Steps

Step	Description	Reference
Step 1	Network Interface IP	Section 7.4.1
Step 2	Configure TLS Context for Google CES	Section 7.4.2
Step 3	Configure Media Realms	Section 7.4.3
Step 4	Configure SIP Signaling Interfaces	Section 7.4.4
Step 5	Configure Proxy Sets and Proxy Address	Section 7.4.5
Step 6	Configure Coders	Section 7.4.6
Step 7	Configure IP Profiles	Section 7.4.7
Step 8	Configure IP Groups	Section 7.4.8
Step 9	Configure Media Security	Section 7.4.9
Step 10	Configure IP to IP Call Routing	Section 7.4.10
Step 11	Configure Message Manipulation Rules	Section 7.4.11

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
AudioCodes VE 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 for troubleshooting Google CES API configuration for Agent Handoff.

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

7.4 AudioCodes VE SBC Configuration

The following is the configuration of AudioCodes VE SBC for Google CES Agent Handoff.

7.4.1 Network Interface IP

- Navigate to **SETUP** menu □ **IP NETWORK** tab □ **CORE ENTITIES** folder □ **IP Interfaces.**
- Configure IP Interfaces for PSTN Gateway and Google CES as shown below.

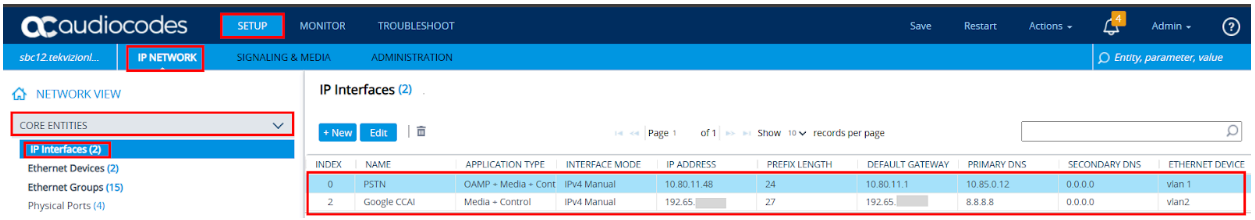


Figure 2: IP Interfaces

7.4.1.1 Configure LAN and WAN VLANs

- Navigate to **SETUP** menu □ **IP NETWORK** tab □ **CORE ENTITIES** folder □ **Ethernet Devices.**
- Configure VLANs for LAN and WAN interfaces as shown below.

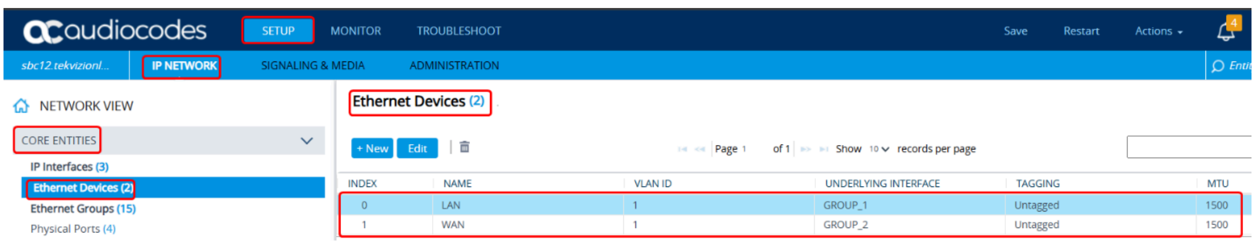


Figure 3: VLAN Configuration

7.4.1.2 Configure Network Interfaces

- Navigate to **SETUP** menu □ **IP NETWORK** tab □ **CORE ENTITIES** folder □ **IP Interfaces.**
- Configure the IP Network interfaces for PSTN Gateway and Google CES as shown below.

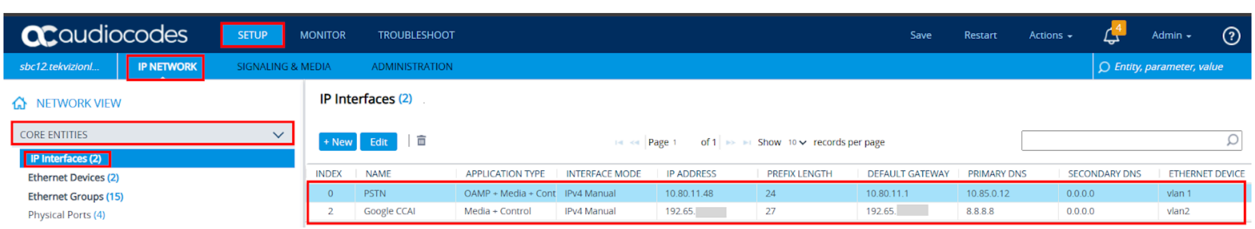


Figure 4: Network Interface Configuration

7.4.2 Configure TLS Context for Google CES

To establish a secure TLS connection between the AudioCodes VE SBC and Google CES, the TLS context must be configured accordingly.

7.4.2.1 Create a TLS Context for Google CES

- Navigate to **SETUP** menu □ **IP NETWORK** tab □ **SECURITY** folder □ **TLS Contexts.**
- Configure TLS context for Google CES as shown below.

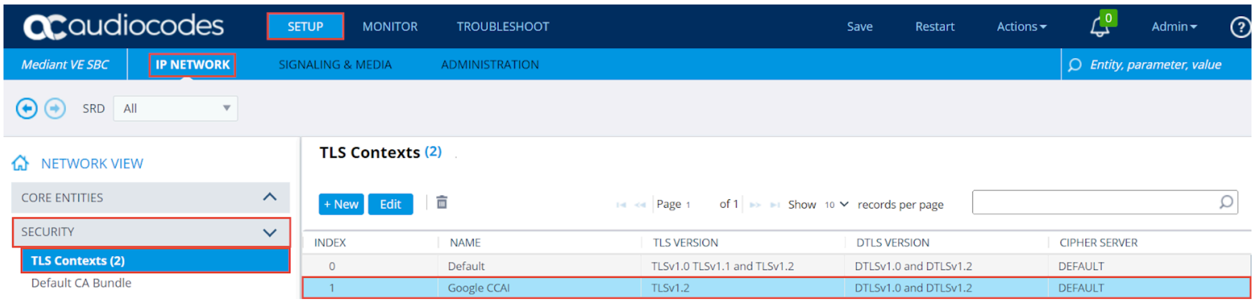


Figure 5: TLS Context for Google CES

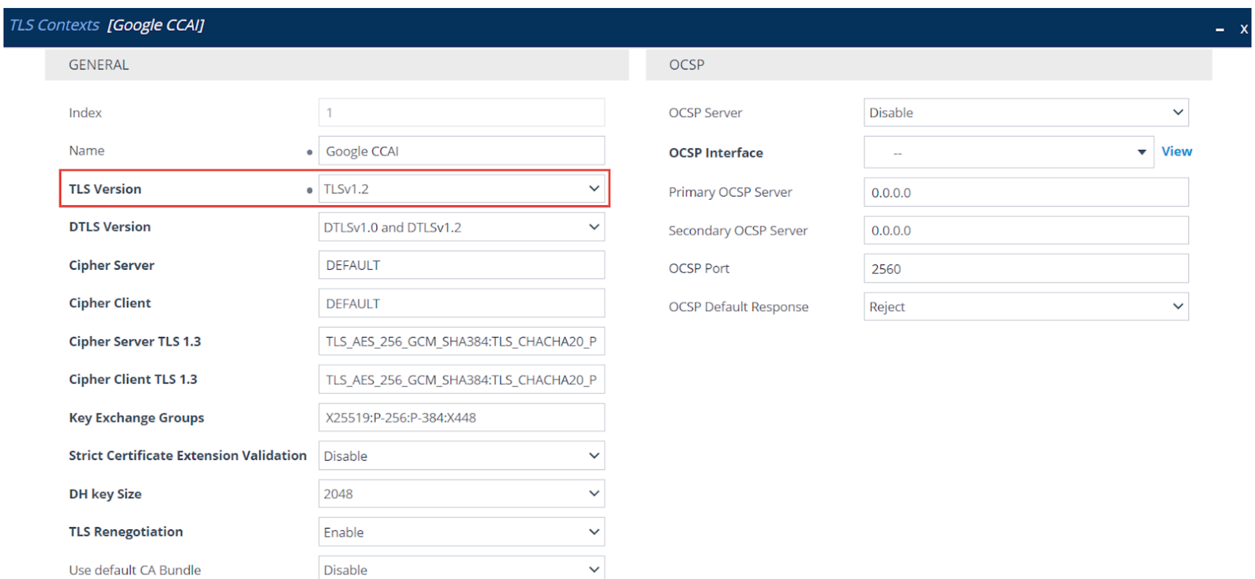


Figure 6: TLS Context for Google CES (Cont.)

7.4.2.2 Generate a CSR and Obtain the Certificate from a Supported CA

- Navigate to **SETUP** menu □ **IP NETWORK** tab □ **SECURITY** folder □ **TLS Contexts.**
- In the TLS context page, select the **Google CES** TLS context index row and click on **Change Certificate** option.

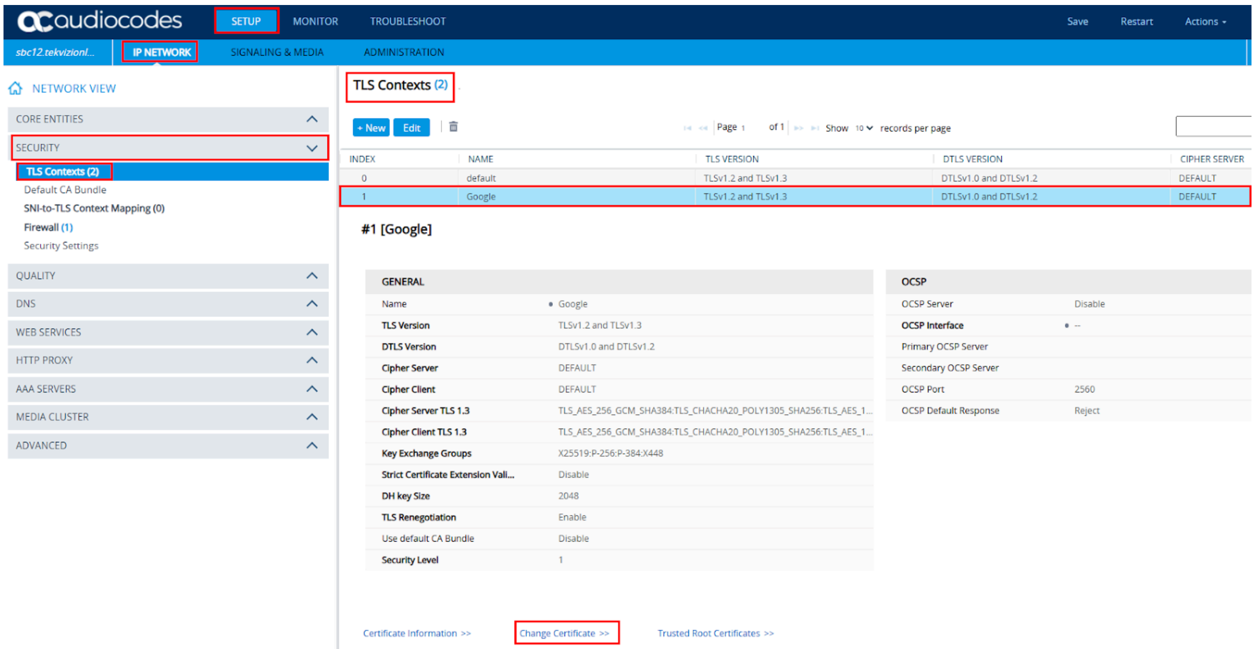


Figure 7: Change Certificate for CSR Generation

- Fill the required details in the Change Certificates link such as 'Common Name'(CN), Private Key Format, Private key size, and Click Generate Private Key.

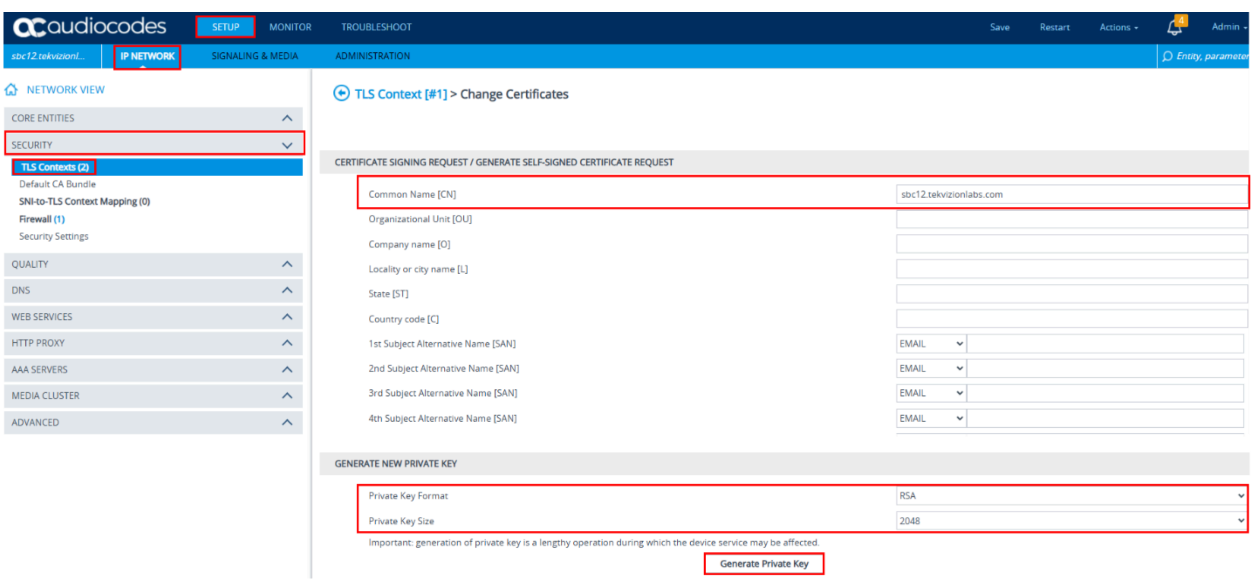


Figure 8: CSR Generation for Google CES TLS Context

- Click on “Create CSR” to generate the CSR and get it signed by certificate authority.

Figure 9: CSR Generation for Google CES TLS Context (Cont.)

7.4.2.3 Upload the AudioCodes VE SBC Certificate

- Navigate to **SETUP** menu **IP NETWORK** tab **SECURITY** folder **TLS Contexts.**
- In the TLS context page, select the Google CES TLS context index row and click on **Change Certificate** option.
- Scroll further down and opt for **Load Device Certificate File** to upload the AudioCodes VE SBC certificate to it.

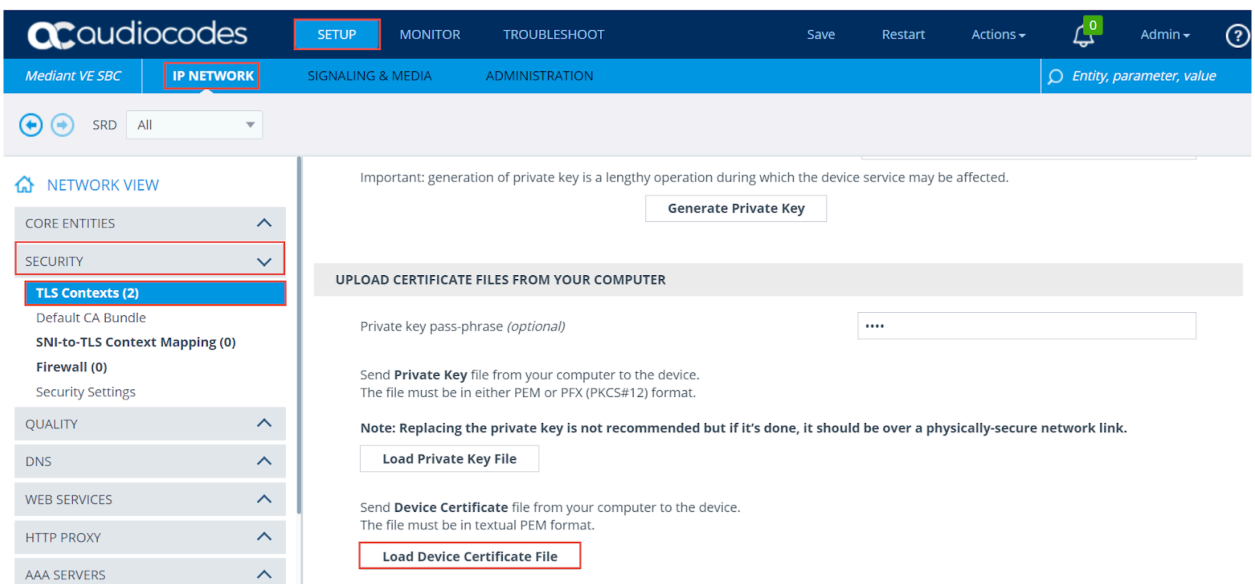


Figure 10: SBC Certificate Upload

Upload Intermediate Certificates:

- Navigate to **SETUP** menu **IP NETWORK** tab **SECURITY** folder **TLS Contexts**
- In the TLS context page, select the Google CES TLS context index row and click on **Trusted Root Certificates** option.

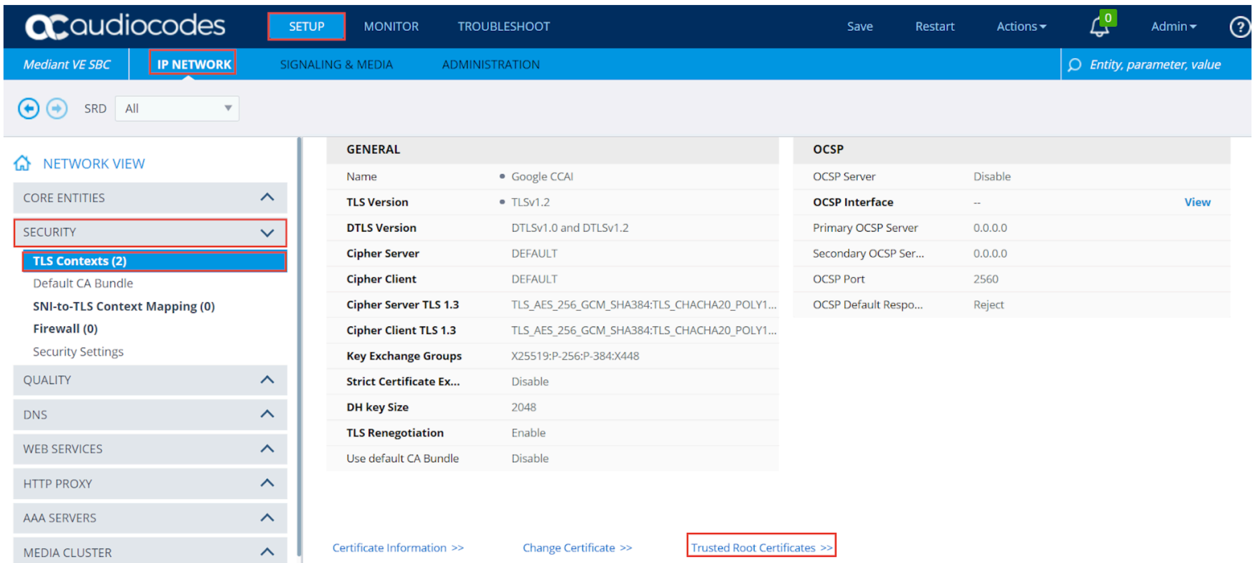


Figure 11: Trusted Root Certificates

- Within the Trusted Root Certificates page, click the Import button and import the Intermediate Certificates.

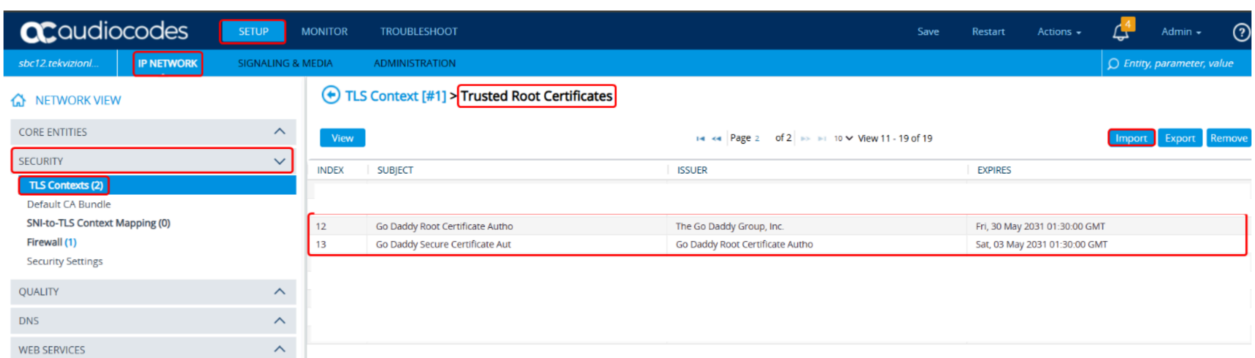


Figure 12: Import Trusted Root Certificates

7.4.2.4 Upload Google Trusted Root Certificate

- Download the Google Root Certificates from the following link https://pki.goog/repository/ and select the label GTS Root R1 only.
- Navigate to **SETUP** menu ▯ **IP NETWORK** tab ▯ **SECURITY** folder ▯ **TLS Contexts.**
- In the TLS context page, select the Google CES TLS context index row and click on the **Trusted Root Certificates** option.
- Within the Trusted Root Certificates page, click the **Import** button and load Google Root Certificate as shown below.

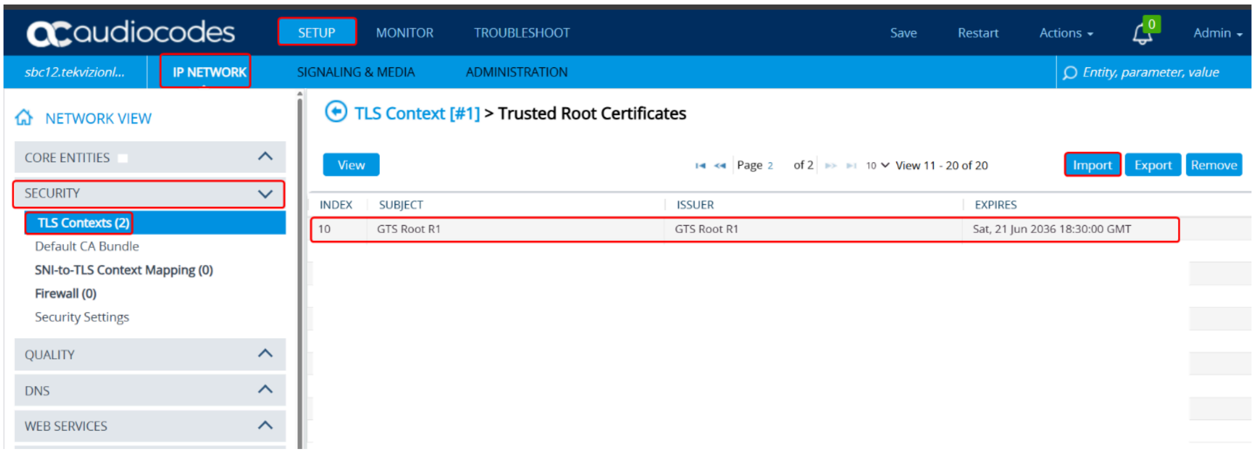


Figure : GTS Root R1 Certificate

7.4.3 Configure Media Realms

- Navigate to **SETUP** meu ▯ **SIGNALING & MEDIA** tab ▯ **CORE ENTITIES** folder ▯ **Media Realms.**
- Configure Media Realms for PSTN Gateway and Google CES as shown below.

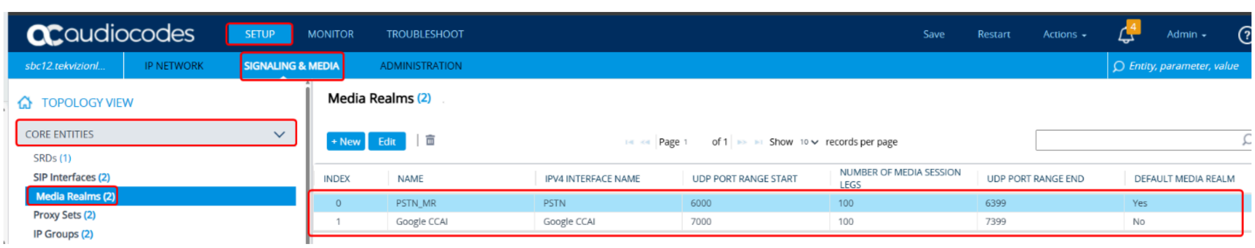


Figure : Configure Media Realms

7.4.4 Configure SIP Signaling Interfaces

- Navigate to **SETUP** menu □ **SIGNALING & MEDIA** tab □ **CORE ENTITIES** folder □ **SIP Interfaces.**
- Configure SIP Signaling Interfaces for PSTN Gateway and Google CES.

Google CES:

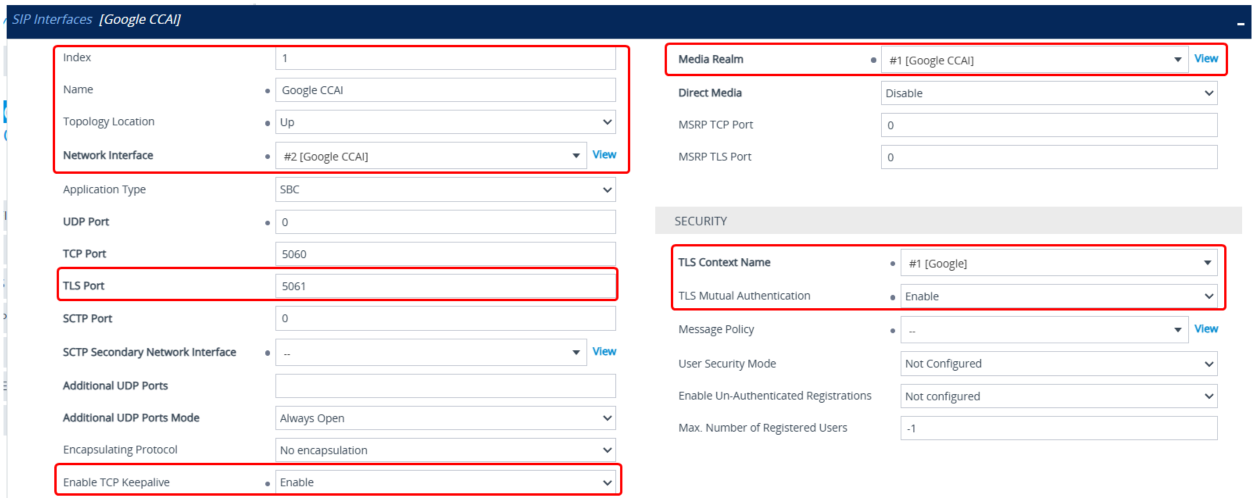


Figure 15: SIP Signaling Interfaces for Google CES

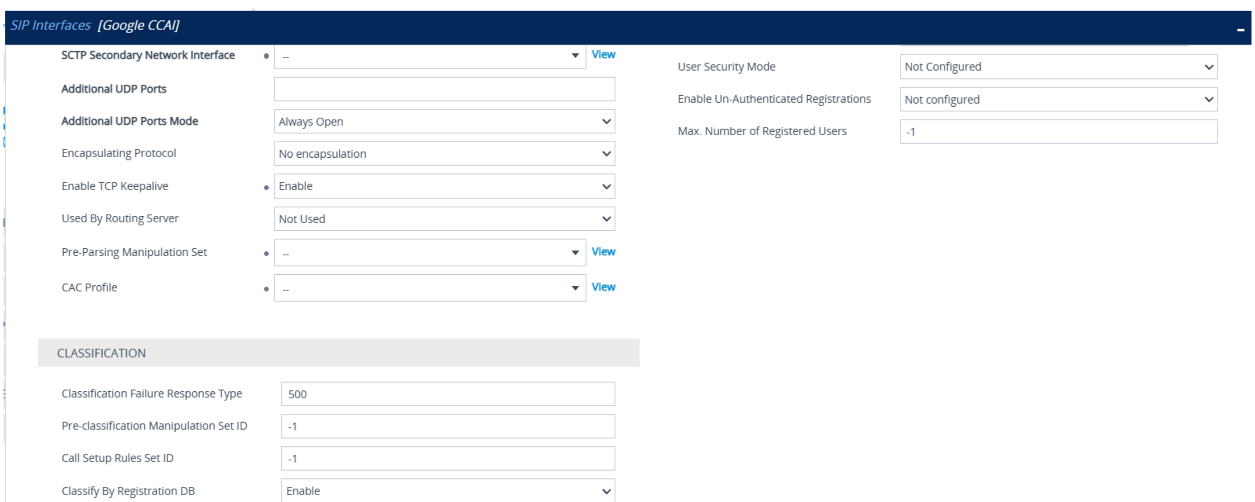


Figure 16: SIP Signaling Interfaces for Google CES (Cont.)

PSTN Gateway:

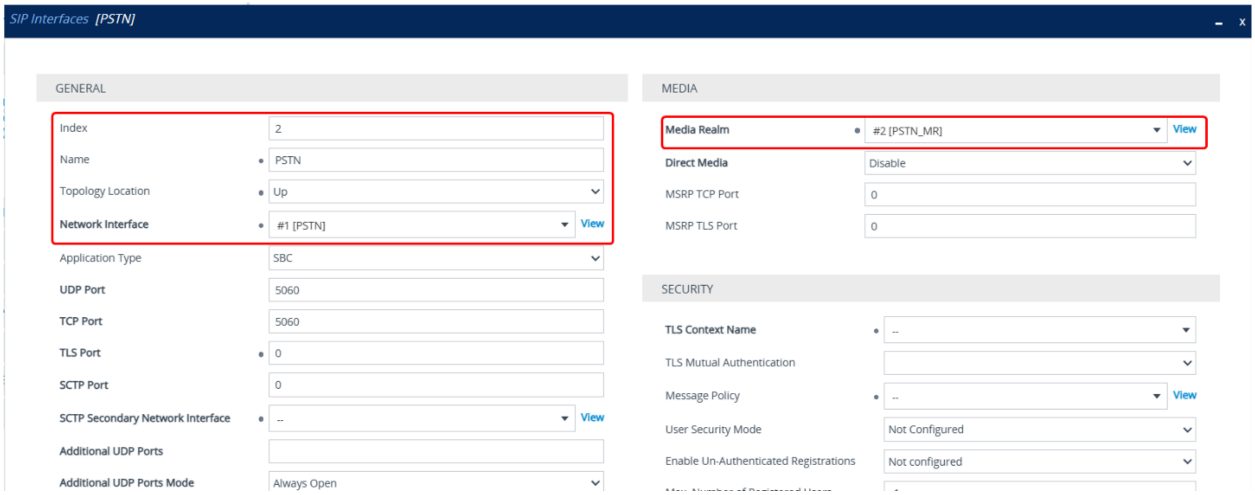


Figure 17: SIP Signaling Interfaces for PSTN Gateway

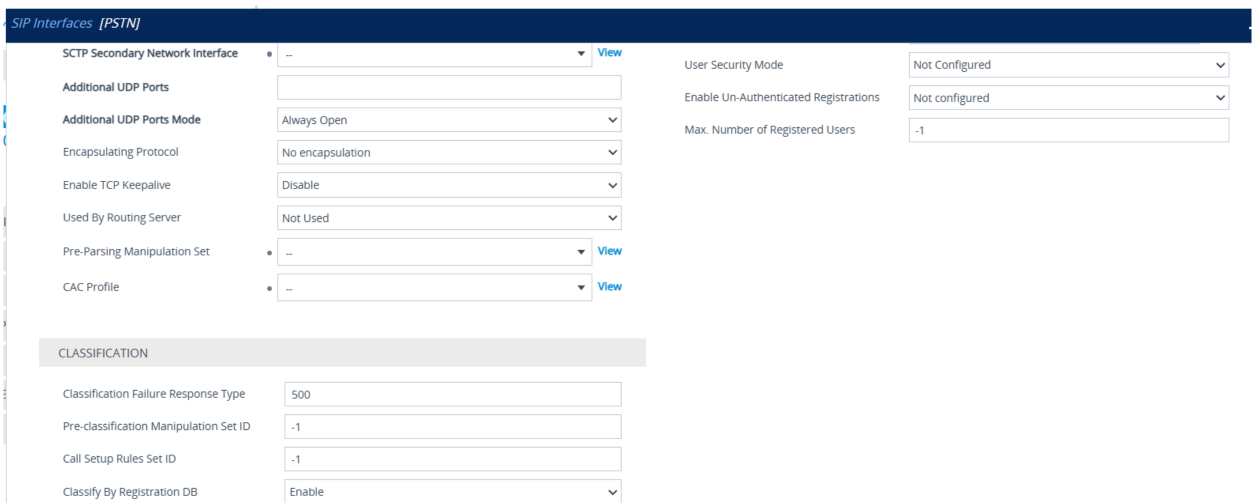


Figure 18: SIP Signaling Interfaces for PSTN Gateway (Cont.)

7.4.5 Configure Proxy Sets and Proxy Address

- Navigate to **SETUP** menu □ **SIGNALING & MEDIA** tab □ **CORE ENTITIES** folder □ **Proxy Sets.**
- Configure proxy sets for PSTN Gateway and Google CES as shown below.

Figure : Configurations of Proxy Sets

Google CES:

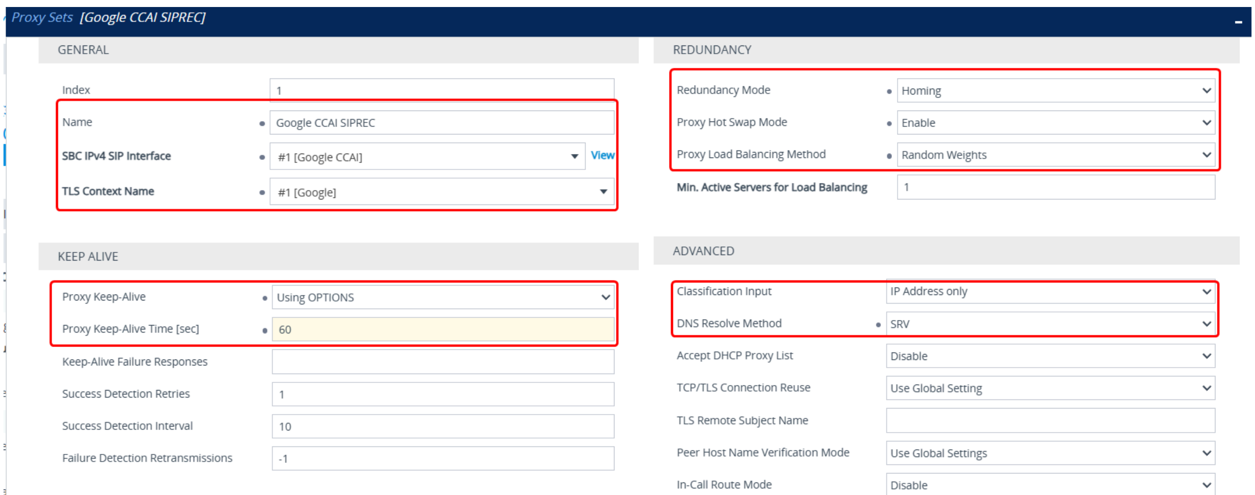


Figure 20: Proxy Set Configuration of Google CES

PSTN Gateway:

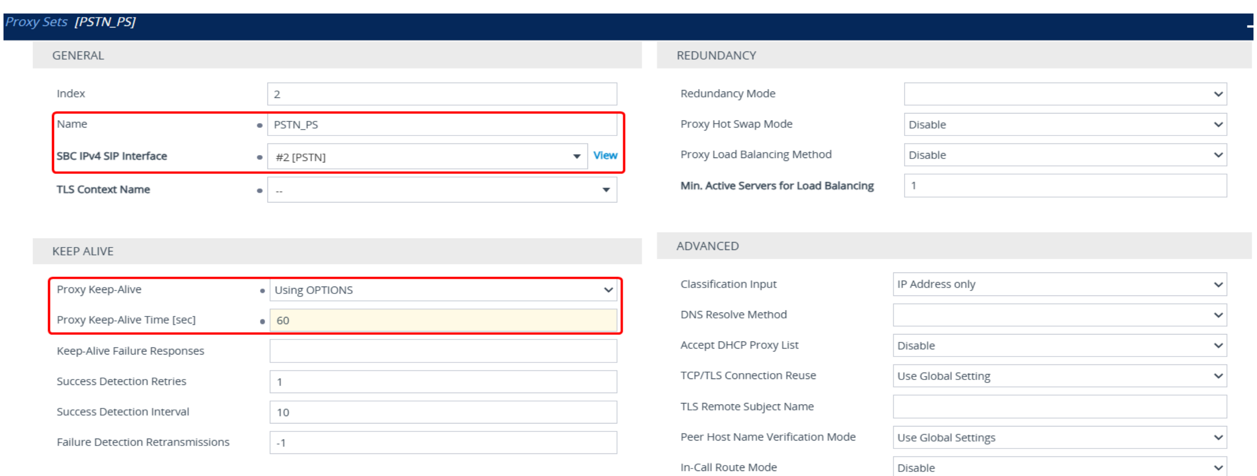


Figure 21: Proxy Set Configuration of PSTN Gateway

Proxy Address Configuration:

- Navigate into **SETUP** menu □ **SIGNALING & MEDIA** tab □ **CORE ENTITIES** folder □ **Proxy Sets.**
- Select the Google CES SIPREC Proxy Set and add the Proxy Address by clicking **Proxy Address items>>** and **+New.**

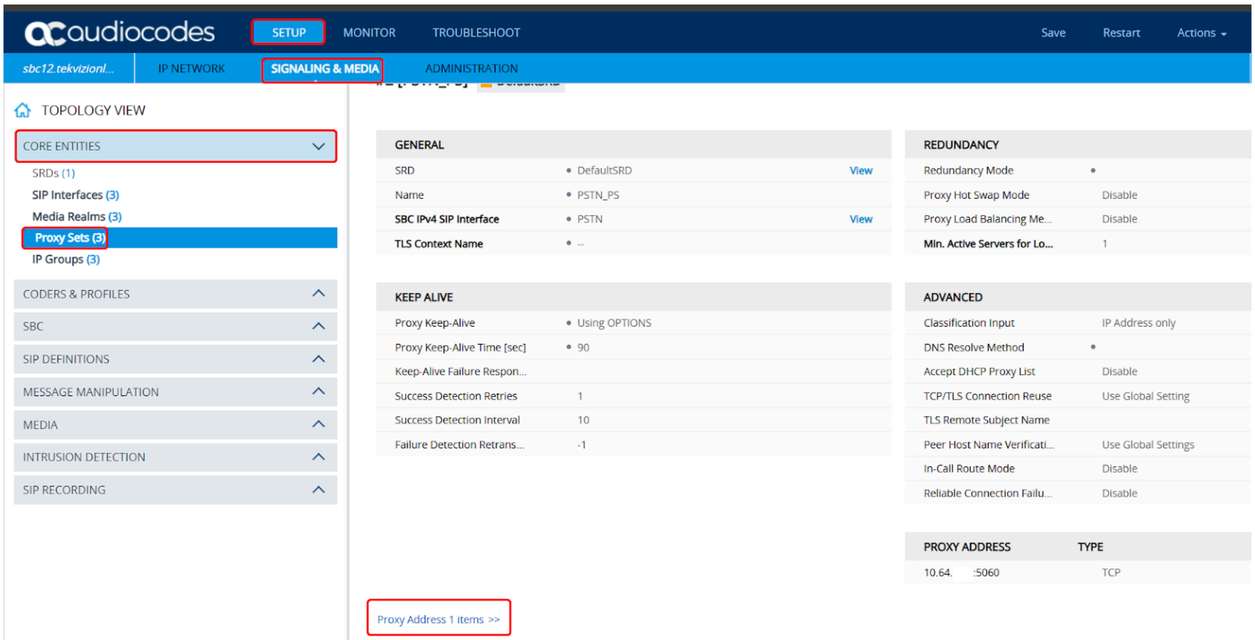


Figure 22: Proxy Address Configuration

- Enter the Google FQDN as proxy Address in the Google Proxy set and select transport type as TLS

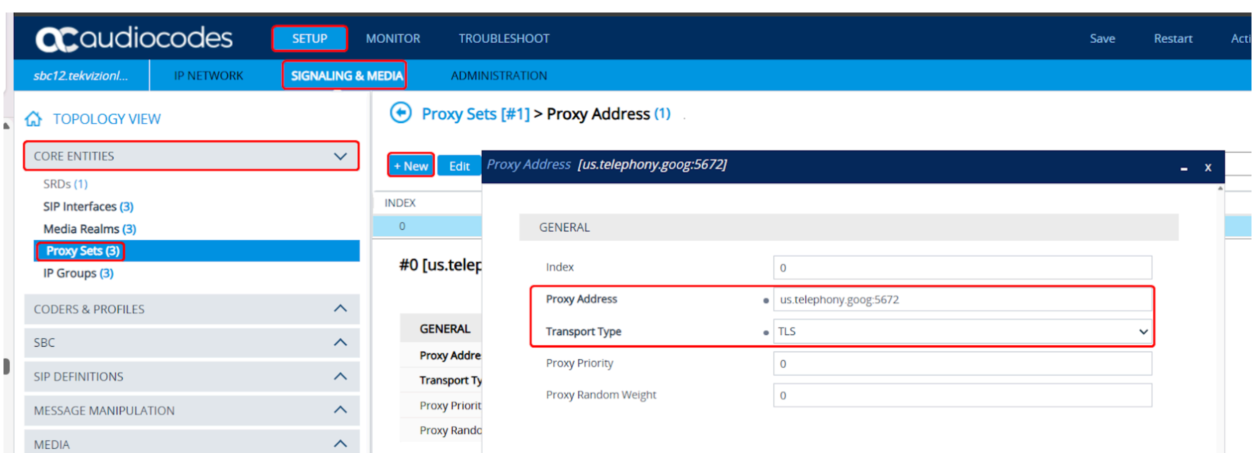


Figure 23: Proxy Address Configuration of Google CES

- Select the PSTN Proxy Set and add the Proxy Address by clicking **Proxy Address X items>>** and **+New.**
- Enter the PSTN gateway IP as Proxy Address in the PSTN proxy set and select transport as TCP.

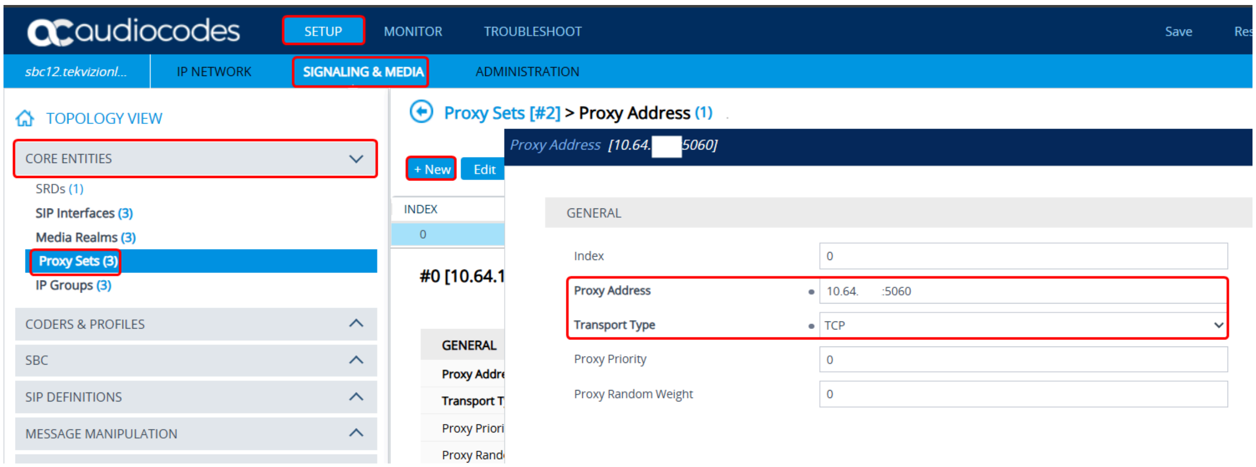


Figure 24: Proxy Address Configuration of PSTN Gateway

7.4.6 Configure Coders

- Navigate **to SETUP** menu □ **SIGNALING & MEDIA** tab □ **CODERS & PROFILES** folder □ **Coder Groups.**
- Configure the required Codecs as shown below.

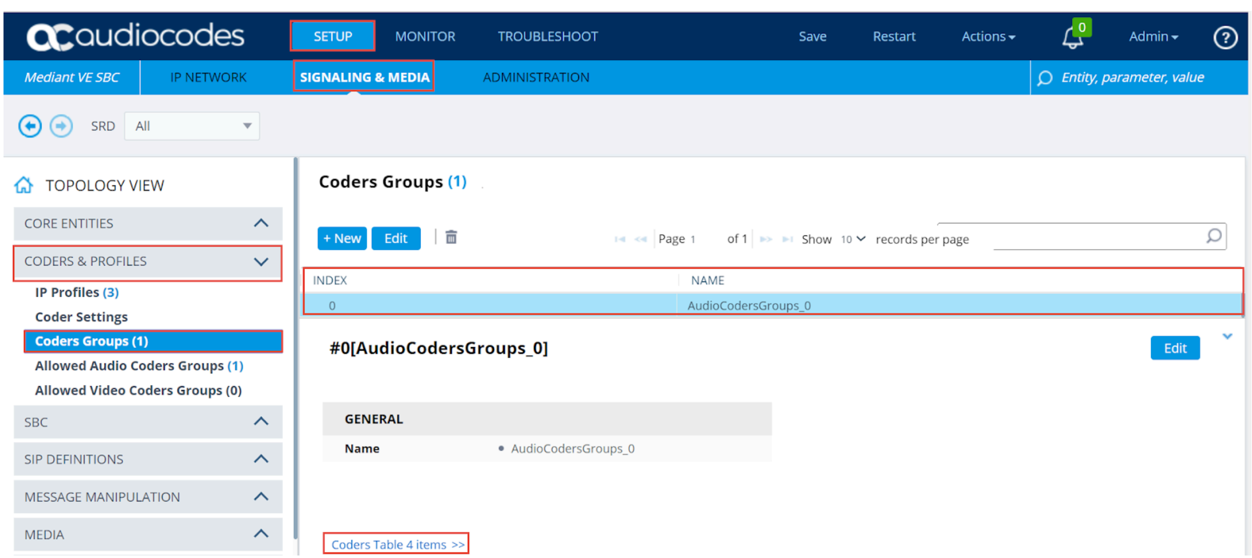


Figure 25: Coders Configurations

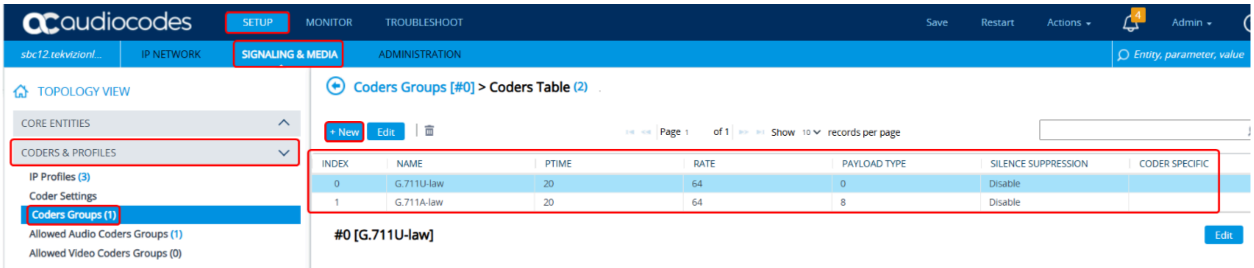


Figure 26: Coders Configurations (Cont.)

To Set a preferred coder for the Google CES:

- Navigate to the **SETUP** menu □ **SIGNALING & MEDIA** tab □ **CODERS & PROFILES** folder □ **Allowed Audio Coders Groups.**

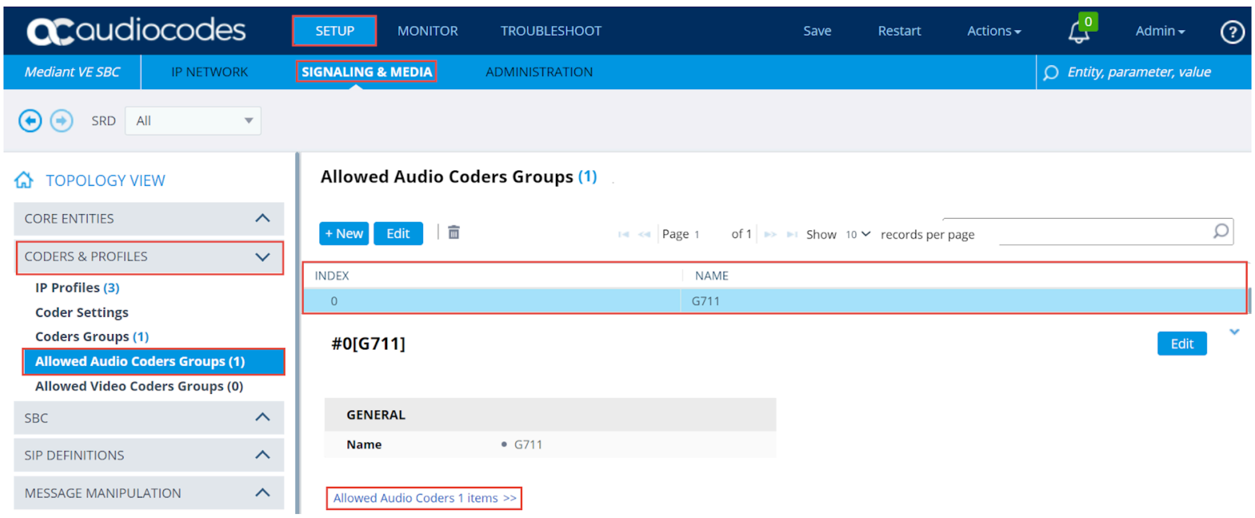


Figure 27: Coders Configurations (Cont.)

- Click **+New** and configure a new Allowed Audio Coders Group for Google CES with the preferred Codec list.
- Assign the configured Allowed Audio Coders Group to the respective Google CES IP Profile.

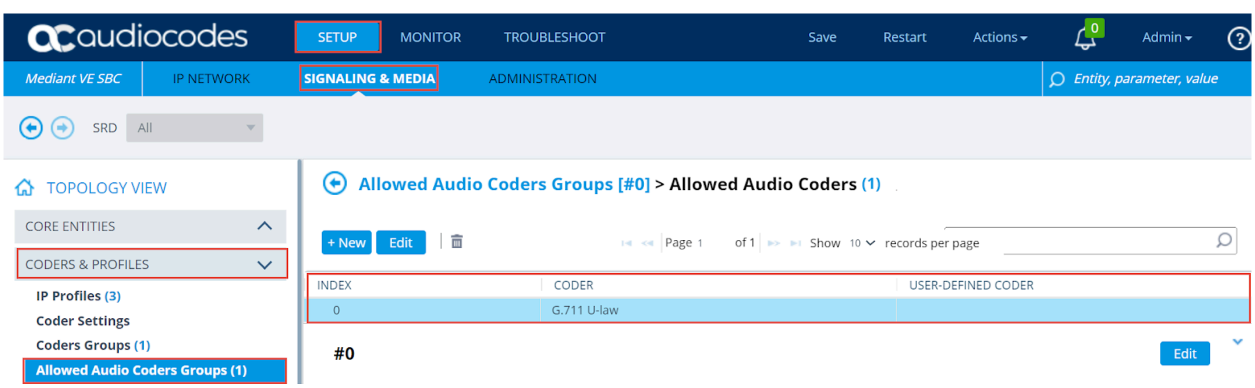


Figure : Coders Configurations (Cont.)

7.4.7 Configure IP Profiles

- Navigate to **SETUP** menu □ **SIGNALING & MEDIA** tab □ **CODERS & PROFILE** folder □ **IP Profiles.**
- IP Profile configuration for Google CES and PSTN Gateway are shown below.

Figure 29: IP Profile Configurations

PSTN Gateway IP:

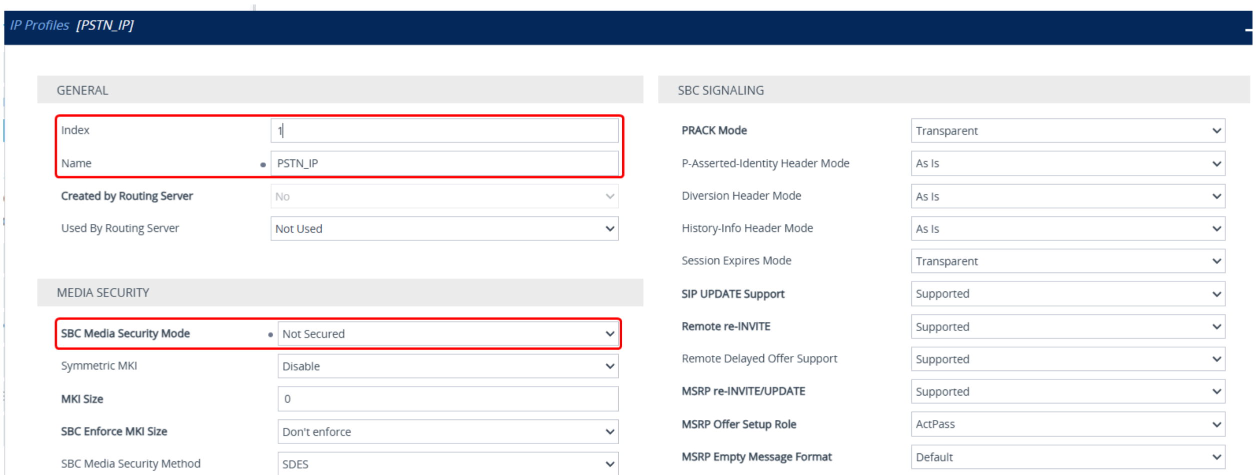


Figure 30: IP Profile Configurations of PSTN Gateway

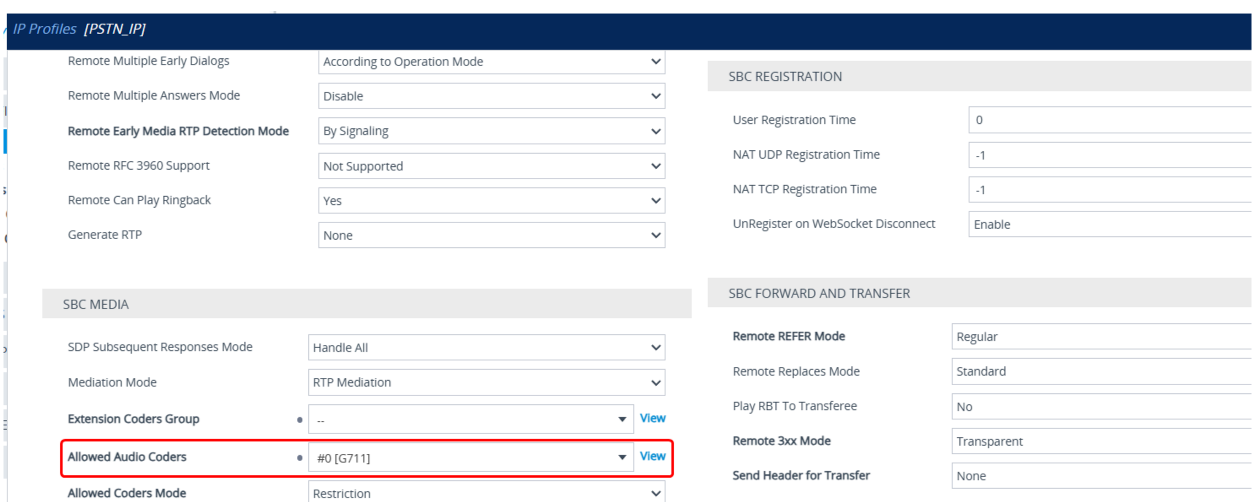


Figure 31: IP Profile Configurations of PSTN Gateway (Cont.)

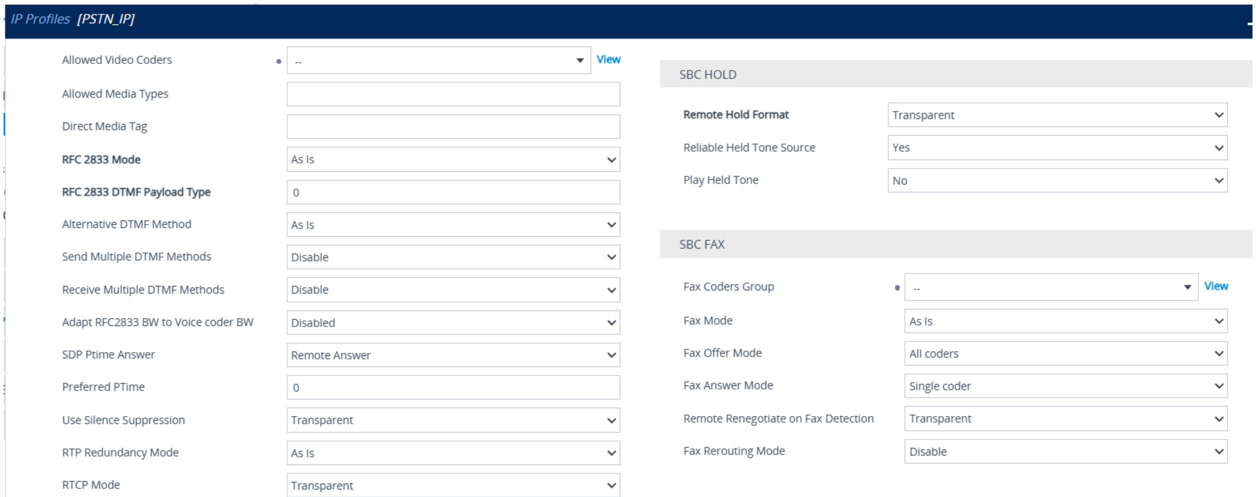


Figure 32: IP Profile Configurations of PSTN Gateway (Cont.)

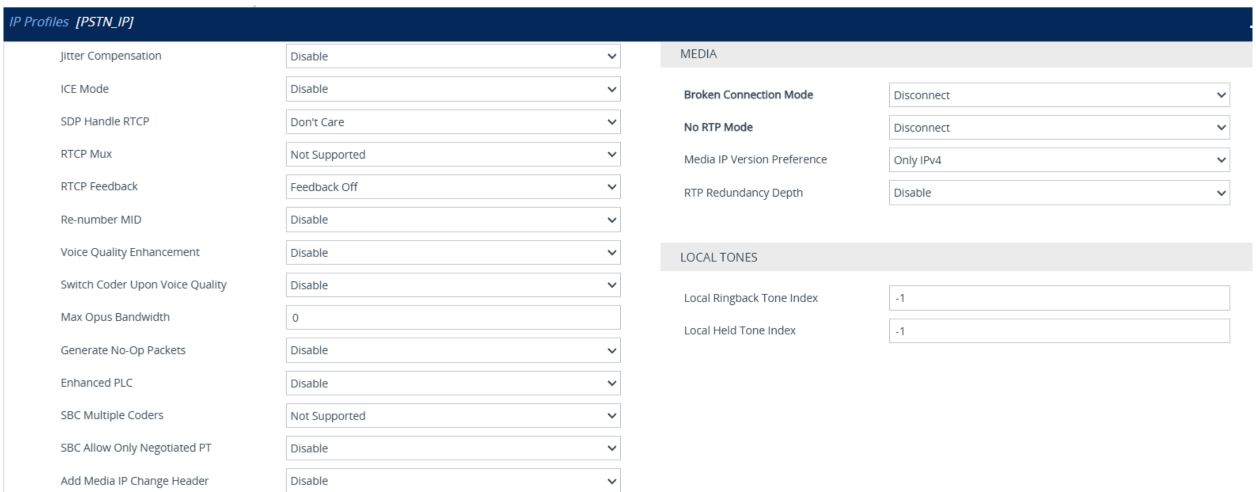


Figure 33: IP Profile Configurations of PSTN Gateway (Cont.)

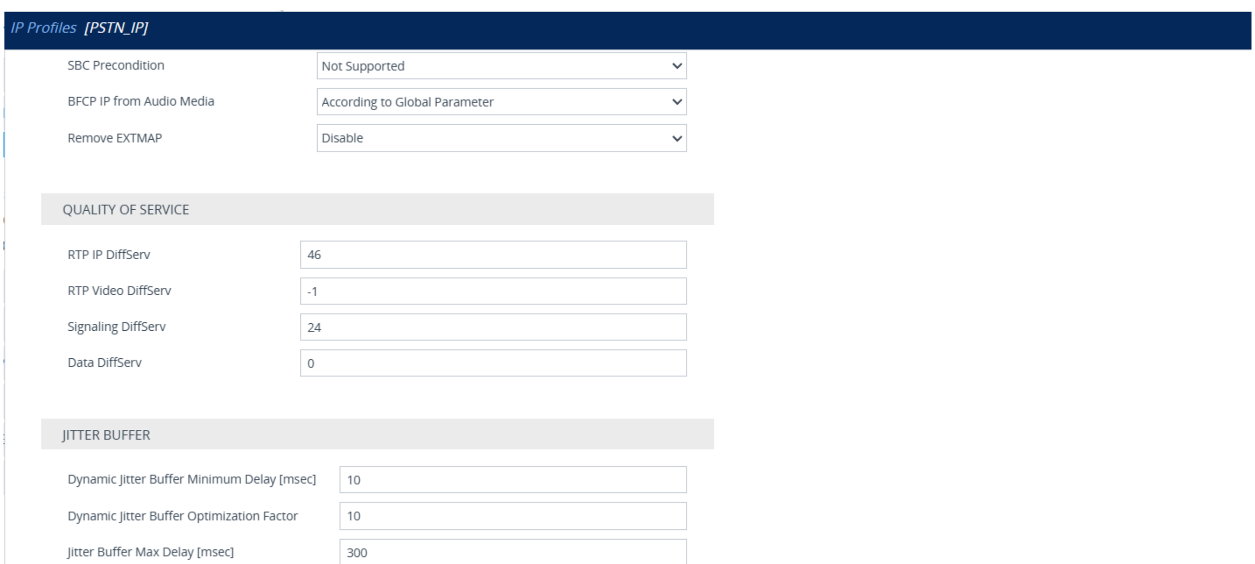


Figure 34: IP Profile Configurations of PSTN Gateway (Cont.)

Google CES:

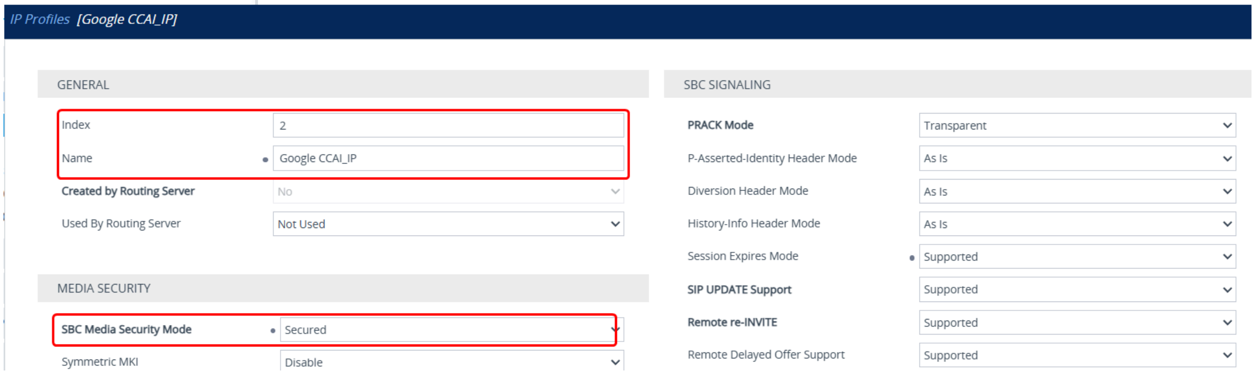


Figure 35: IP Profile configurations of Google CES

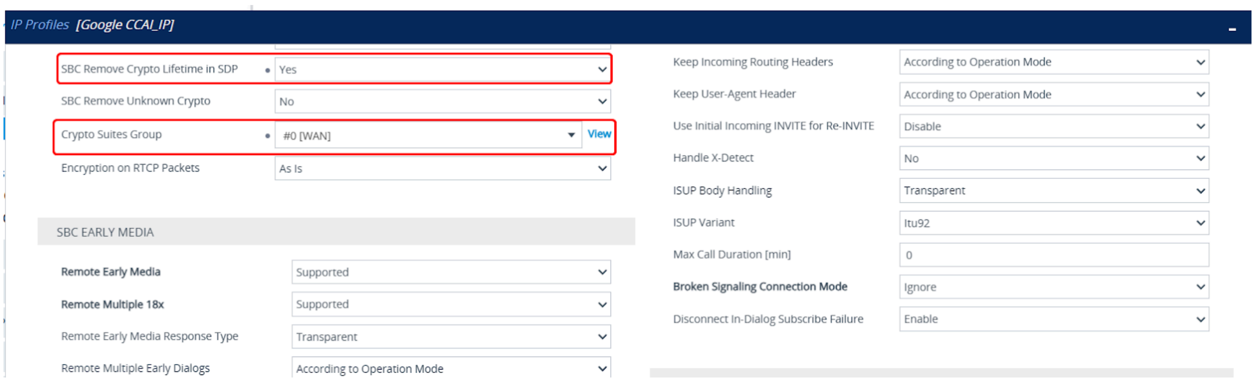


Figure 36: IP Profile Configurations of Google CES (Cont.)

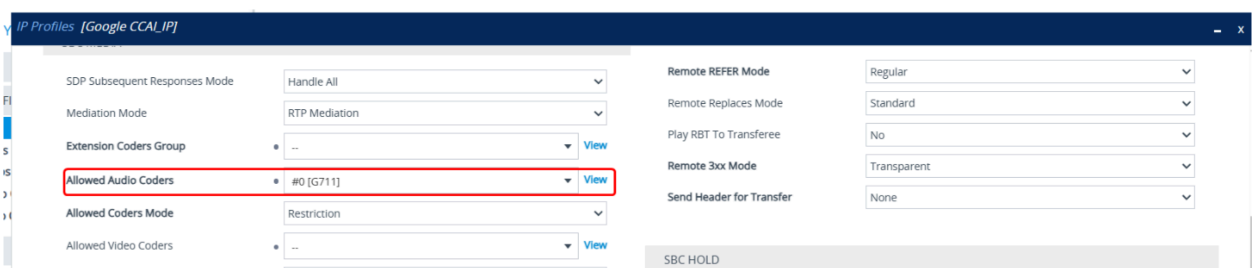


Figure 37: IP Profile Configurations of Google CES (Cont.)

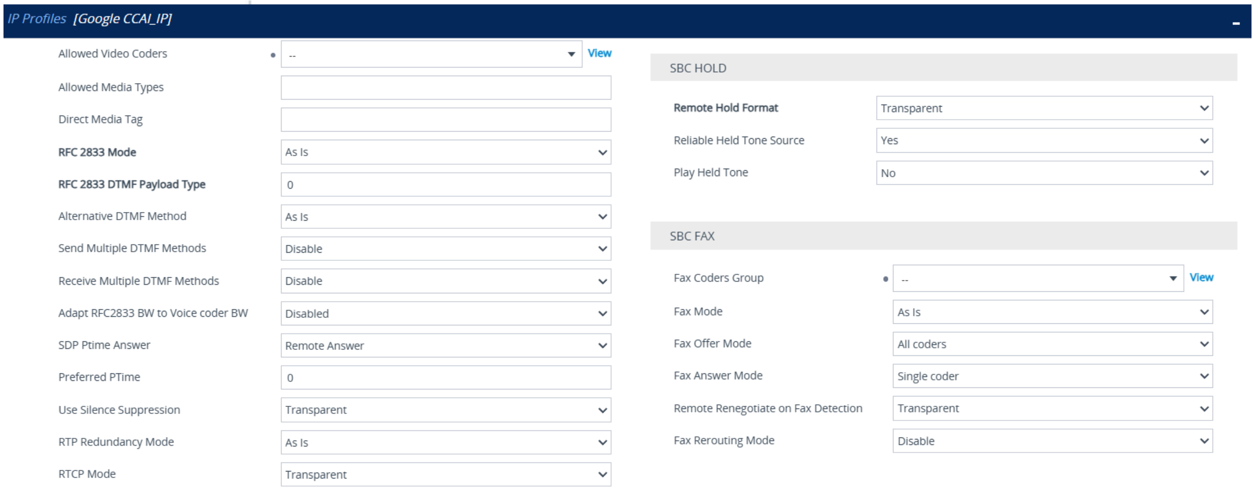


Figure 38: IP Profile Configurations of Google CES (Cont.)

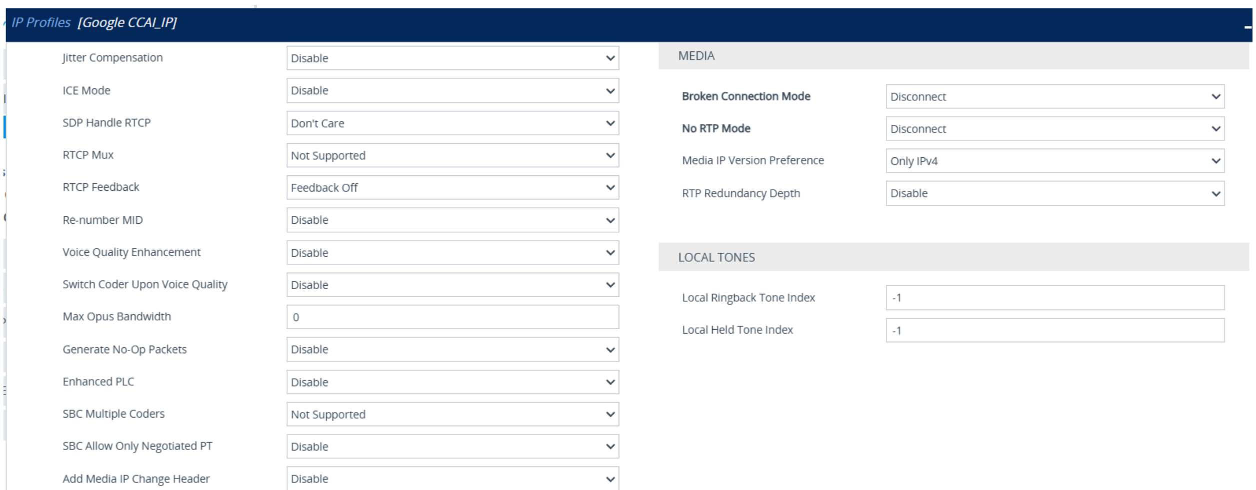


Figure 39: IP Profile Configurations of Google CES (Cont.)

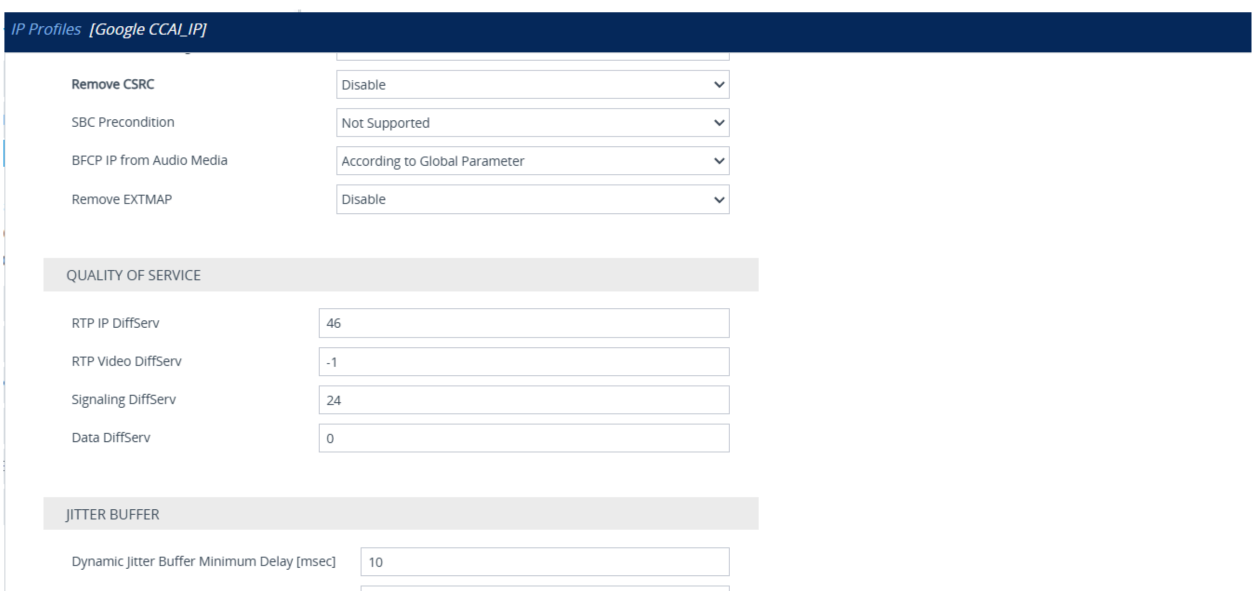


Figure 40: IP Profile Configurations of Google CES (Cont.)

7.4.8 Configure IP Groups

- Navigate to **SETUP** menu **SIGNALING & MEDIA** tab **CORE ENTITIES** folder **IP Groups**
- IP Groups Config towards Google CES and PSTN Gateway are shown below.

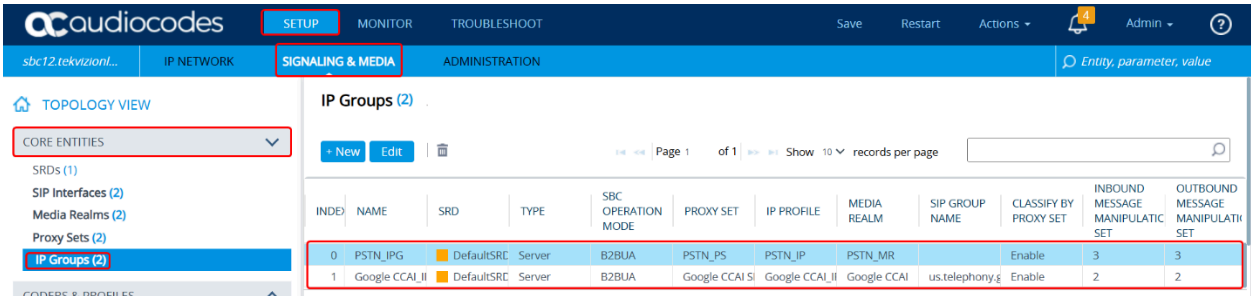


Figure 41: IP Group Configurations

- Select the respective Proxy Set, IP Profile, Media Realm and Media TLS Context for Google IP Group and enter Google FQDN as SIP Group Name.

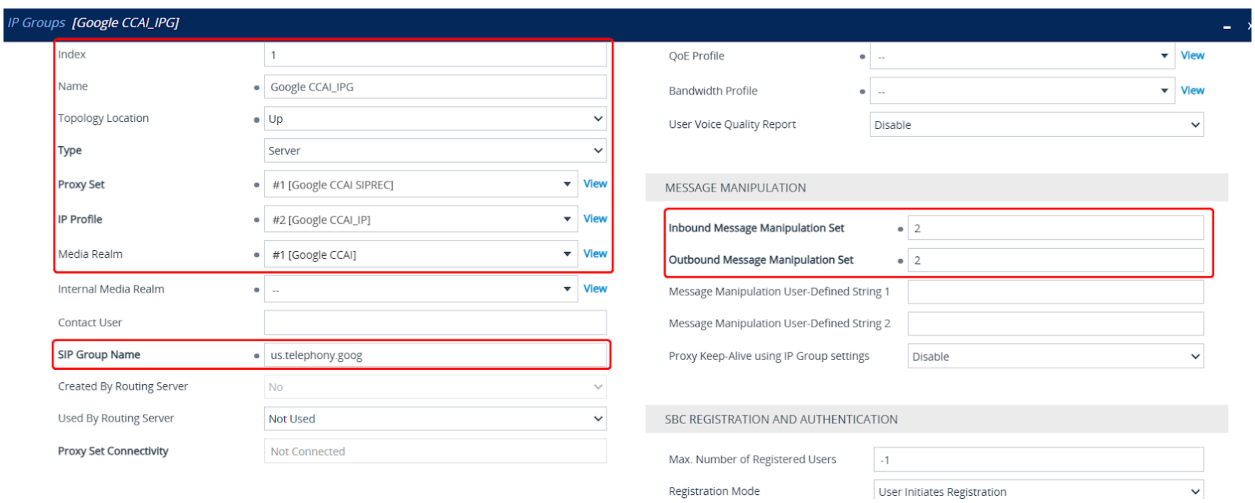


Figure 42:IP Group Configurations of Google CES

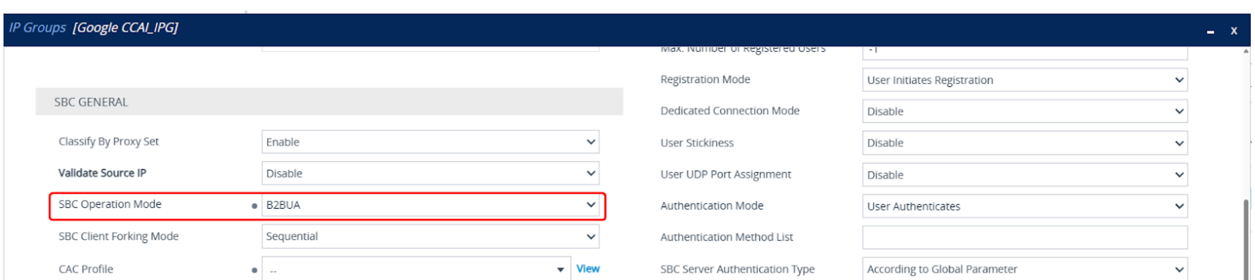


Figure 43: IP Group Configurations of Google CES (Cont.)

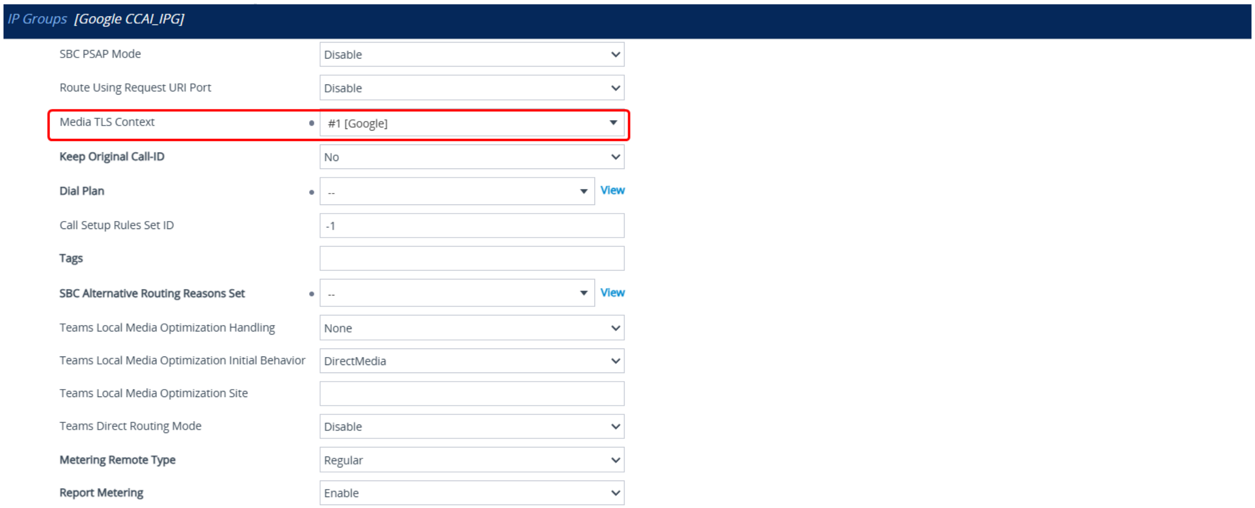


Figure 44: IP Group Configurations of Google CES (Cont.)

- Select the respective Proxy Set, IP Profile and Media Realm for PSTN Gateway IP Group and enter the PSTN Gateway IP as SIP Group name.

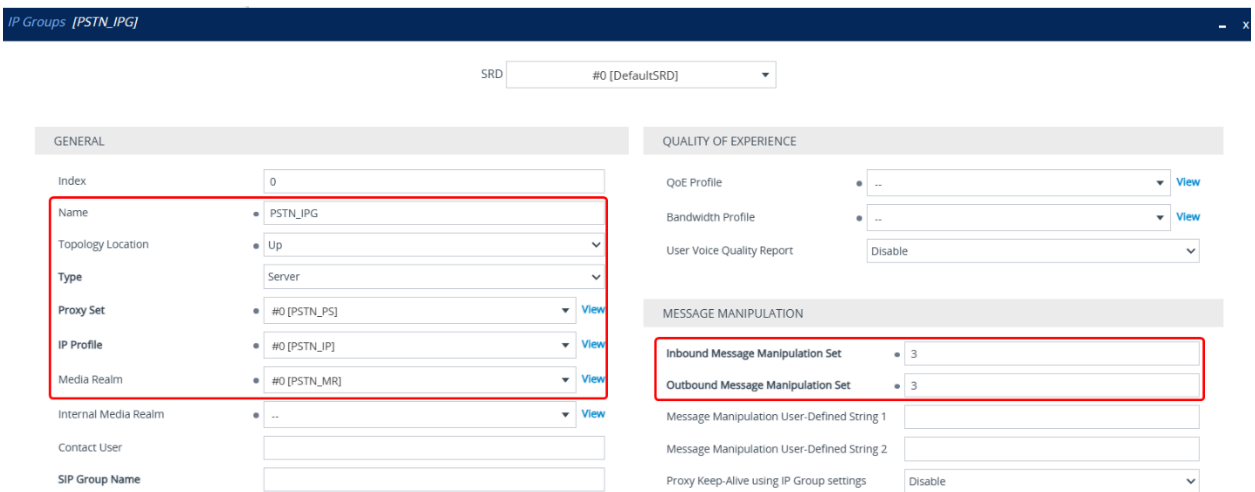


Figure 45: IP Group Configurations of PSTN Gateway

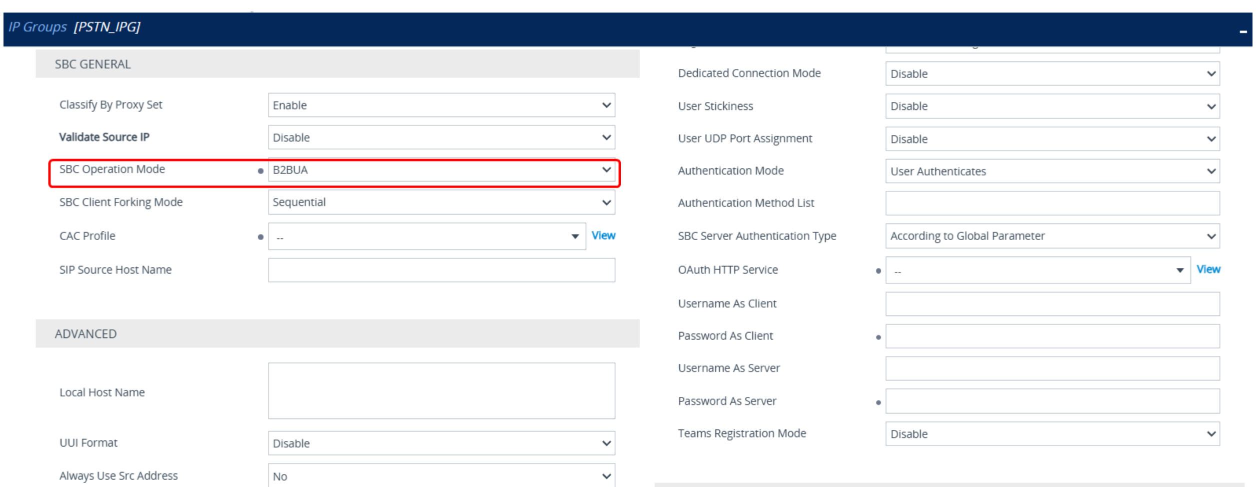


Figure 46: IP Group Configurations of PSTN Gateway (Cont.)

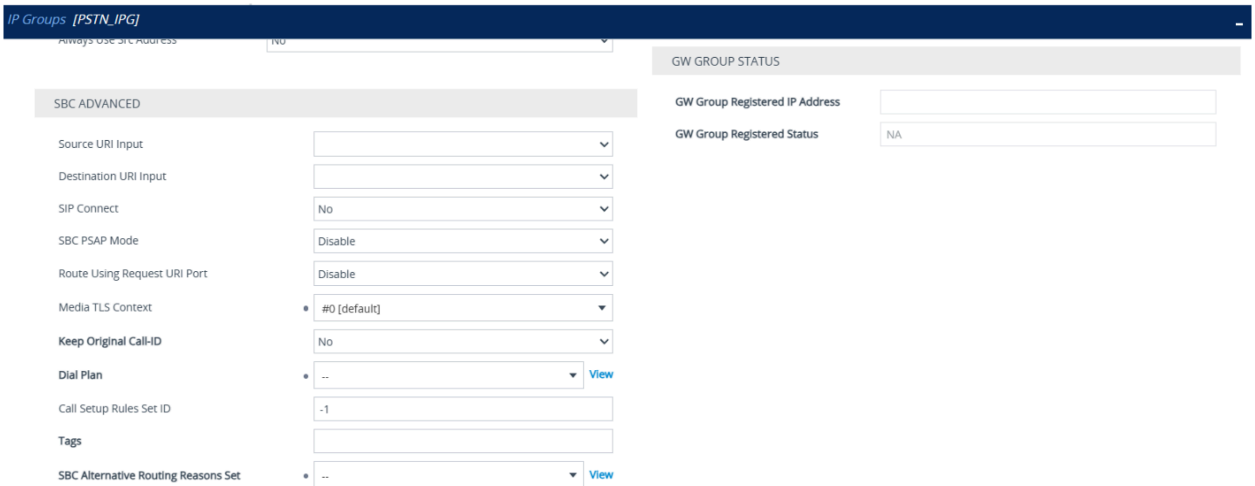


Figure 47: IP Group Configurations of PSTN Gateway (Cont.)

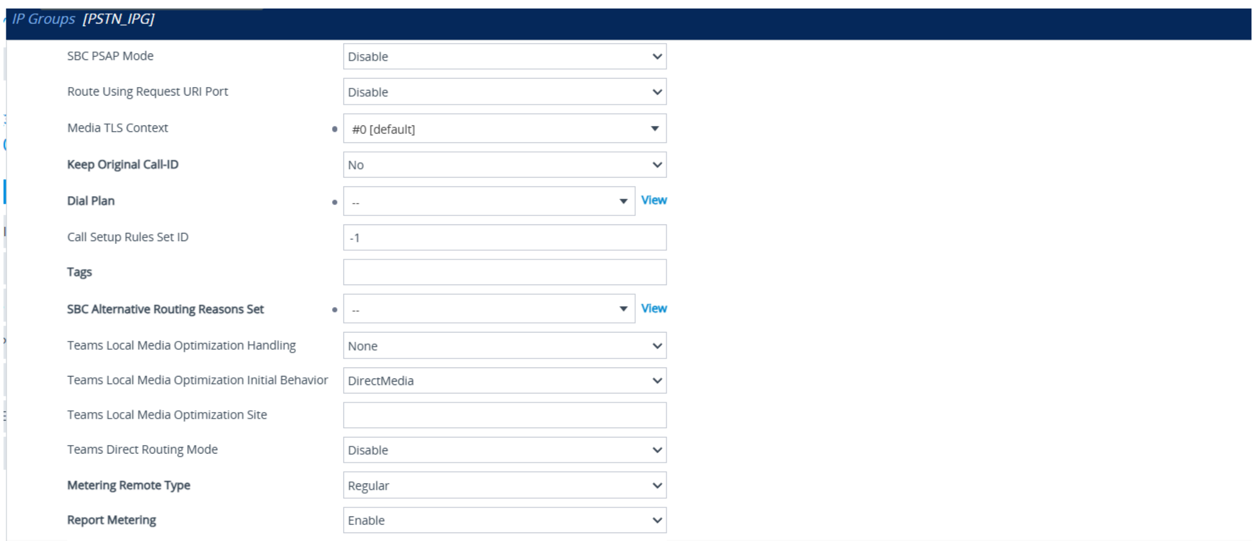


Figure 48: IP Group Configurations of PSTN Gateway (Cont.)

7.4.9 Configure Media Security

- Navigate to **SETUP** menu □ **SIGNALING & MEDIA** tab □ **MEDIA** folder □ **Media Security.**
- Enable Media Security as shown below.

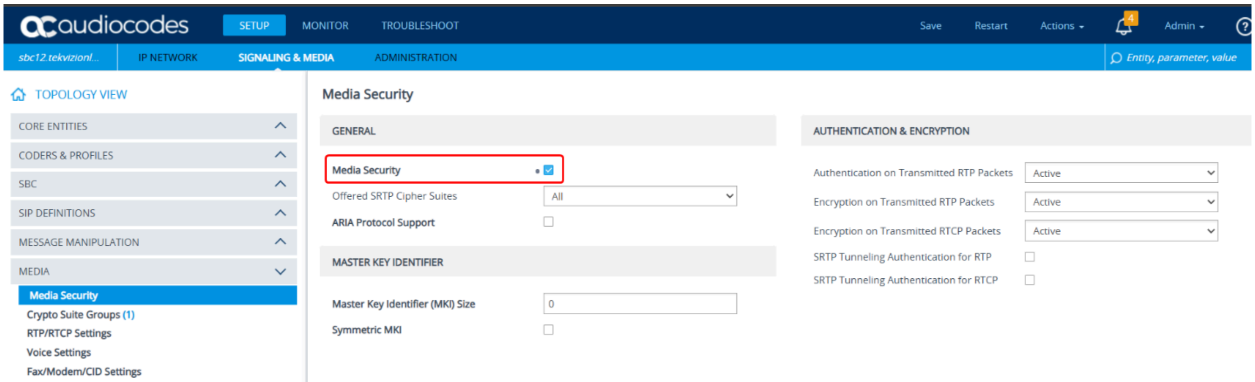


Figure 49: Media Security Configuration

7.4.10 Configure IP to IP Call Routing

- Navigate to **SETUP** menu □ **SIGNALING & MEDIA** tab □ **SBC** folder □ **Routing** □ **IP-to-IP Routing**
- Configure required routing rules as shown below.

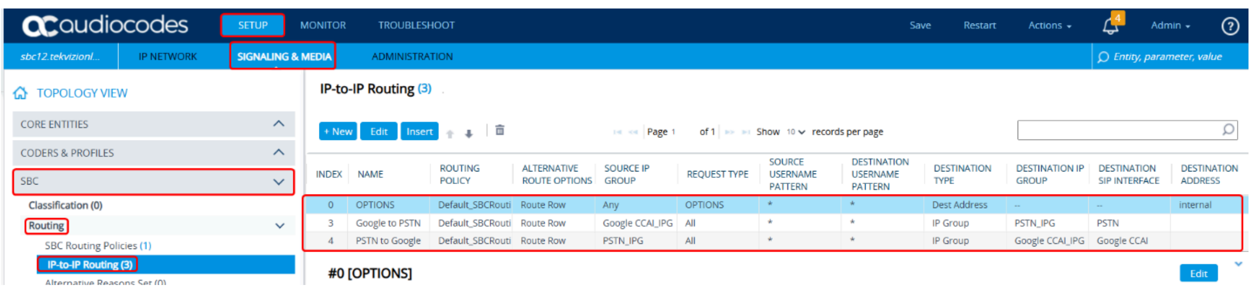


Figure 50: IP to IP Routing

7.4.11 Configure Message Manipulation Rules

- Navigate to **SETUP** menu □ **SIGNALING & MEDIA** tab □ **MESSAGE MANIPULATIONS** folder □ **Message Manipulations**
- Configure message manipulation towards Google CES as shown below.

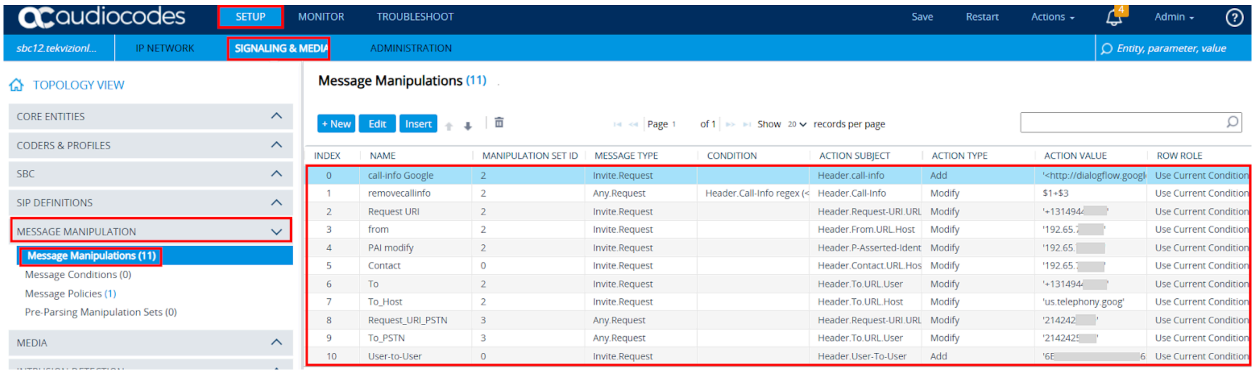


Figure 51:Message Manipulation for Google CES

- Below header rule is created to add Call-Info header towards Google CES with the Dialog Flow API request along with the Conversation ID.
- **Conversation on the Fly** is set to True in Goole CES using REST API. Conversation ID is randomly generated by AudioCodes SBC for each call.
- New Value is set to **<http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898XX/conversations/Sr_'+ Header.Call-ID.ID+'>;purpose=Goog-ContactCenter-Conversation.**

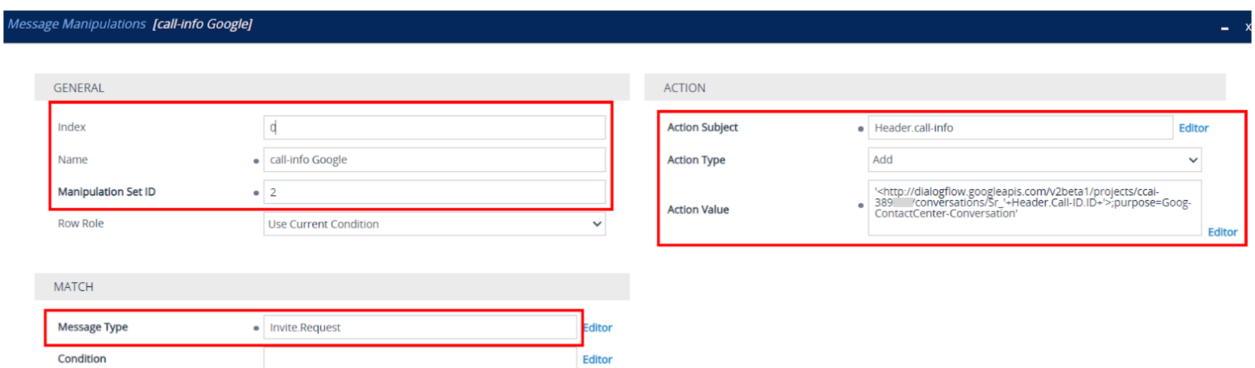


Figure 52: Message Manipulation: Call Info towards Google CES

- Below header rule is created to eliminate 192.65.X.X SBC WAN IP details from the call-Info header towards Google CES.

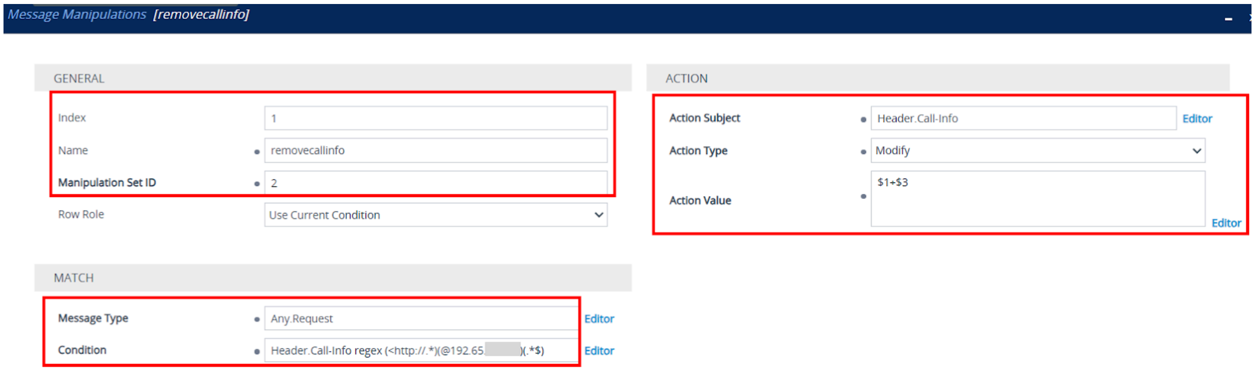
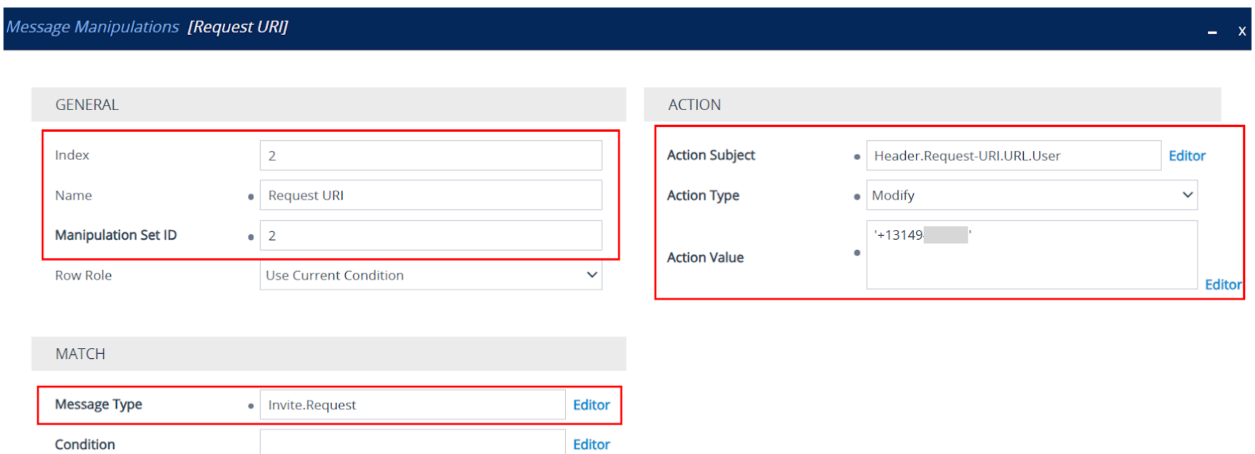


Figure 53: Message Manipulation: Call Info Modification towards Google CES

- Below header rule is created to change Request URI towards Google CES as '+1349XXXXXX'



- Below header rule is created to change from header host part towards Google CES as '192.65.X.X'

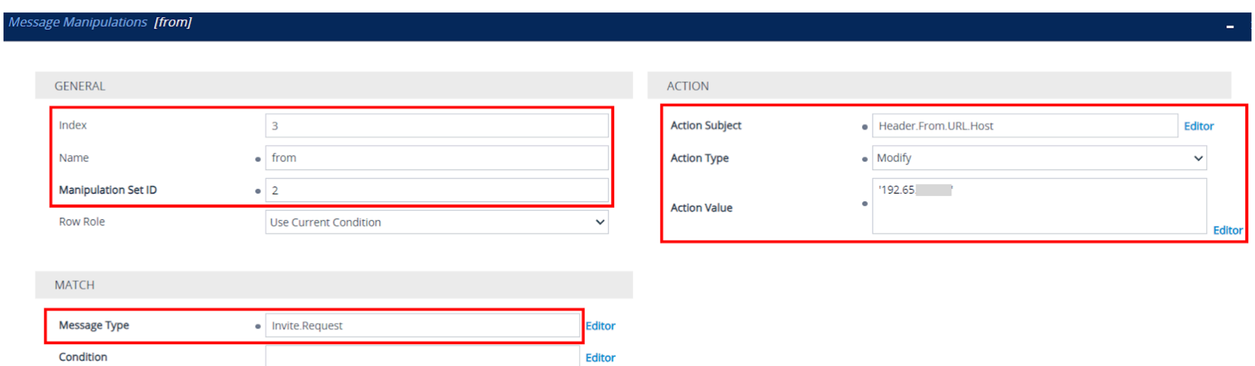


Figure 54: Message Manipulation: From Header host Part Modification towards Google CES

- Below header rule is created to change P-Asserted Identity host part towards Google CES as ‘192.65.X.X’

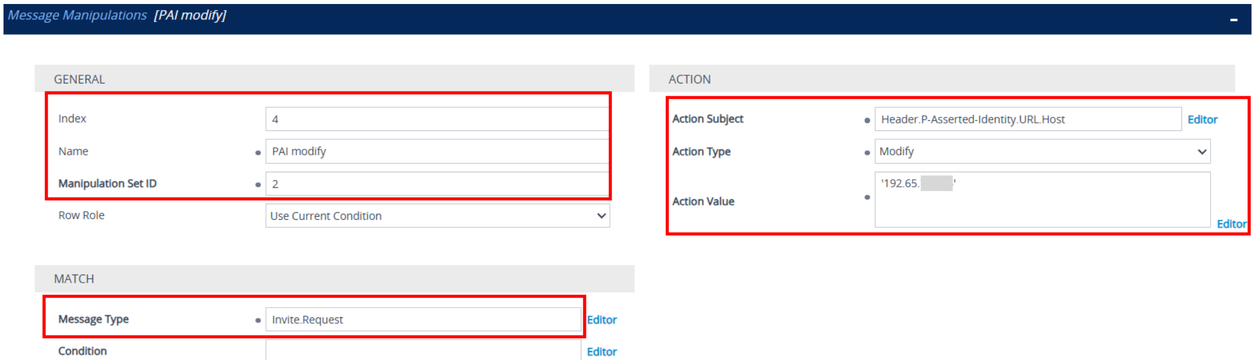


Figure 55: Message Manipulation: PAI Host Part Modification towards Google CES

- Below header rule is created to change To User part towards Google CES as ‘+1349XXXXXX

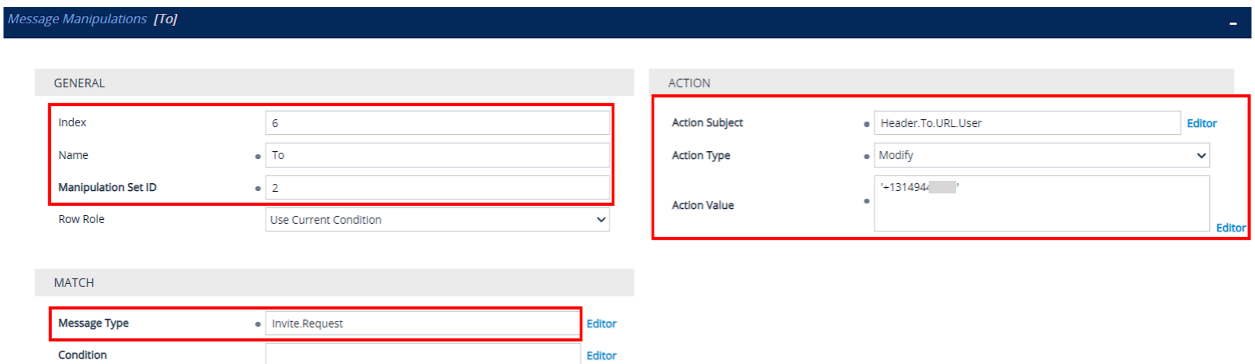


Figure 56: Message Manipulation: To User Part Modification towards Google CES

- Below header rule is created to change To host part towards Google CES as ‘us.telephony.goog’

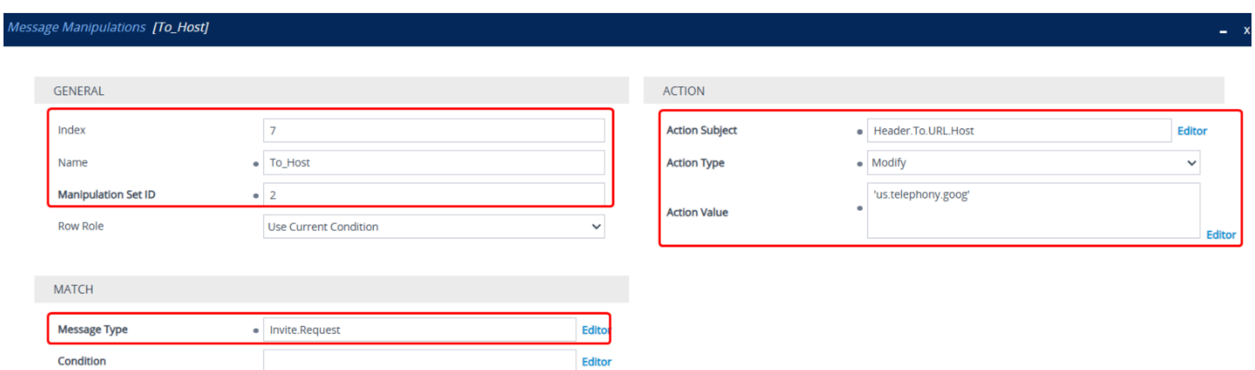


Figure 57: Message Manipulation: To Host Part Modification towards Google CES

Manipulation towards Google CES via UUI header:

- This SIP manipulation to insert a Hexa encoded value as a User-to-User header during Google CES Agent-Handoff.

  The URL : http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898XX/conversations/SR_QAZ
  Hexa Encoded : 6XXXXXXXXXXXXXXXXXXXXXXXXXXX5A

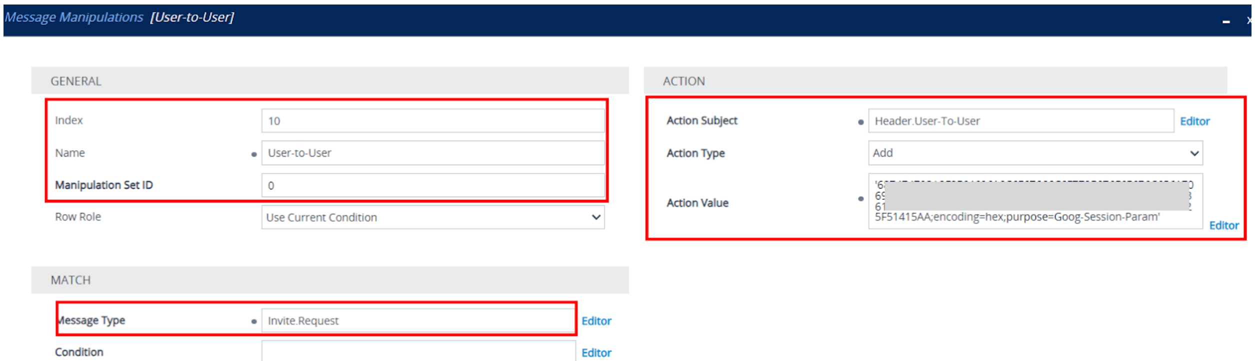


Figure 58: Message Manipulation: User-to-User header addition towards Google CES

Manipulation for Inbound messages from Google CES

- Below is the manipulation to modify Request-URI and To headers during call hand-off to an agent.

Request URI header:

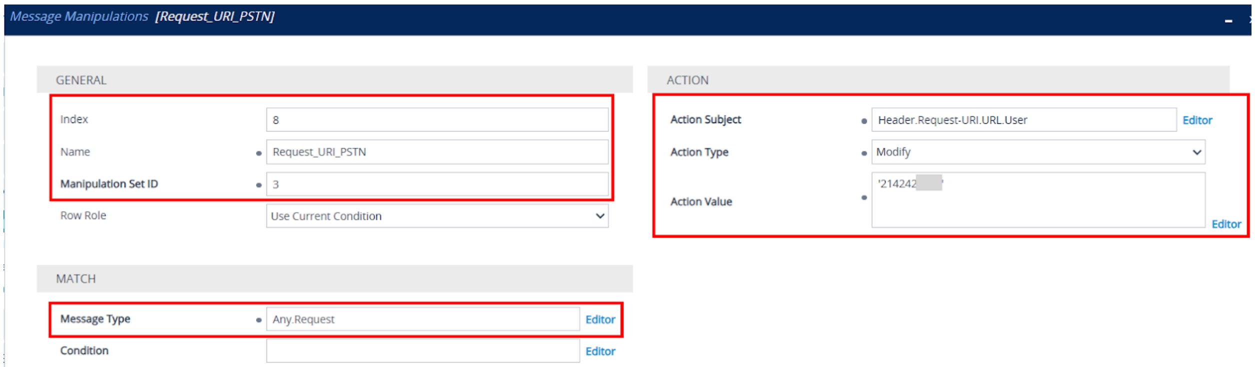


Figure 59: Message Manipulation: Request URI header towards PSTN

To Header:

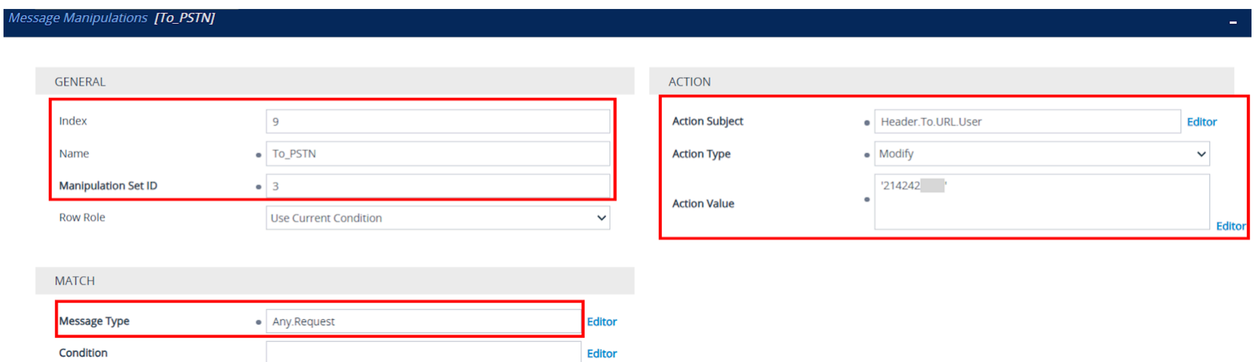


Figure 60: Message Manipulation: To header towards PSTN

8 SIP INVITE To GOOGLE CES

8.1 SIP INVITE for SIPREC call

```
INVITE sip:+13614@us.telephony.goog;user=phone SIP/2.0
Via: SIP/2.0/TLS 192.65.:5061;alias;branch=z9hG4bKac1515878973
Max-Forwards: 70
From: <sip:Audiocodes@192.65.;user=phone>;tag=1c1451797144
To: <sip:Audiocodes@us.telephony.goog;user=phone>
Call-ID: 14526490261920251384@192.65.
CSeq: 1 INVITE
Contact: <sip:192.65.:5061;transport=tls>;src
Supported: timer,replaces,resource-priority,sdp-anat
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INI
Require: siprec
Session-Expires: 1800;refresher=uas
Min-SE: 90
User-Agent: Mediant VE SBC/v.7.60A.100.022
Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/ccai-389/conversations/
Sr 145 84>;purpose=Goog-ContactCenter-Conversation
Content-Type: multipart/mixed;boundary=boundary_ac1c32
Content-Length: 2798
--boundary_ac1c32
Content-Type: application/sdp
v=0
o=AudiocodesGW 1438053757 169388785 IN IP4 192.65.
s=SBC-Call
c=IN IP4 192.65.
t=0 0
m=audio 7064 RTP/SAVP 0 101
c=IN IP4 192.65.
a=ptime:20
a=sendonly
a=label:1
a=fmtp:101 0-15,16
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:hH6VwqXGmE4z2J8Nnp6IXQefrfah4JxiX4ogHir0
a=crypto:2 AES_CM_128_HMAC_SHA1_32 inline:rtqOQzfvjF5y/Ajk0b0DySxQmA/Sh+6GTjko78ik
a=crypto:3 AES_256_CM_HMAC_SHA1_80 inline:WpppEpYoE4mUe7JfXKdDCzcpUMP1fGsXftyeZErq1q82u61/qRD+cdQ612fCwg==
a=crypto:4 AES_256_CM_HMAC_SHA1_32 inline:I1IjZjIMiUkl1RE00FQyHkGoHuC6THFMk+w9tif1xU+ZwPcx6KDUsQNMd87G0w==
m=audio 7068 RTP/SAVP 0 101
c=IN IP4 192.65.
a=ptime:20
a=sendonly
a=label:2
a=fmtp:101 0-15,16
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:7k4TFERScLdu+C6C9TdXTCHzN+0fYwuEqW+hcGOi
a=crypto:2 AES_CM_128_HMAC_SHA1_32 inline:3QQ96EGBNFmJiwXj6wXXx1fLfhUV9Yt0aSgUGz+D
a=crypto:3 AES_256_CM_HMAC_SHA1_80 inline:QbNDrW3B+ElV54j6+vqwfijFfwwOW7WC/kLkKjo7wdfeTieZm/nJvVDk5tijRA==
a=crypto:4 AES_256_CM_HMAC_SHA1_32 inline:0/kJ2CCnZhm6jwR6Db6HJQpkk3L0fVS8EieqO3RSitgeappqrnFimeZ/C1XzfA==
```

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 SIP GTP there will be a multiple media line with *a=sendrecv*
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 61: SIPREC Call

8.2 SIP INVITE for GTP call

```
INVITE sip:+13149@us.telephony.goog SIP/2.0
Via: SIP/2.0/TLS 192.65.:5061;alias;branch=z9hG4bKac210501
Max-Forwards: 69
From: "Pradeep Gopal" <sip:214550@192.65.>;tag=1c1363300463
To: <sip:+131494@us.telephony.goog>
Call-ID: 18511134392682025130320192.65.
CSeq: 1 INVITE
Contact: <sip:21455@192.65.:5061;transport=tls>
Supported: 100rel,timer,replaces,resource-priority,sdp-anat
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER,
Expires: 180
Session-Expires: 1800;refresher=uas
Min-SE: 1800
User-Agent: Mediant VE SBC/v.7.60A.100.022
P-Asserted-Identity: "Pradeep Gopal" <sip:21455@192.65.>
Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/ccai-389/conversations/
Sr_18 32>;purpose=Goog-ContactCenter-Conversation
Date: Tue, 26 2025 13:00:55 GMT
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 633
Cisco-Guid: 2295206282-2175996
Timestamp: 1756213255
v=0
o=CiscoSystemsSIP-GW-UserAgent 658476784 1219741378 IN IP4 192.65.
s=SIP Call
c=IN IP4 192.65.
t=0 0
m=audio 7192 RTP/SAVP 0 101
c=IN IP4 192.65.
a=fmtp:101 0-16
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:gV51G4J4rFnAKwmtb6U9Hd9OyEsbBVwNrIkQgBVQ
a=crypto:2 AES_CM_128_HMAC_SHA1_32 inline:3HvTldAZ/quJlW23Wchll4yL iVXym2tXurW
a=crypto:3 AES_256_CM_HMAC_SHA1_80 inline:7JjlVVOjEOONIo9gRZztFnaLGISUgSqFlafy6751dZYMyq+LGRB5BSyH0d8djw==
a=crypto:4 AES_256_CM_HMAC_SHA1_32 inline:UvPiO2AA0vl3J3RUyBEU8mx5/Hrojmrgs/T/hIEA0hJ3KvfvS1cFJ3aZkRQBYA==
```

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.

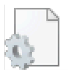
For SIPREC, there can be single media lines with *a=sendonly*, for SIP GTP there will be a multiple media line with *a=sendrecv*
Encrypted SRTP and the allocated port range should be used (Port: 16384-32767) else CES will not receive audio.

It must be a supported crypto suite by Google.

Figure 62:GTP Call

9 AudioCodes VE SBC Running configuration

Attached is the AudioCodes VE SBC Running configuration.



AudioCodes - Running config.ini

10 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 +13149XXXXXX, "speak		PASSED	Call was connected successfully with live agent and when performing conversation "Speak to an agent", a new INVITE was sent

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		to an agent", +1-972-852-XXXX should then ring and get connected as the agent			from Google to transfer to an agent and call gets connected successfully with both-way audio.
39	UUI headers	Call goes to virtual agent, say "end the call", validate that the SIP BYE has a UUI header Make calls to +13149XXXXXX, "end the call", check SIP BYE and ensure there is one or more (identify if there are 3 or 1) UUI headers with purpose Goog-Session-Param		PASSED	Calls get connected successfully to live agents and when performing conversation "End the call", the call gets disconnected normally with 3 UUI header.
40	SIP REFER	Call goes to virtual agent, say "send a sip refer", validate that a SIP REFER is received to 972-852-XXXX. Make calls to +1314XXXXXX "send a sip refer", SIP REFER should be received with refer to set to 972-852-XXXX		PASSED	REFER request & transfer is successful.