Configuration Guide for Google CES Call Recording Using AudioCodes VE SBC 7.60A.100.022

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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 **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 Call Recording with AudioCodes VE SBC 7.60A.100.022.

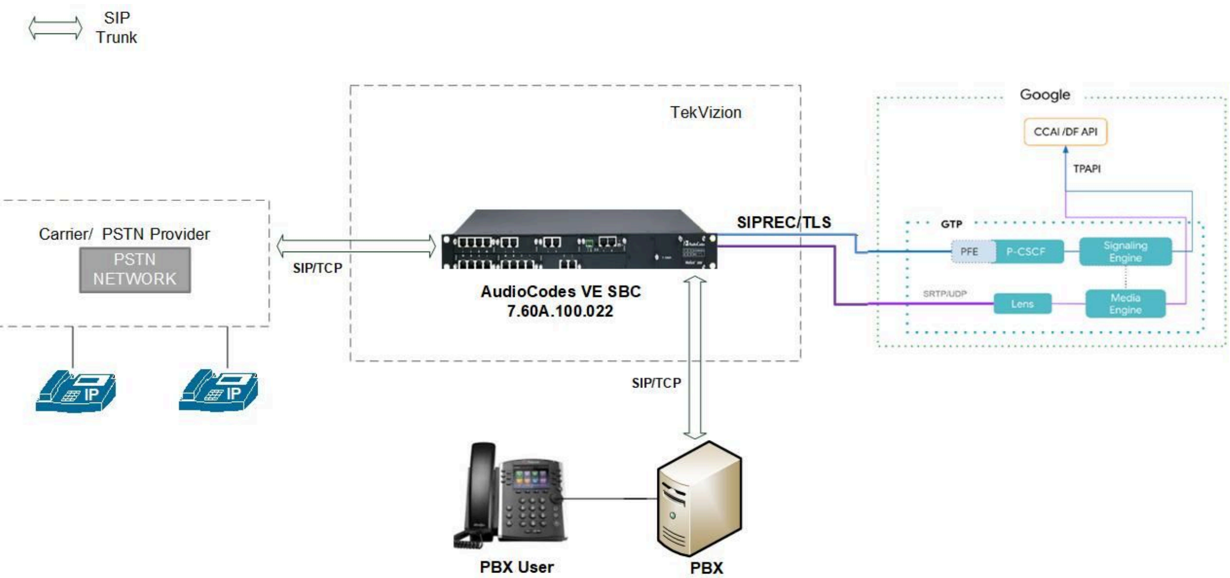


Figure 1: SIP Trunk Lab Reference Network

The lab network consists of the following components:

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

3 Hardware Components

- AudioCodes VE SBC running on ESXi host version 6.7.0

4 Software Requirements

- AudioCodes VE SBC Software 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
AudioCodes VE SBC	7.40A.500.786

6 Features

6.1 Features Tested for Google CES Call Recording

- Basic Inbound calls
- Call Hold and Resume
- Call Transfer (Blind and Consultative transfer)
- Conference

6.2 Features Not Tested for Google CES Call Recording

- None

6.3 Caveats and Limitations

DTLS	DTLS towards Google CES is not tested

6.4 Failed Test Case

- 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 SIP Recording	Section 7.4.11
Step 12	Configure Message Manipulation Rules	Section 7.4.12
Step 13	Configure Message Manipulation Rules (Participation label)	Section 7.4.13

7.2 IP Address Worksheet

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

Table 3 – IP Address Worksheet

Component	IP Address
Google CES	
Signaling	us.telephony.goog:5672
Media	74.125.X.X
OnPrem PBX	
LAN IP Address	10.80.X.X
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 Call recording.

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 Call Recording.

7.4.1 Network Interface IP

- Navigate to **SETUP** menu □ **IP NETWORK** tab □ **CORE ENTITIES** folder □ **IP Interfaces.**
- Configure IP Interfaces for OnPrem PBX, 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.

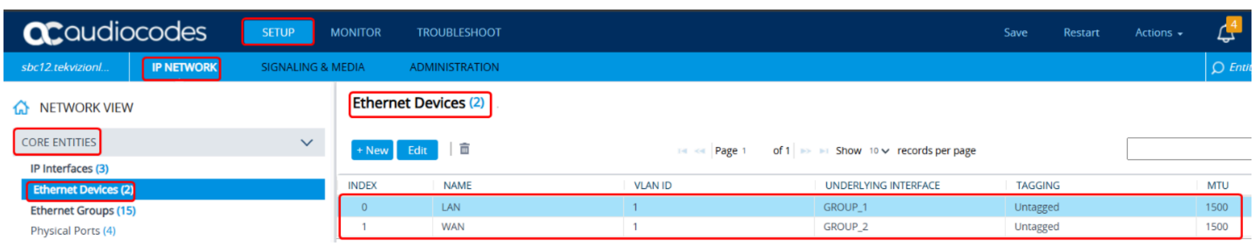


Figure 3: VLAN Configuration

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 OnPrem PBX, PSTN Gateway and Google CES as shown below.

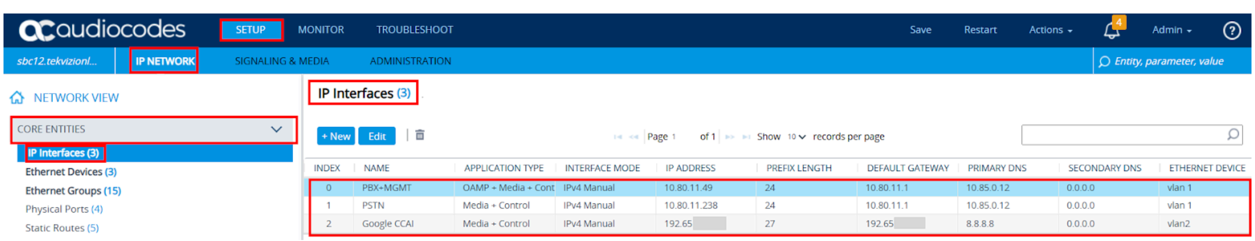


Figure 4: Network Interface Configuration

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.

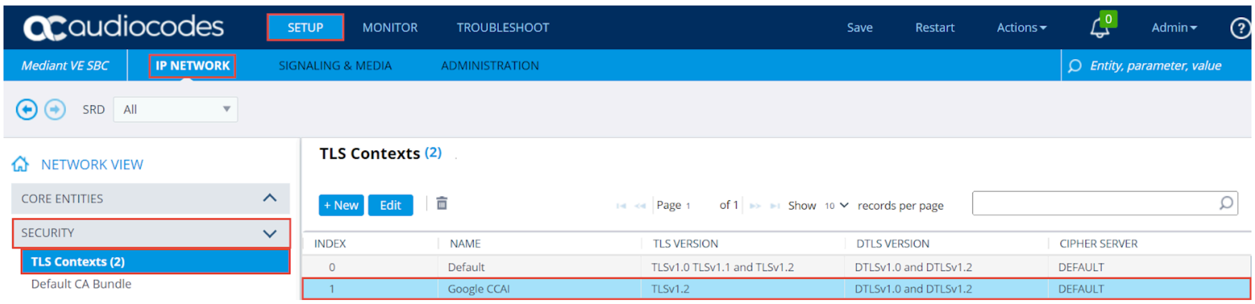


Figure 5: TLS Context for Google CES

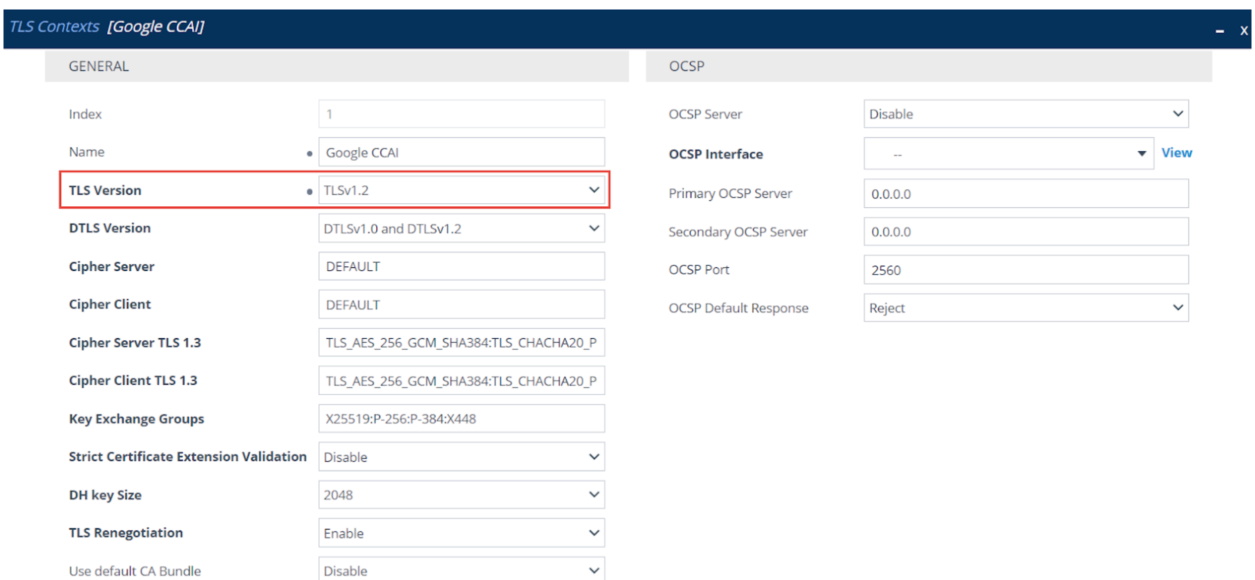


Figure 6: TLS Context for Google CES (Cont.)

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.

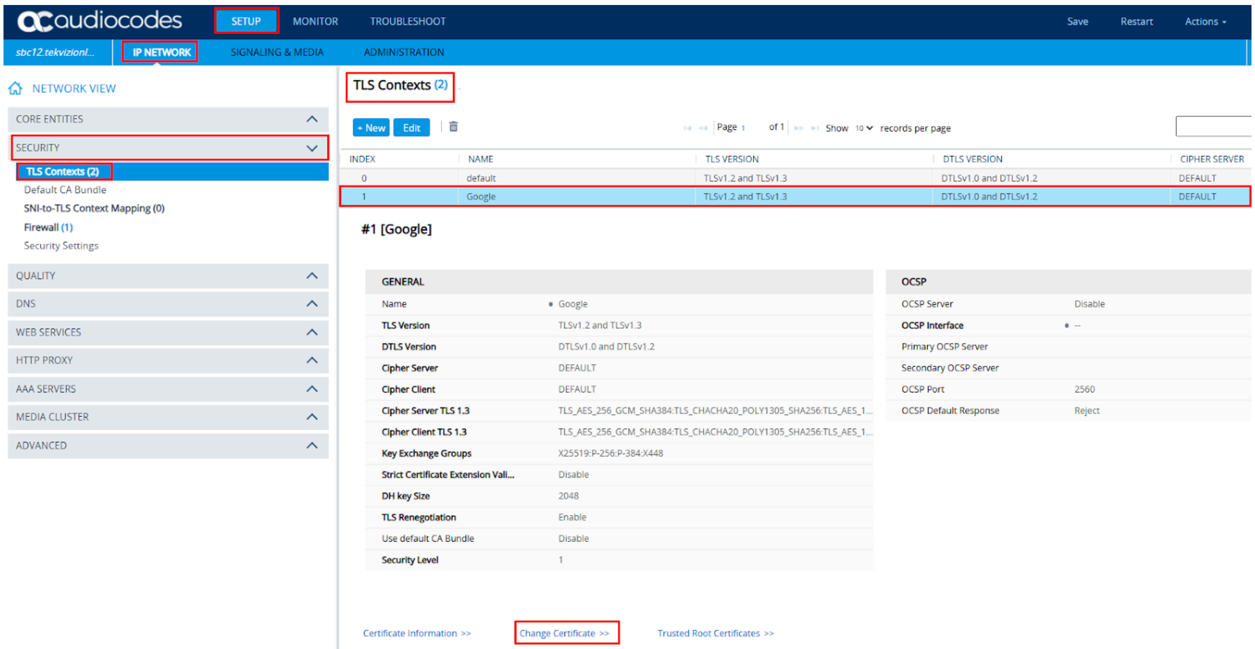


Figure 7: Change Certificate for CSR Generation

- 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.

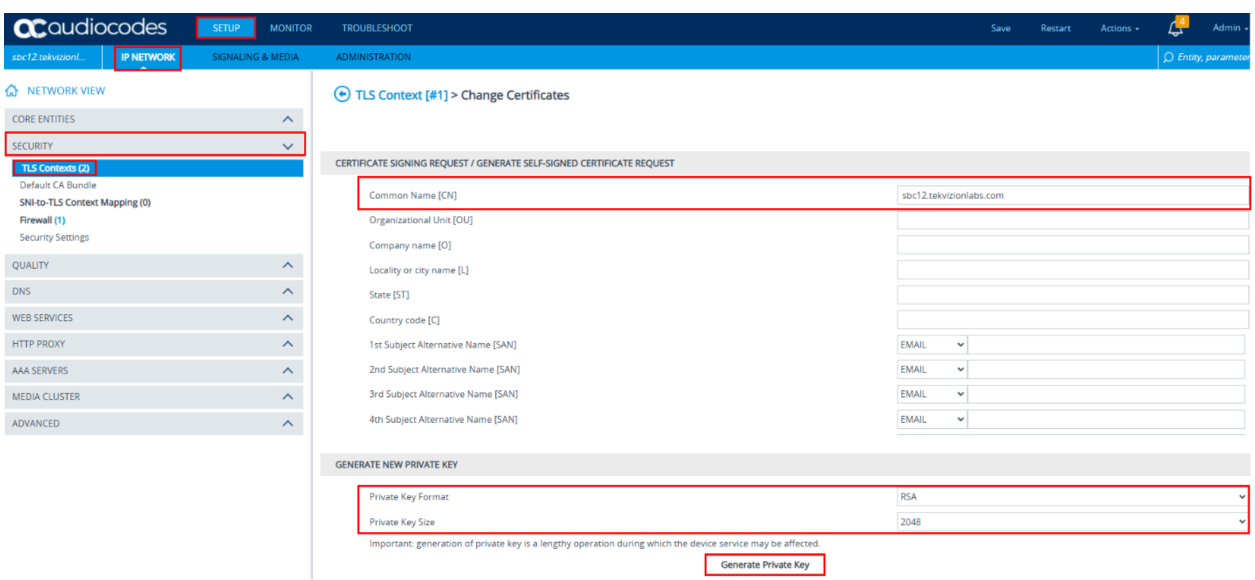


Figure 8: CSR Generation for Google CES TLS Context

- 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

Upload AudioCodes VE SBC server 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.

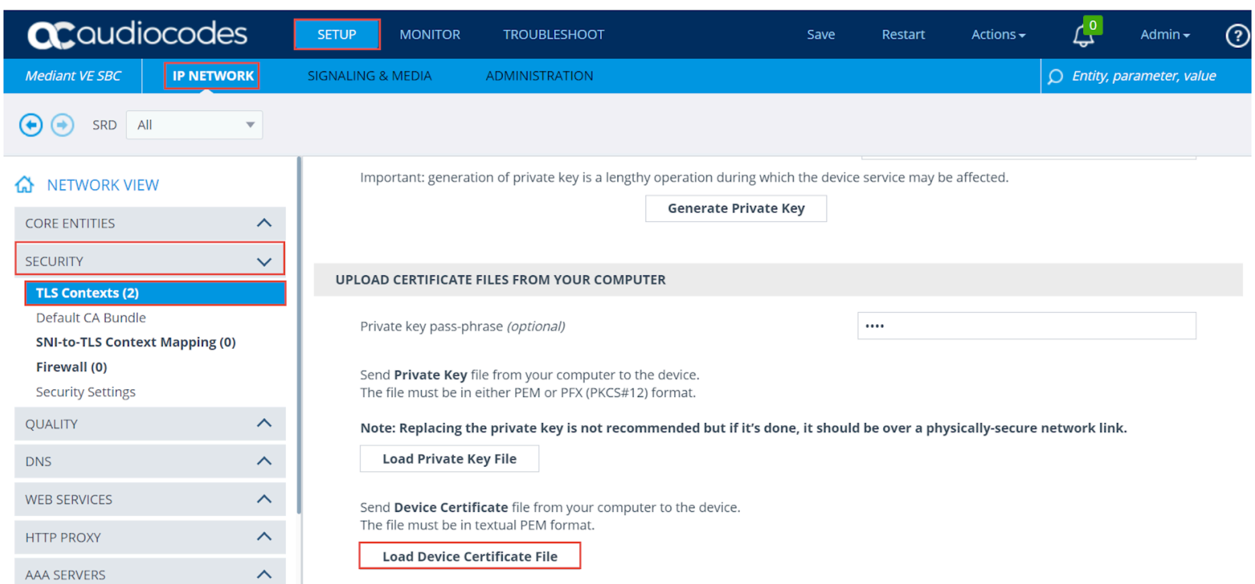


Figure 10: SBC Certificate Upload

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.

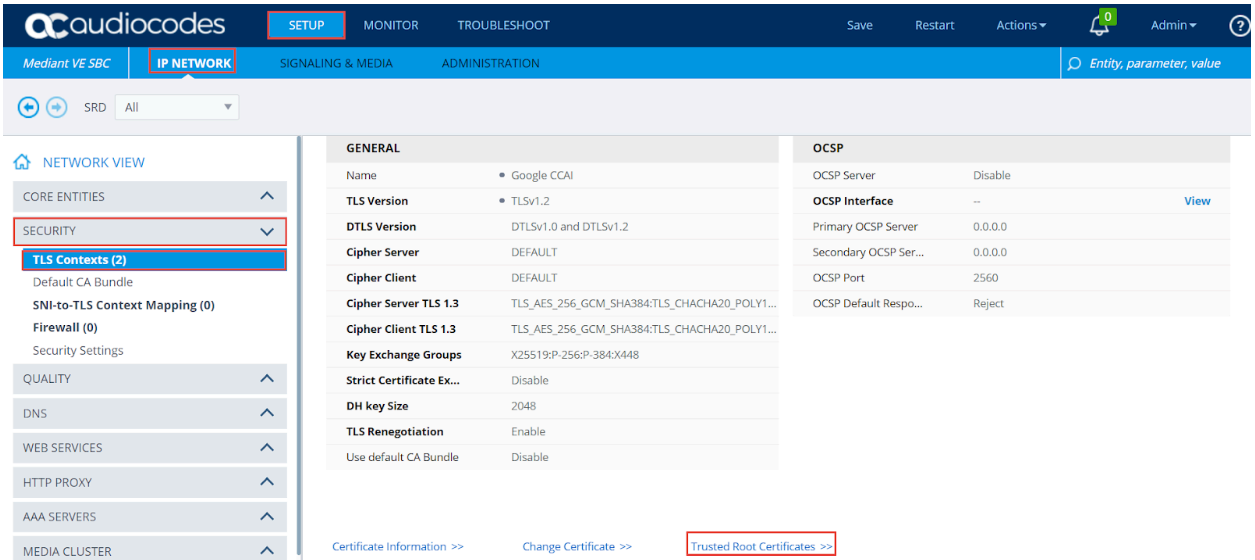


Figure 11: Trusted Root Certificates

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

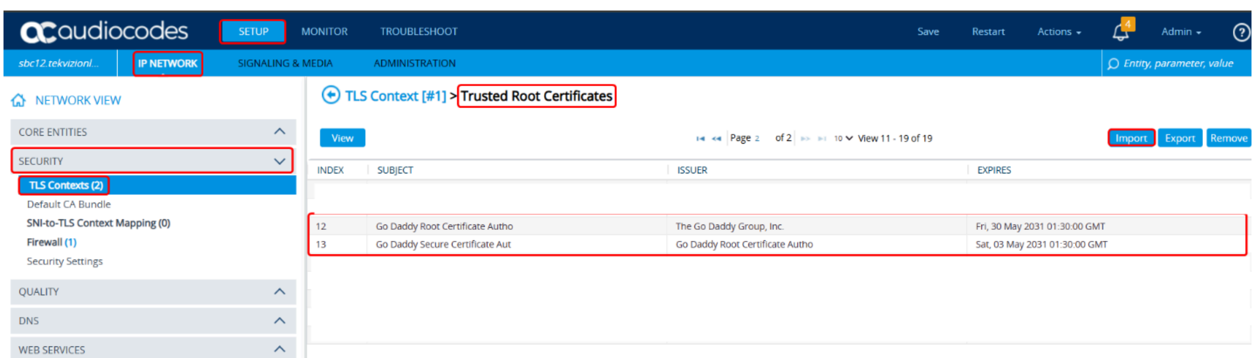


Figure 12: Import intermediate Certificates

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.
- 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.

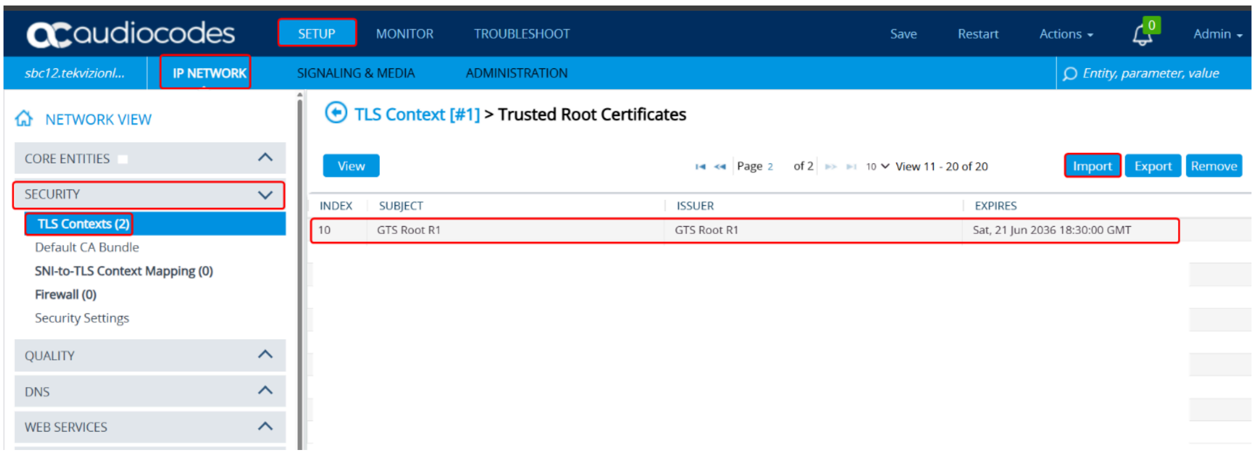


Figure : GTS Root R1 Certificate

7.4.3 Configure Media Realms

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

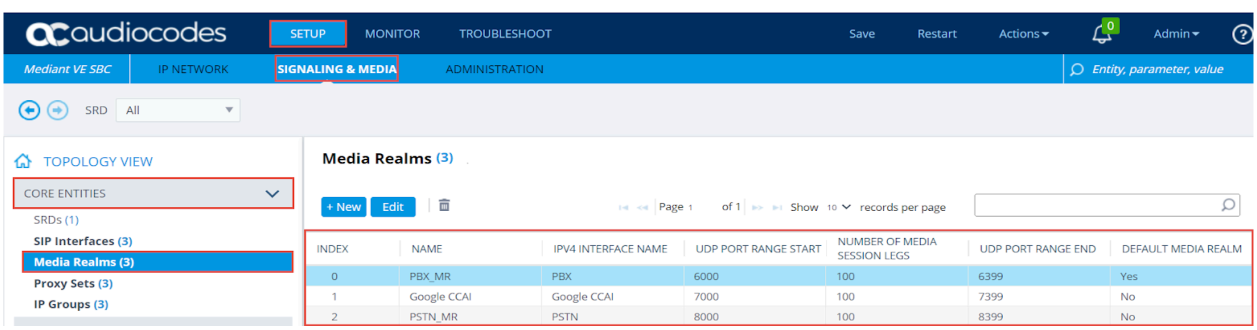


Figure : Configure Media Realms

7.4.4 Configure SIP Signaling Interfaces

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

OnPrem PBX:

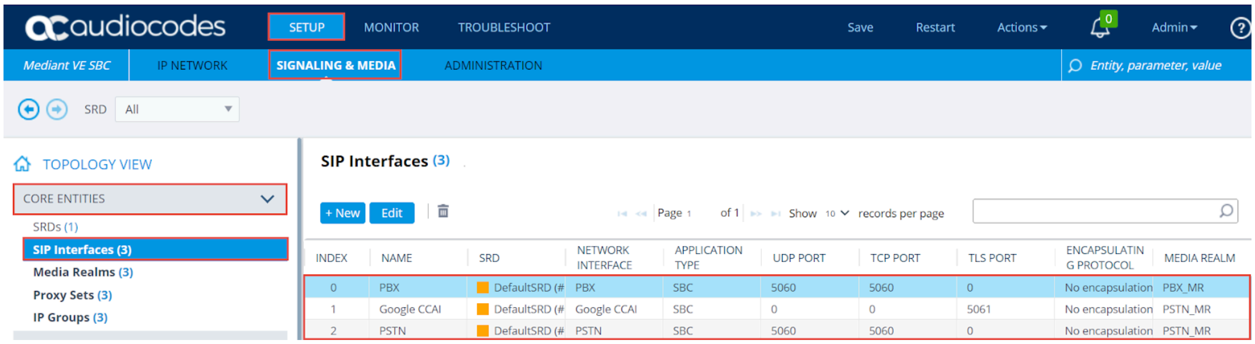


Figure : SIP Signaling Interfaces

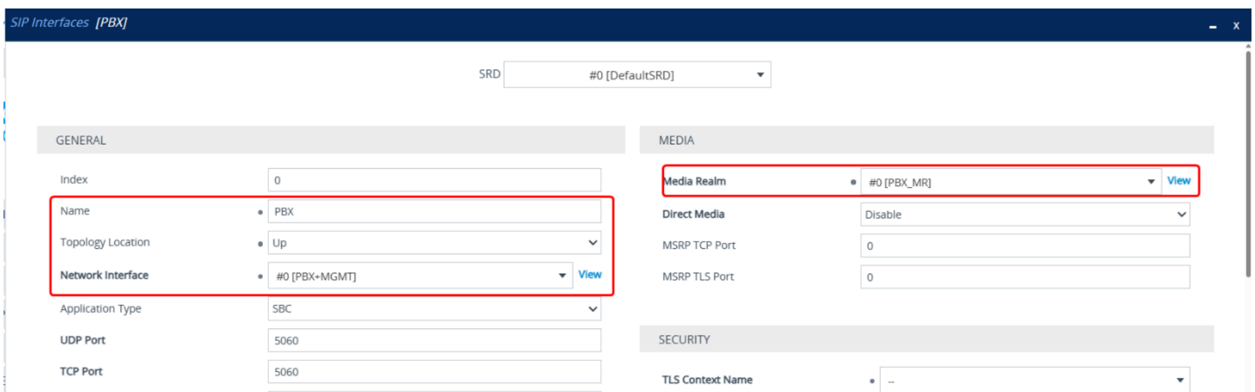


Figure : SIP Signaling Interfaces for OnPrem PBX

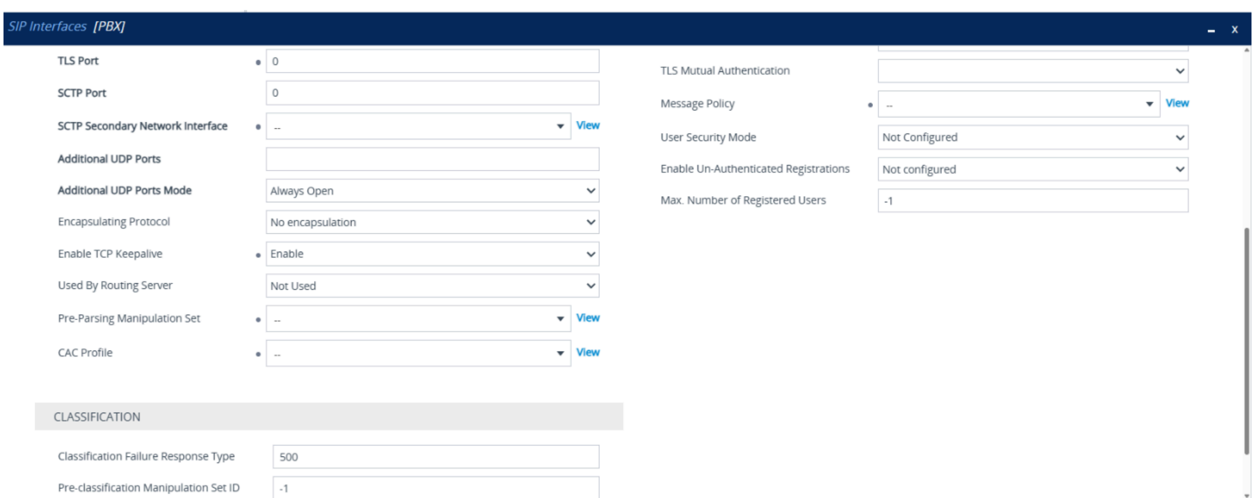


Figure 17 : SIP Signaling Interfaces for OnPrem PBX (Cont.)

Google CES:

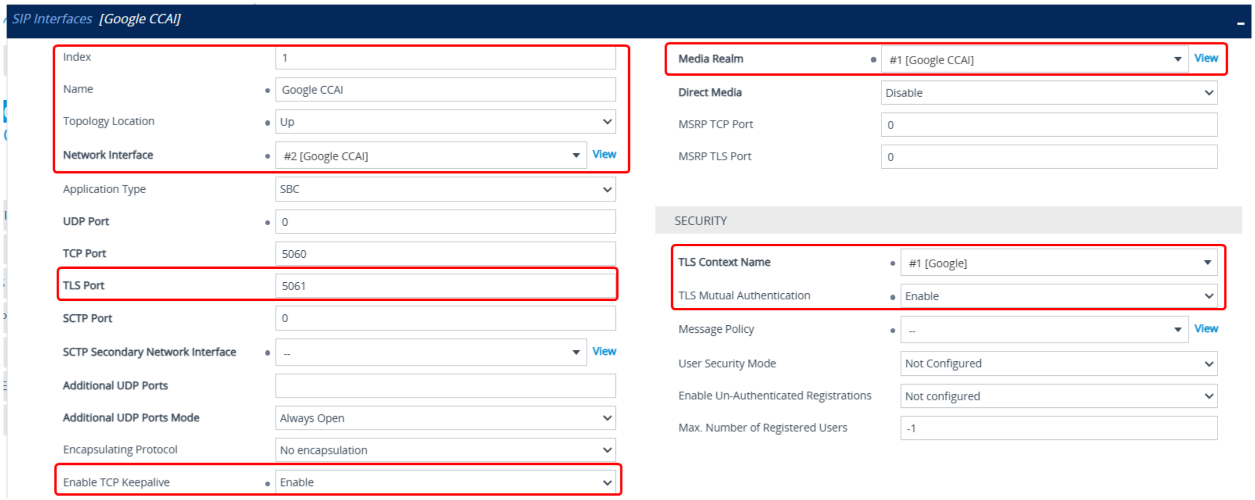


Figure : SIP Signaling Interfaces for Google CES

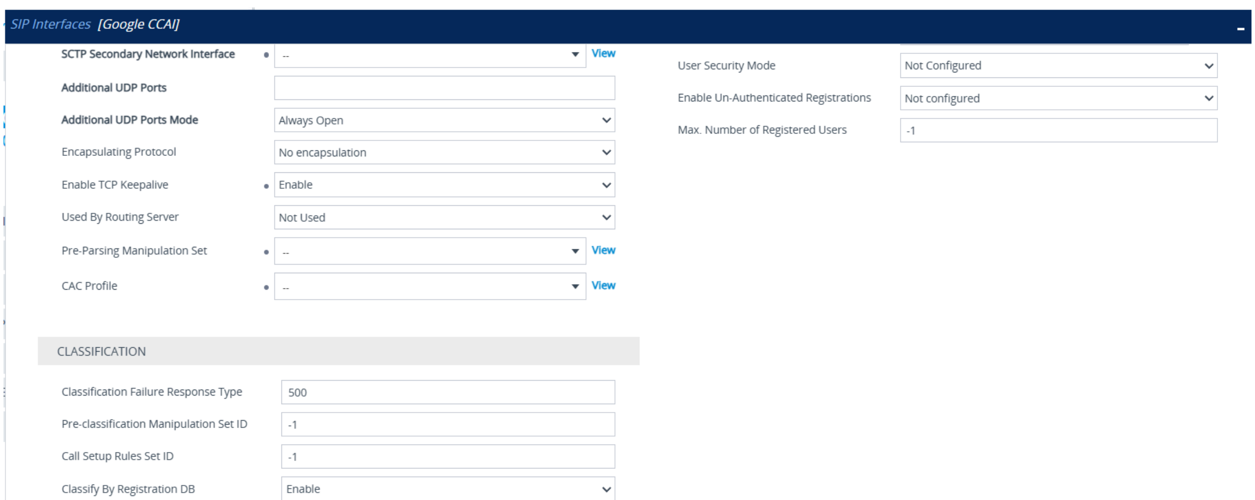


Figure 19: SIP Signaling Interfaces for Google CES (Cont.)

PSTN Gateway:

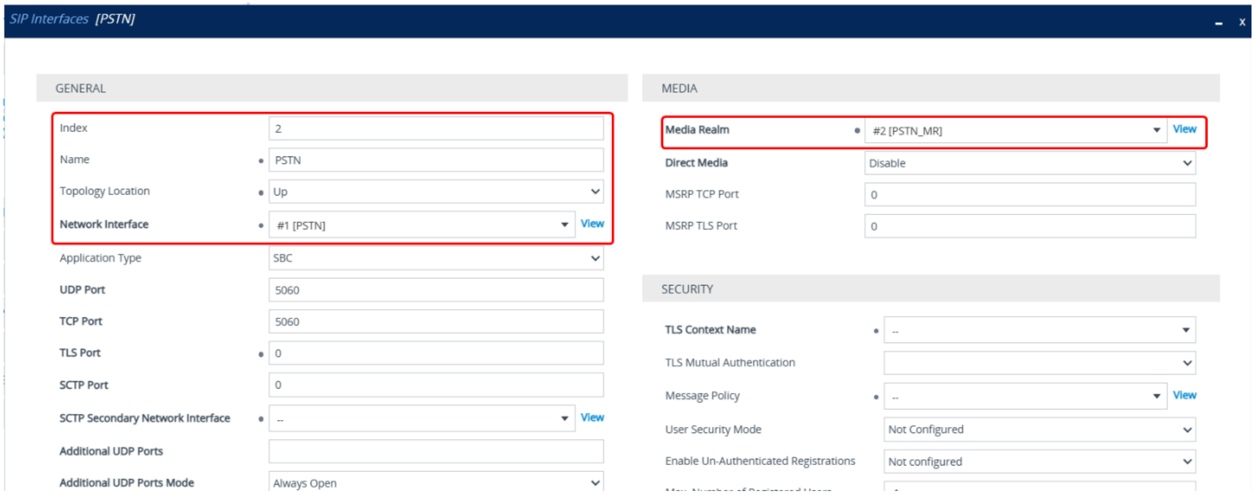


Figure : SIP Signaling Interfaces for PSTN Gateway

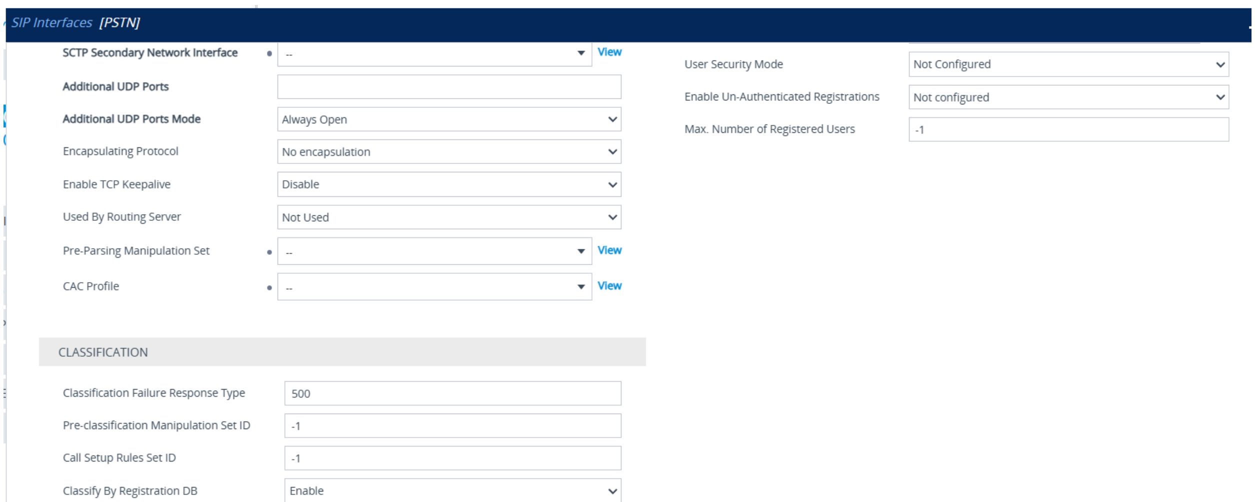


Figure 21: SIP Signaling Interfaces for PSTN Gateway (Cont.)

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 OnPrem PBX, PSTN Gateway and Google CES as shown below.

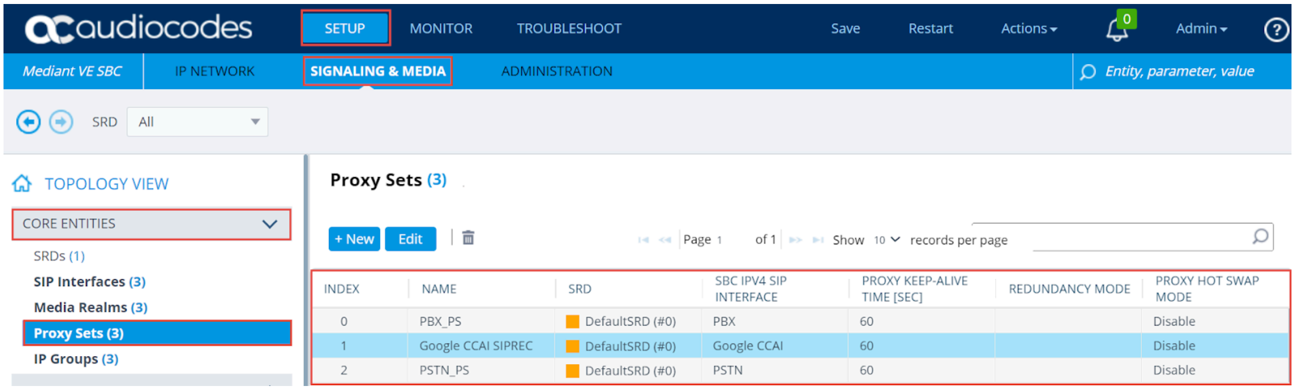


Figure : Configurations of Proxy Sets

OnPrem PBX:

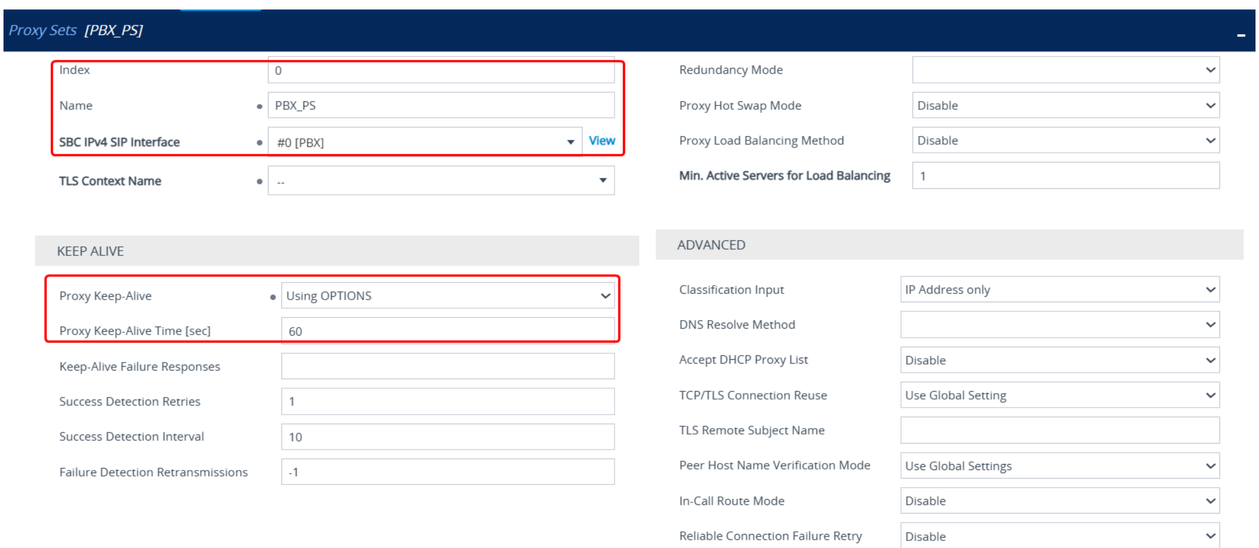


Figure : Proxy Set Configuration of OnPrem PBX

Google CES:

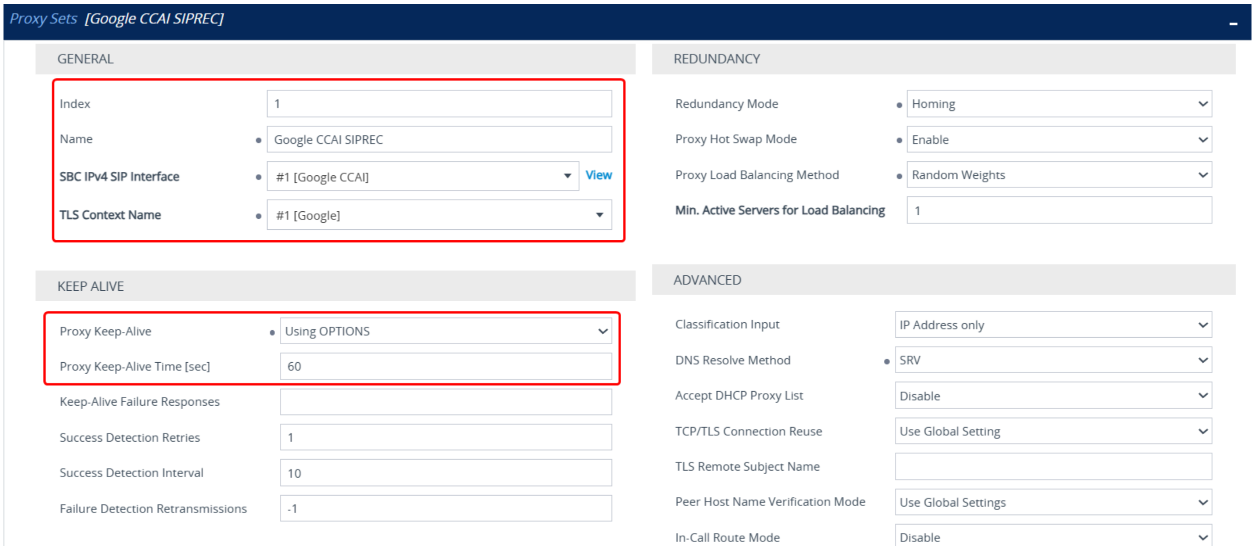


Figure : Proxy Set Configuration of Google CES

PSTN Gateway:

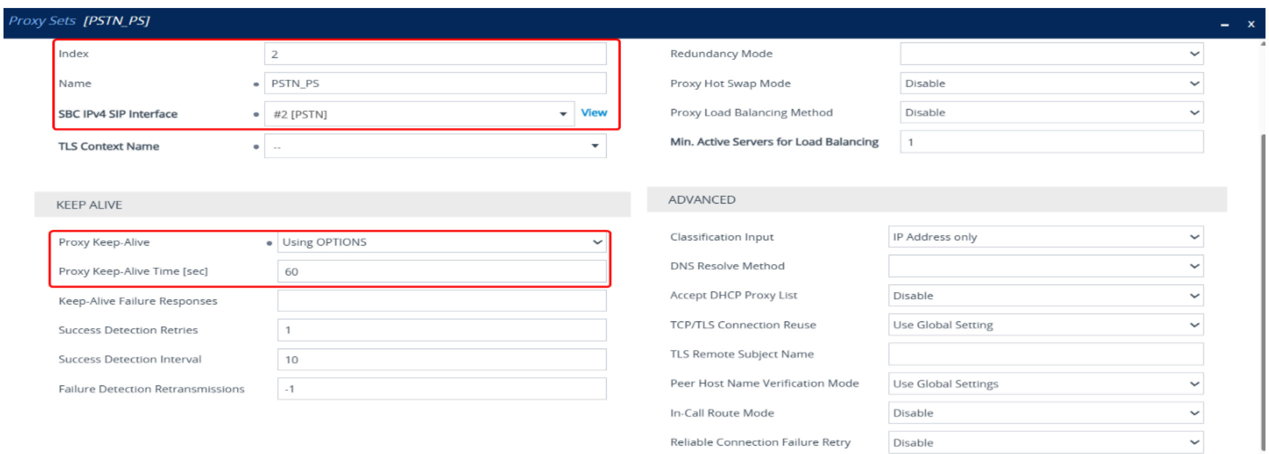


Figure : Proxy Set Configuration of PSTN Gateway

Proxy Address Configuration:

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

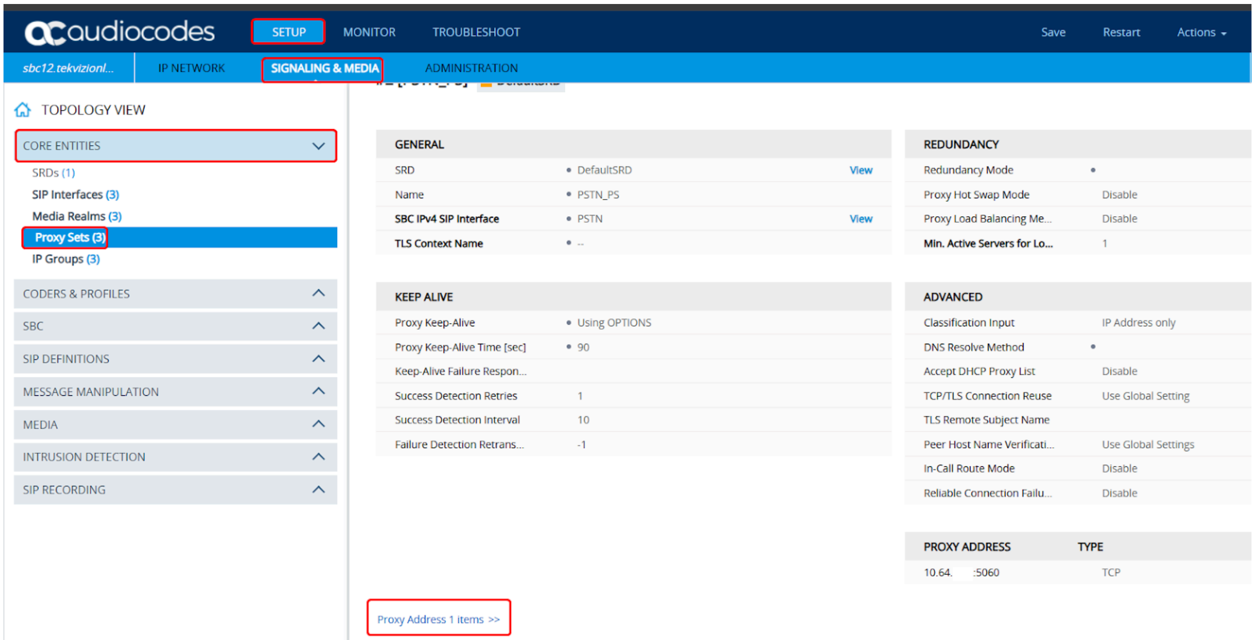


Figure 26: Proxy Address Configuration

- Enter the OnPrem PBX IP as Proxy Address in the PBX Proxy set and select transport type as TCP

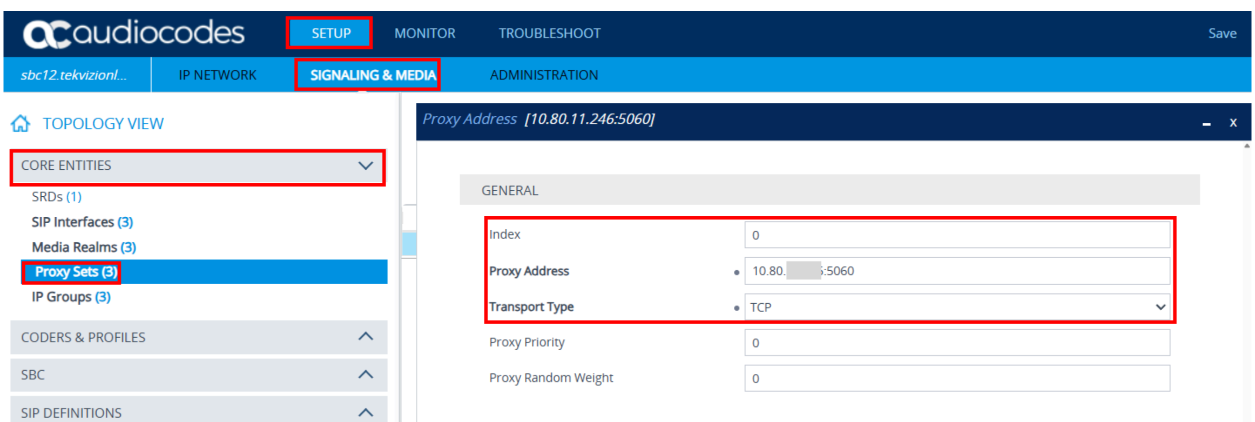


Figure : Proxy Address Configuration of OnPrem PBX

- Select the Google CES SIPREC Proxy Set and add the Proxy Address by clicking **Proxy Address items>>** and **+New.**
- Enter the Google CES FQDN as proxy Address in the Google Proxy set and select transport type as TLS

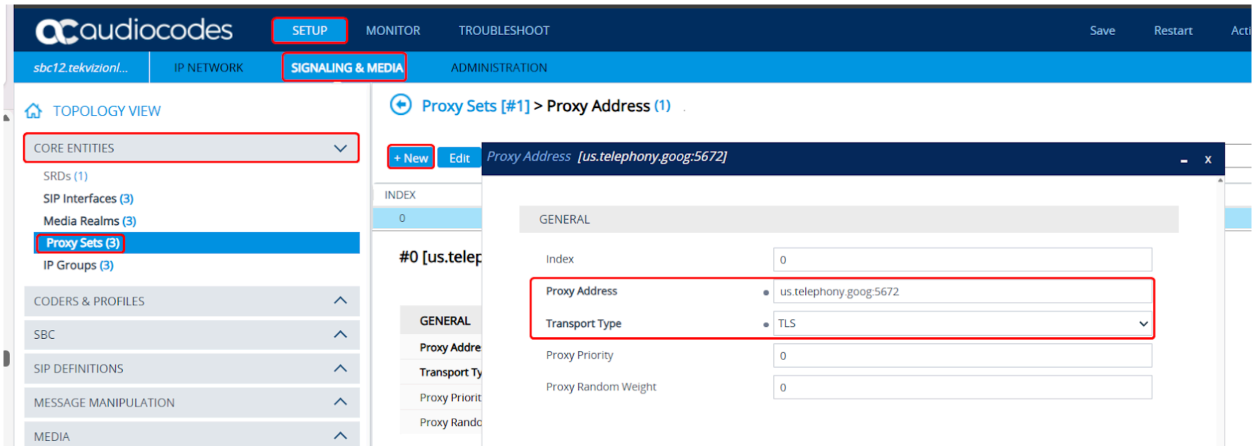


Figure : Proxy Address Configuration of Google CES

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

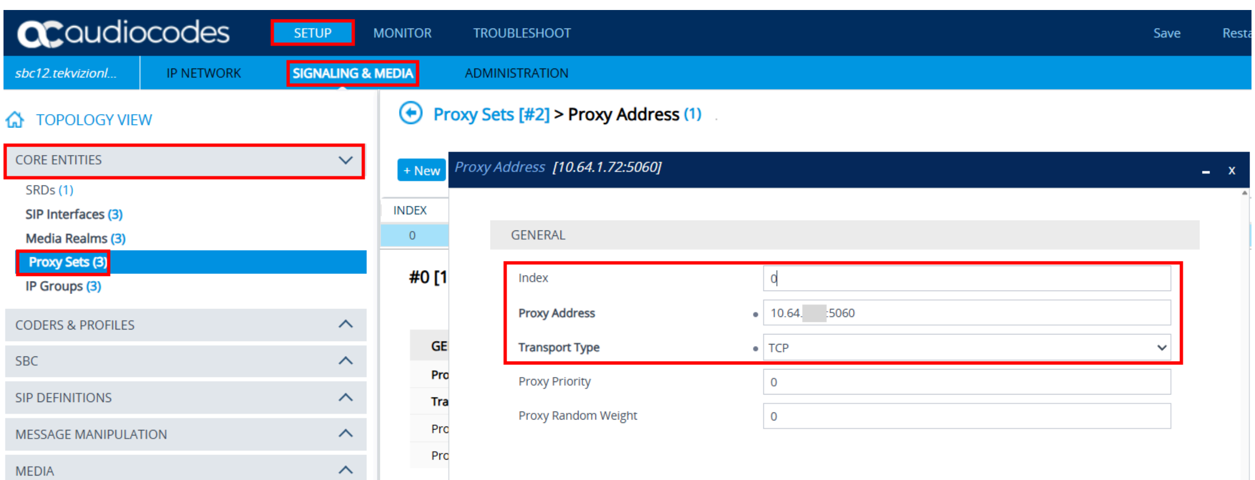


Figure 29: Proxy Address Configuration of PSTN Gateway

7.4.6 Configure Coders

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

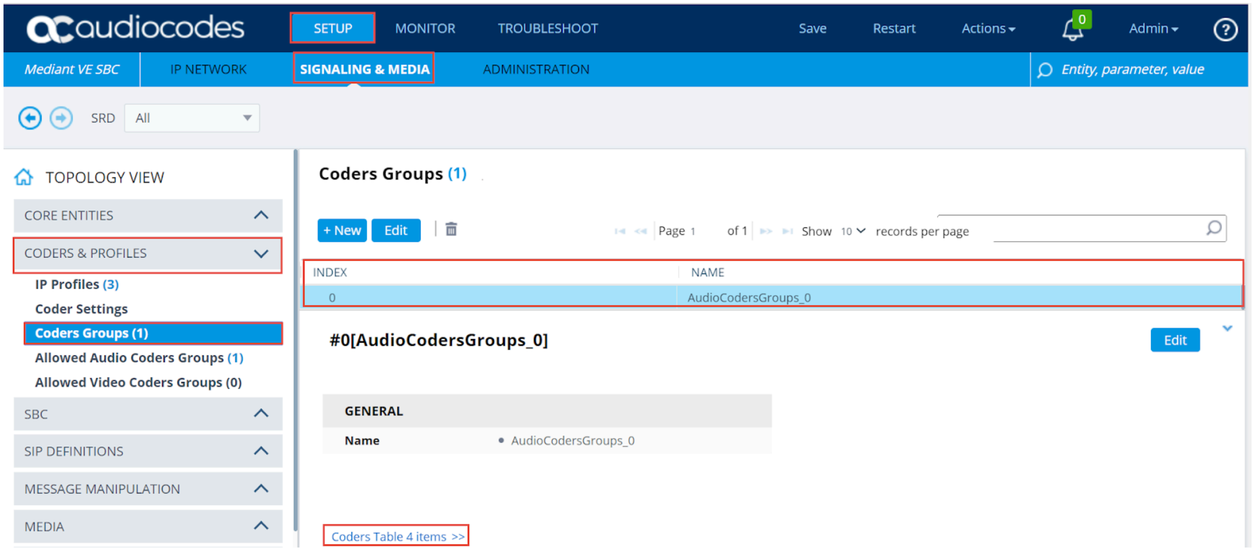


Figure 30: Coders Configurations

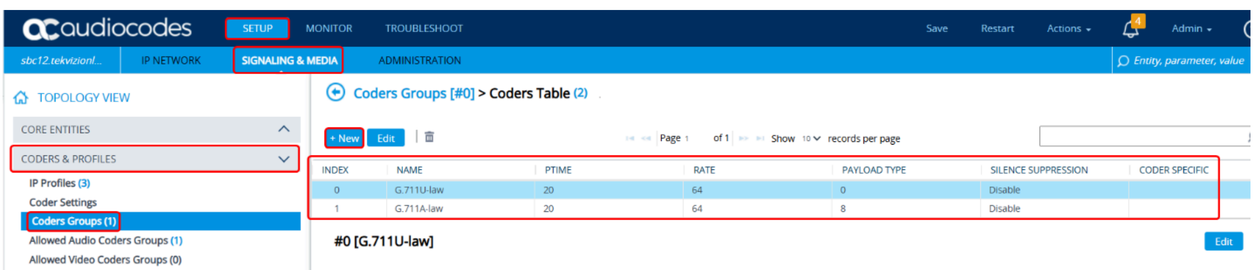


Figure 31: Coders Configurations (Cont.)

To Set a preferred coder for the Google CES:

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

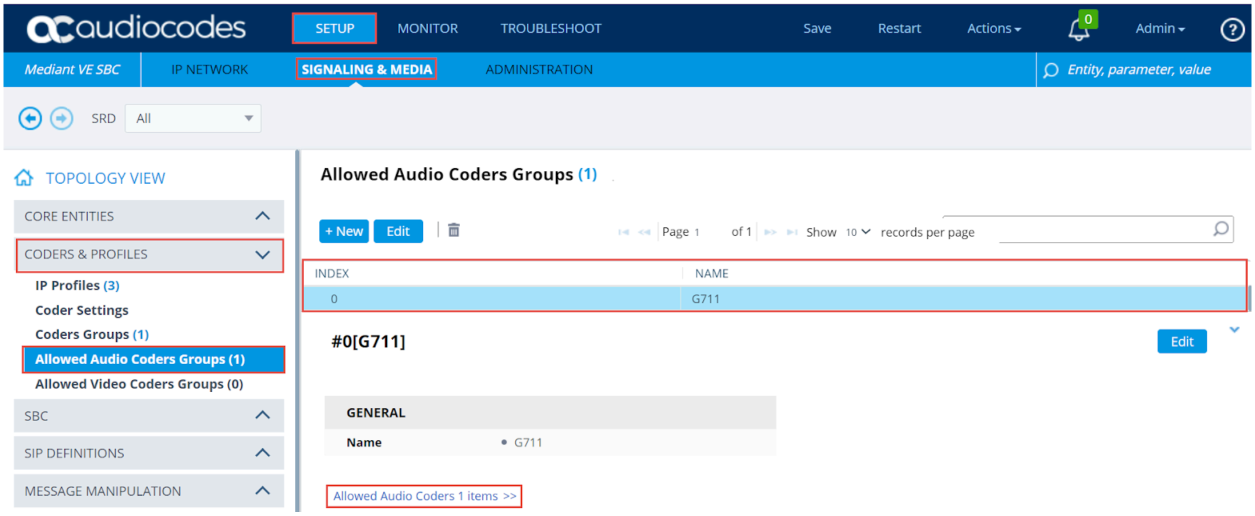


Figure 32: Coders Configurations (Cont.)

- 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.

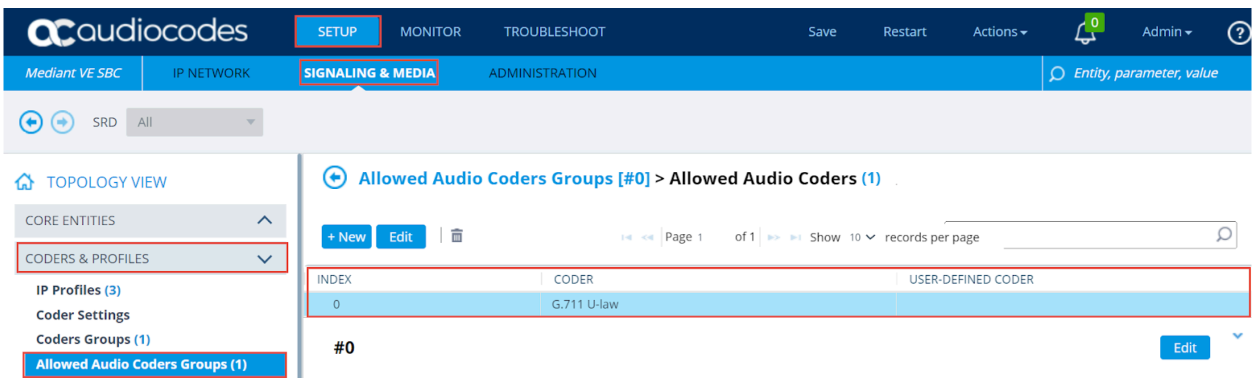


Figure : Coders Configurations (Cont.)

7.4.7 Configure IP Profiles

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

OnPrem PBX IP:

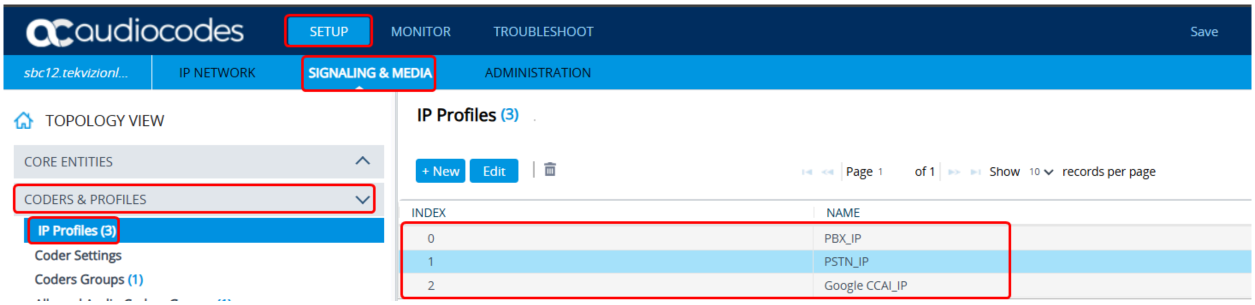


Figure 34: IP Profile Configurations

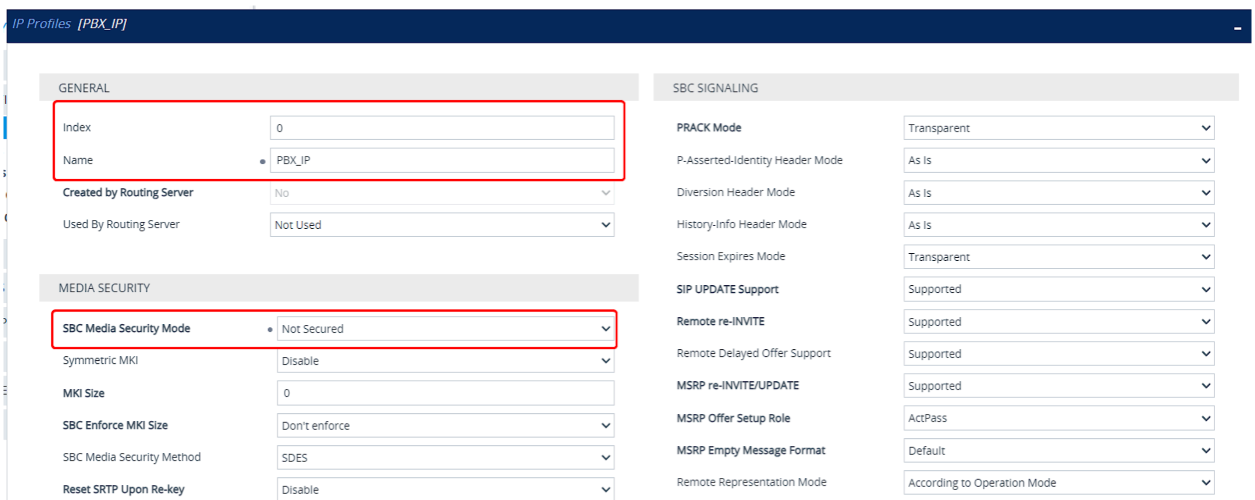


Figure 35: IP Profile Configurations of OnPrem PBX

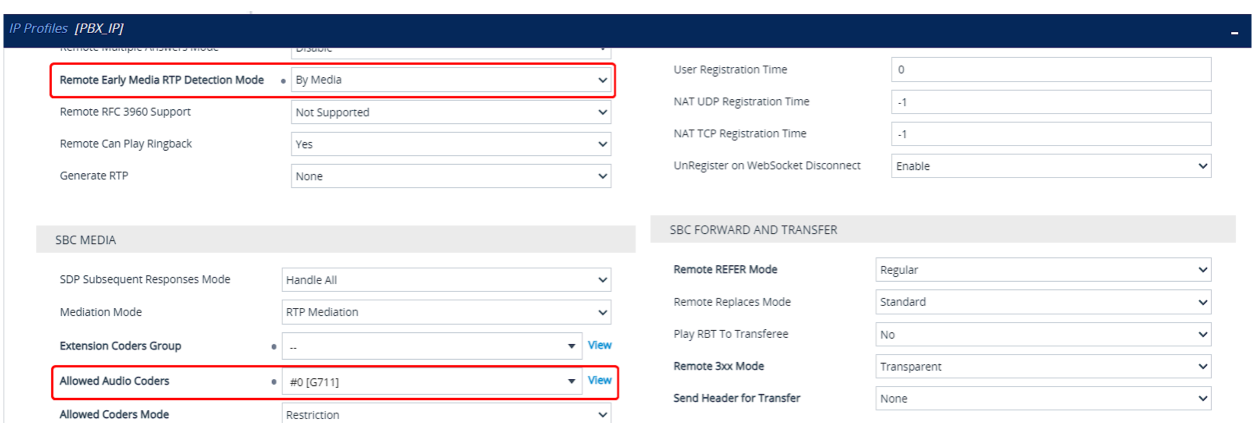


Figure 36: IP Profile Configurations of OnPrem PBX (Cont.)

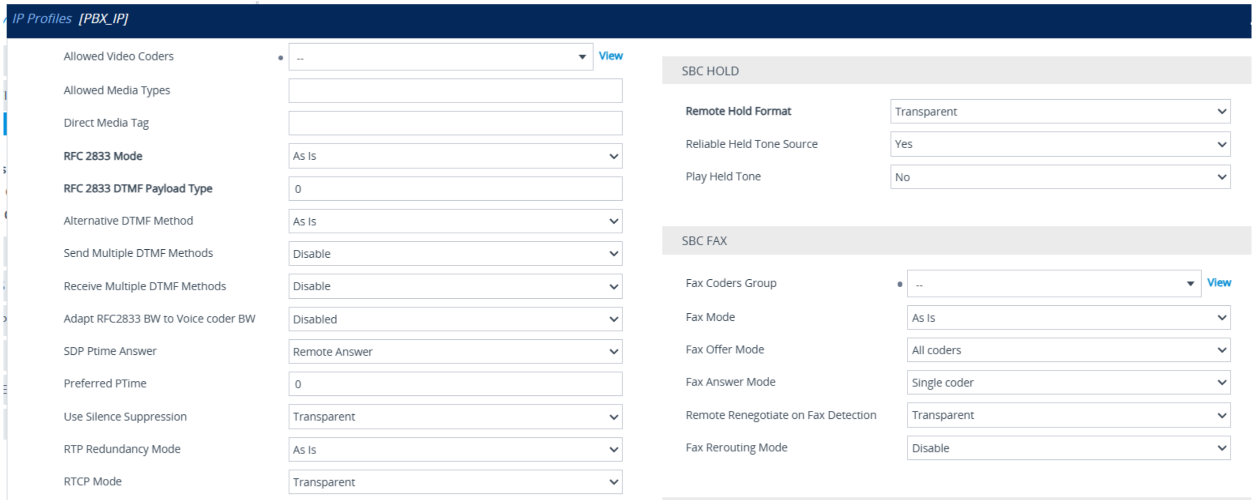


Figure 37: IP Profile Configurations of OnPrem PBX (Cont.)

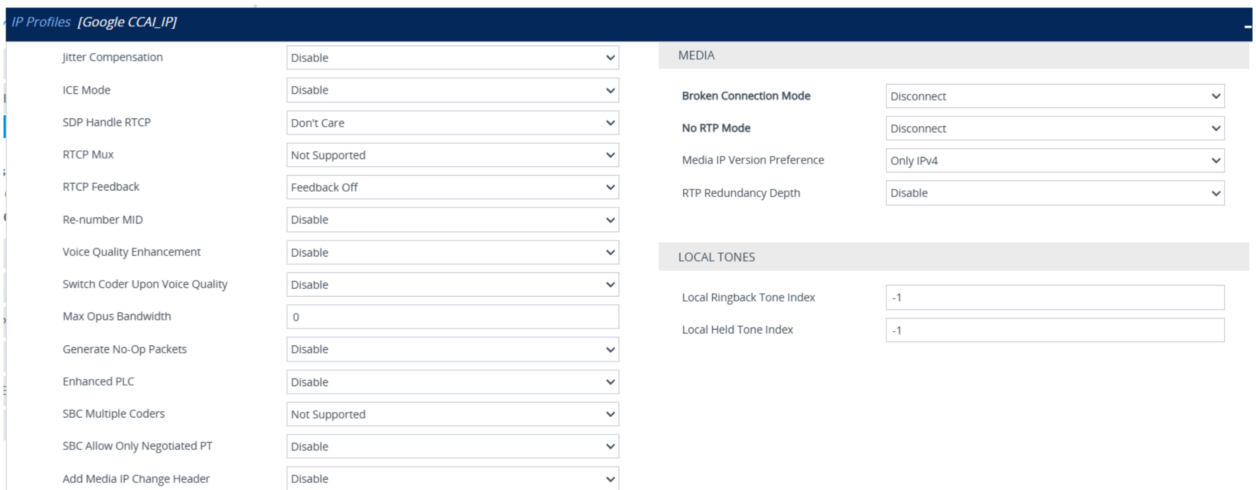


Figure 38: IP Profile Configurations of OnPrem PBX (Cont.)

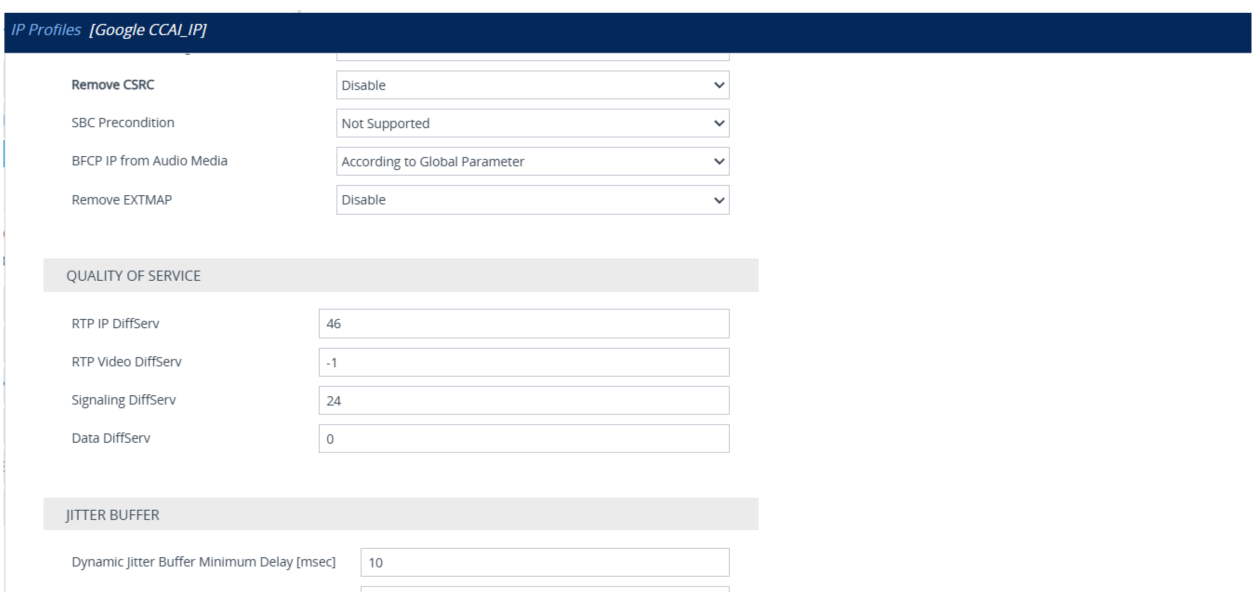


Figure 39: IP Profile Configurations of OnPrem PBX (Cont.)

PSTN Gateway:

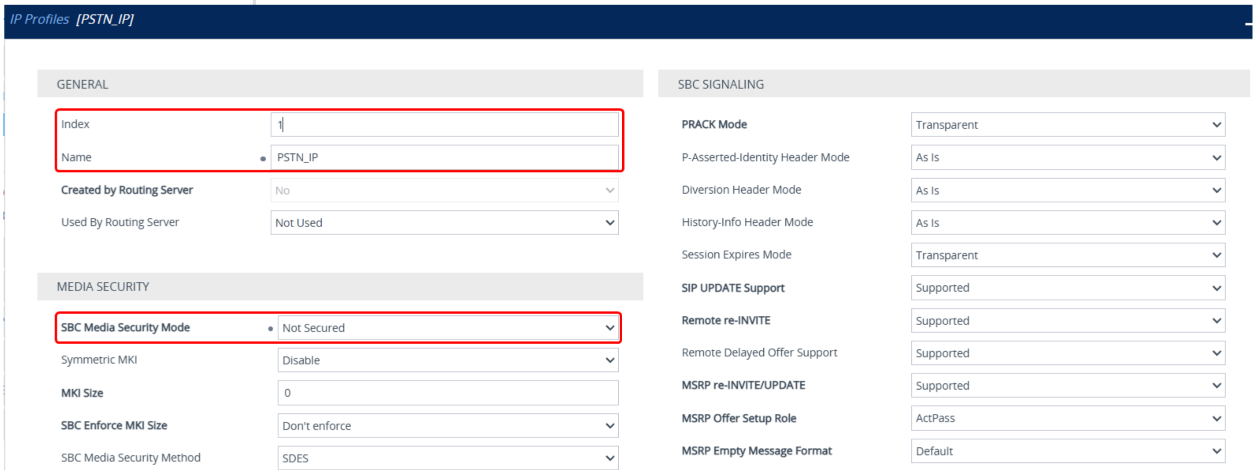


Figure 40: IP Profile Configurations of PSTN Gateway

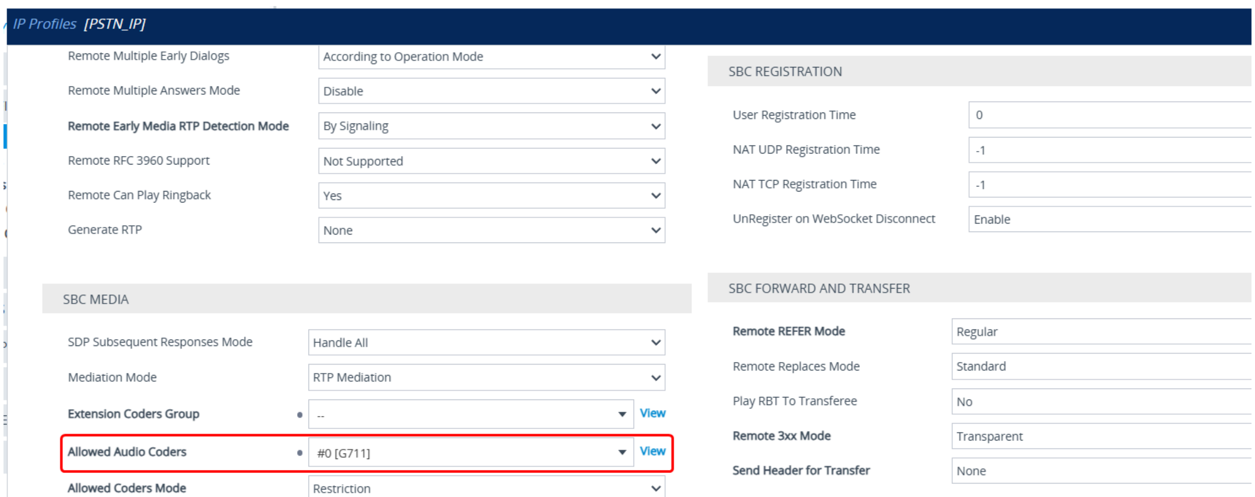


Figure 41: IP Profile Configurations of PSTN Gateway (Cont.)

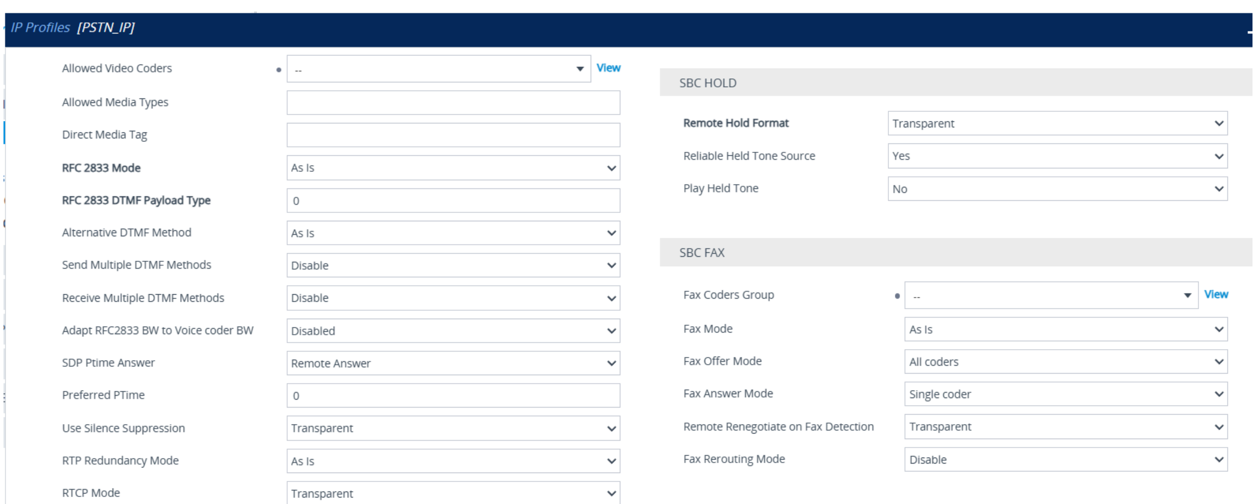


Figure 42: IP Profile Configurations of PSTN Gateway (Cont.)

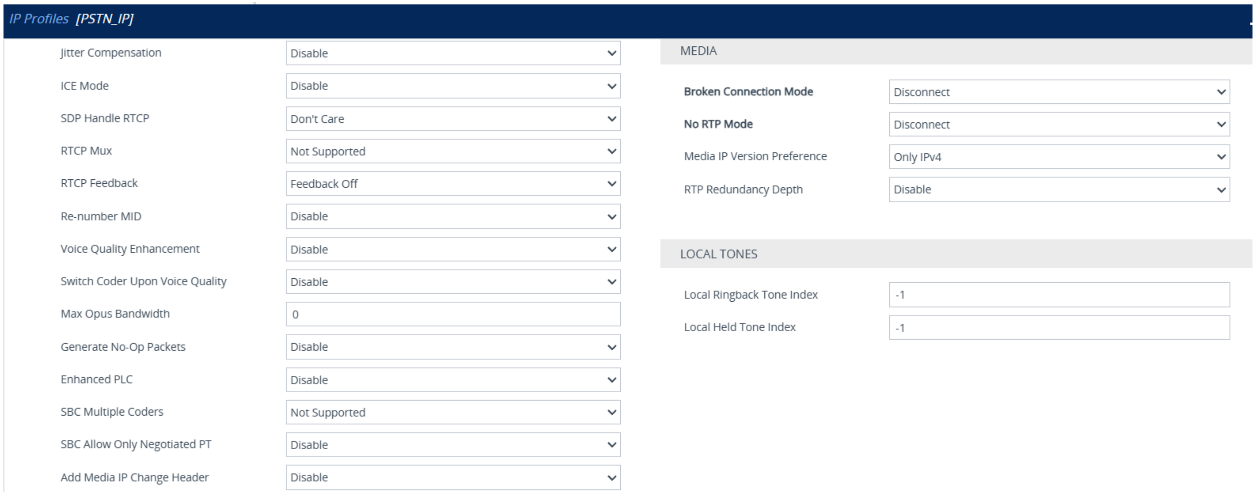


Figure 43:IP Profile Configurations of PSTN Gateway (Cont.)

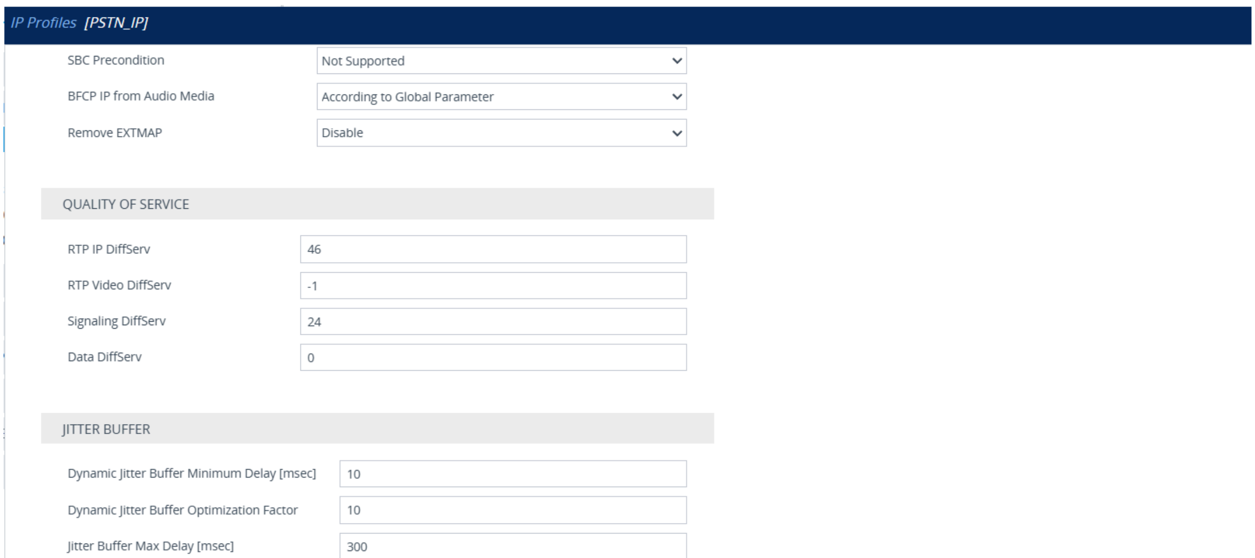


Figure 44:IP Profile Configurations of PSTN Gateway (Cont.)

Google CES:

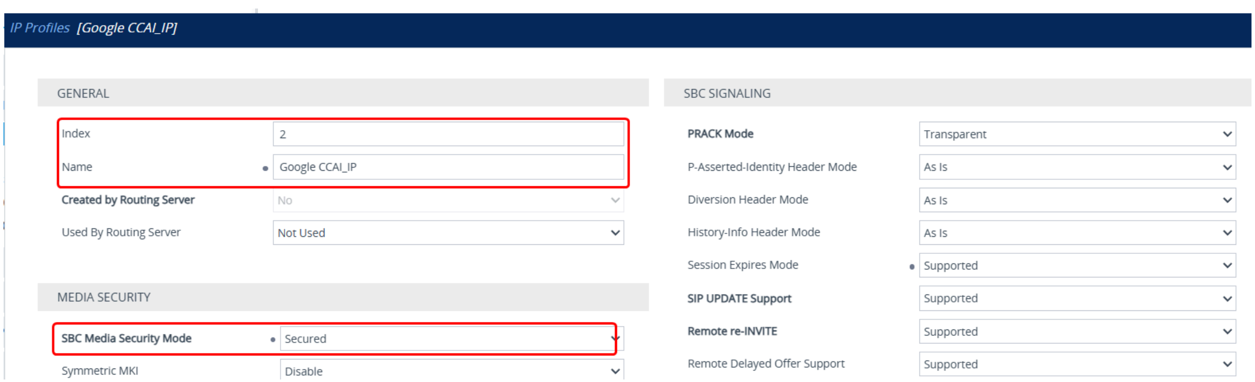


Figure 45: IP Profile configurations of Google CES

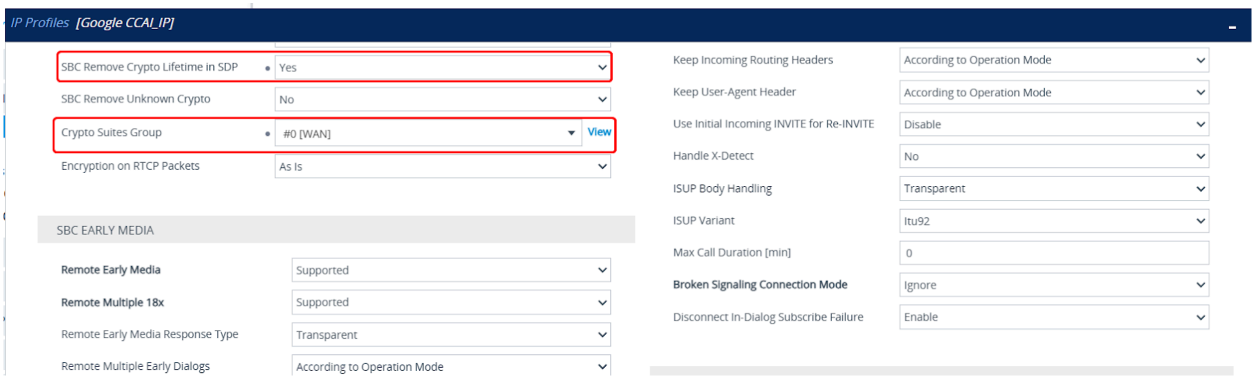


Figure 46: IP Profile Configurations of Google CES (Cont.)

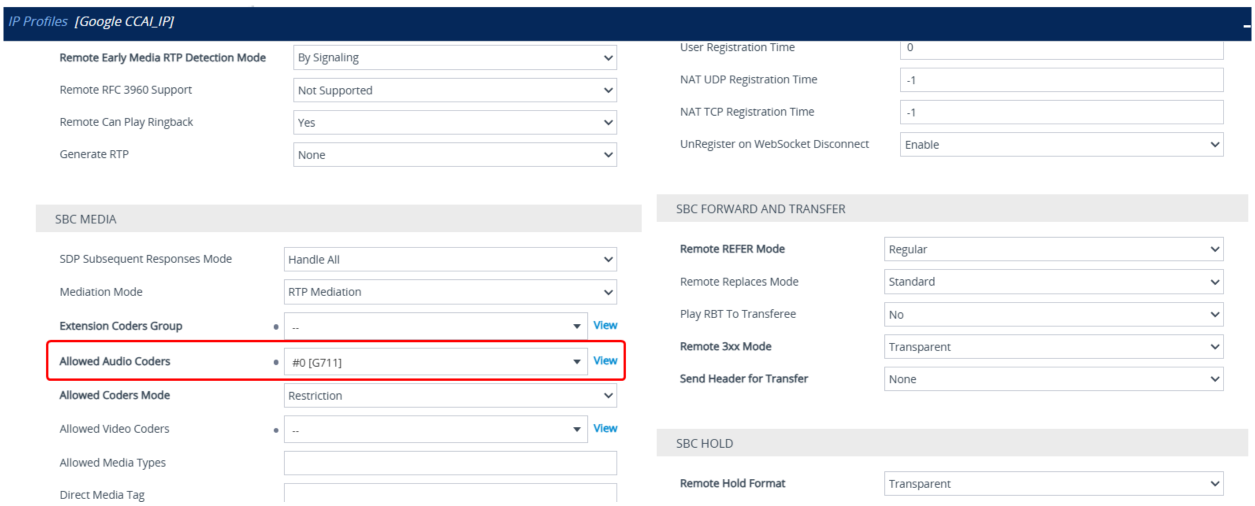


Figure 47: IP Profile Configurations of Google CES (Cont.)

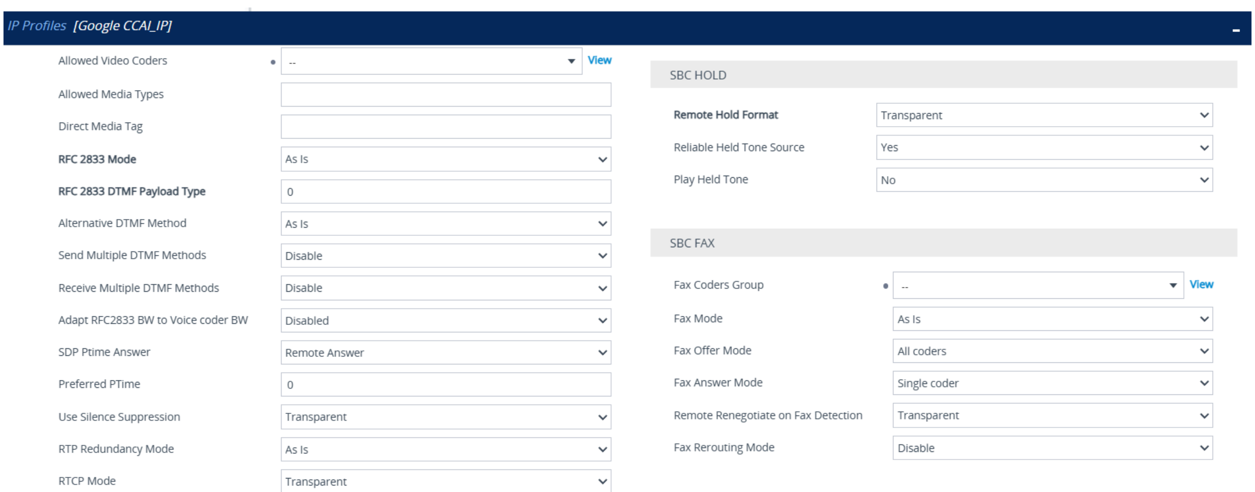


Figure 48: IP Profile Configurations of Google CES (Cont.)

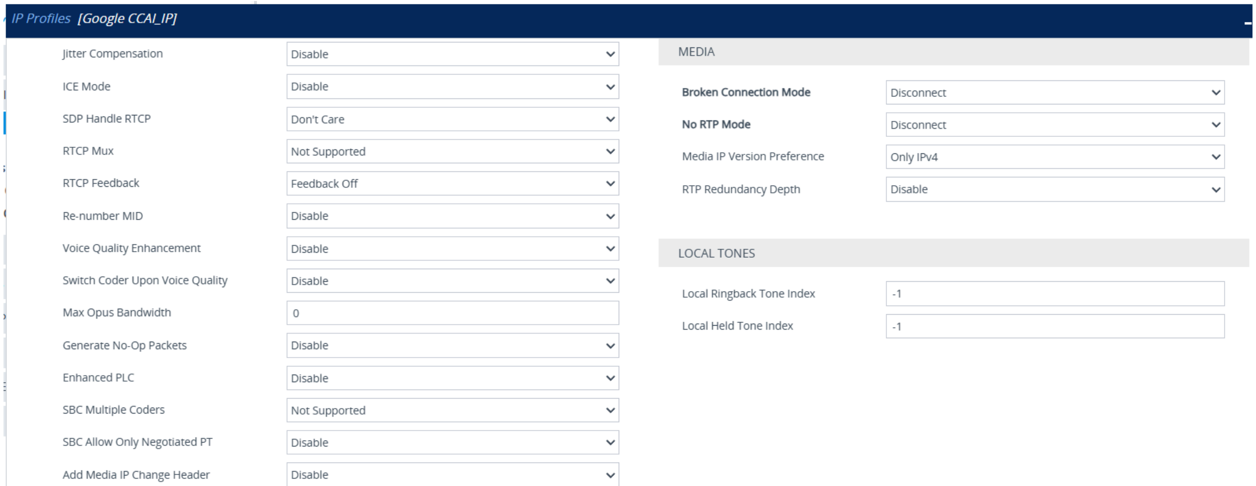


Figure 49: IP Profile Configurations of Google CES (Cont.)

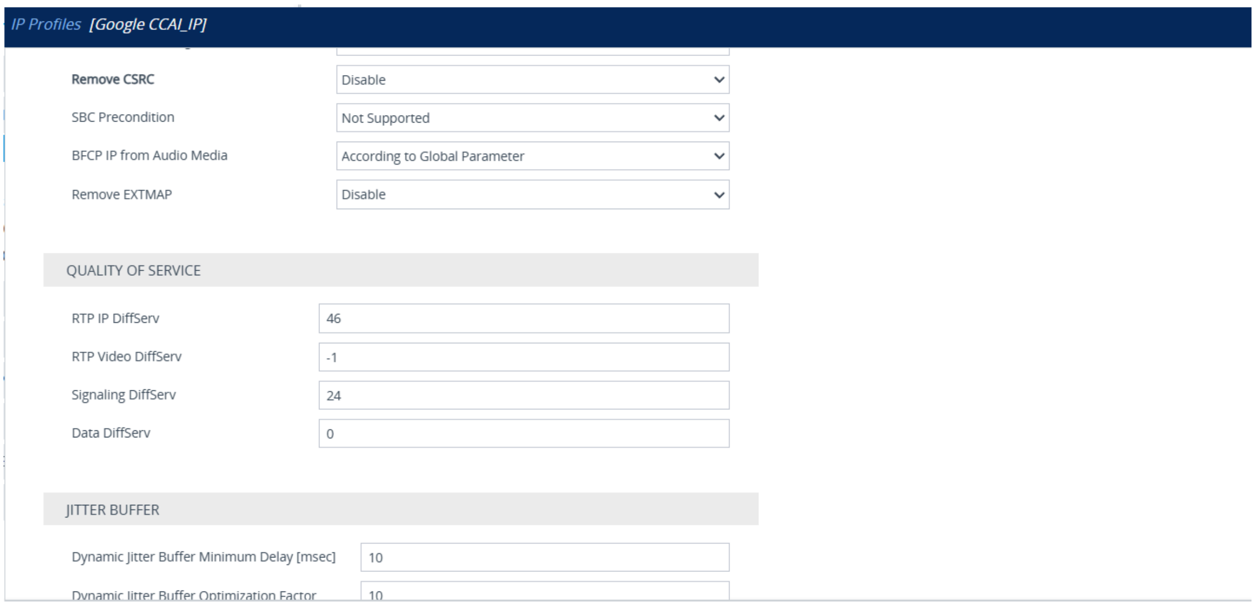


Figure 50: IP Profile Configurations of Google CES (Cont.)

7.4.8 Configure IP Groups

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

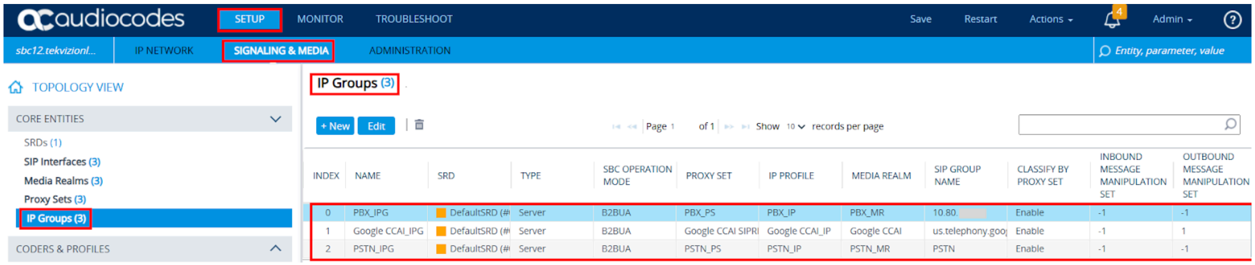


Figure 51:IP Group Configurations

- Select the respective Proxy Set, IP Profile, and Media Realm for OnPrem PBX IP Group and enter the OnPrem PBX IP as SIP Group name

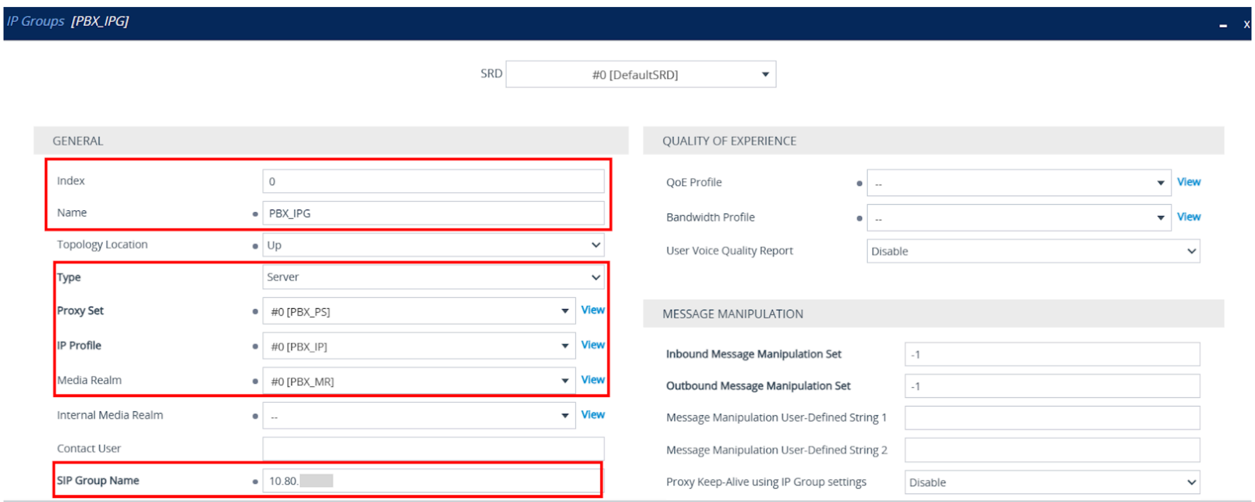


Figure 52: IP Group configurations of OnPrem PBX

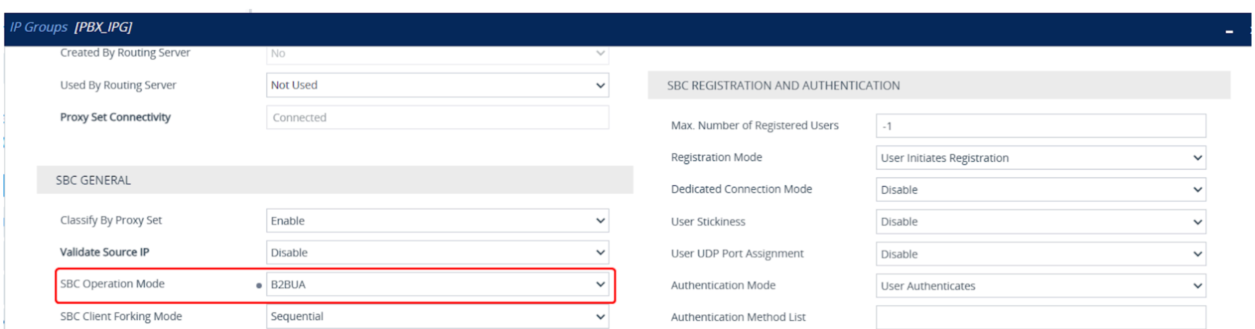


Figure 53: IP Group Configurations of OnPrem PBX (Cont.)

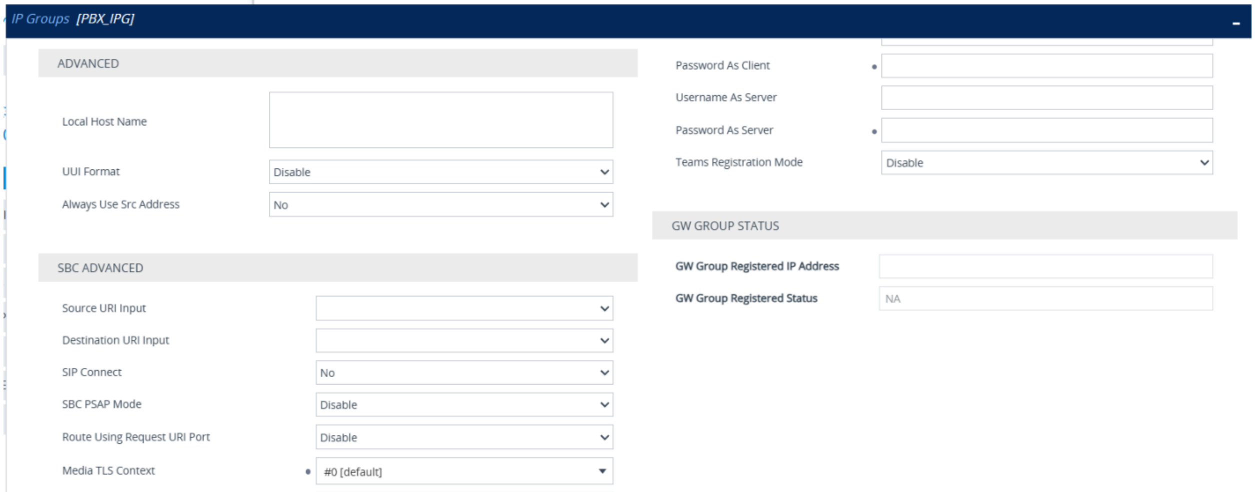


Figure : IP Group Configurations of OnPrem PBX (Cont.)

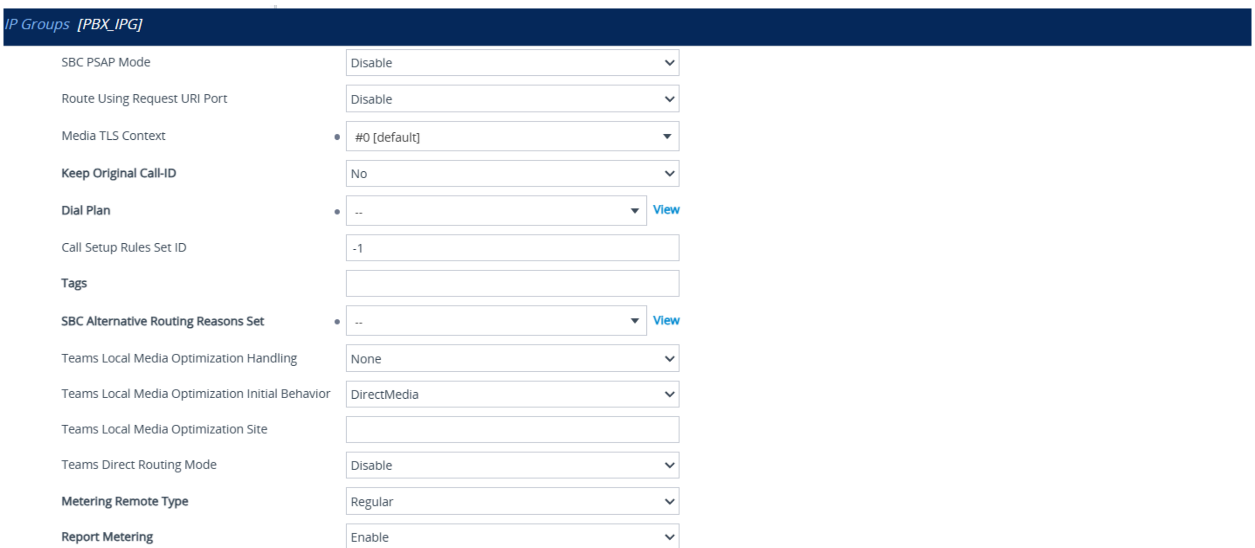


Figure 55: IP Group Configurations of OnPrem PBX (Cont.)

- 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.

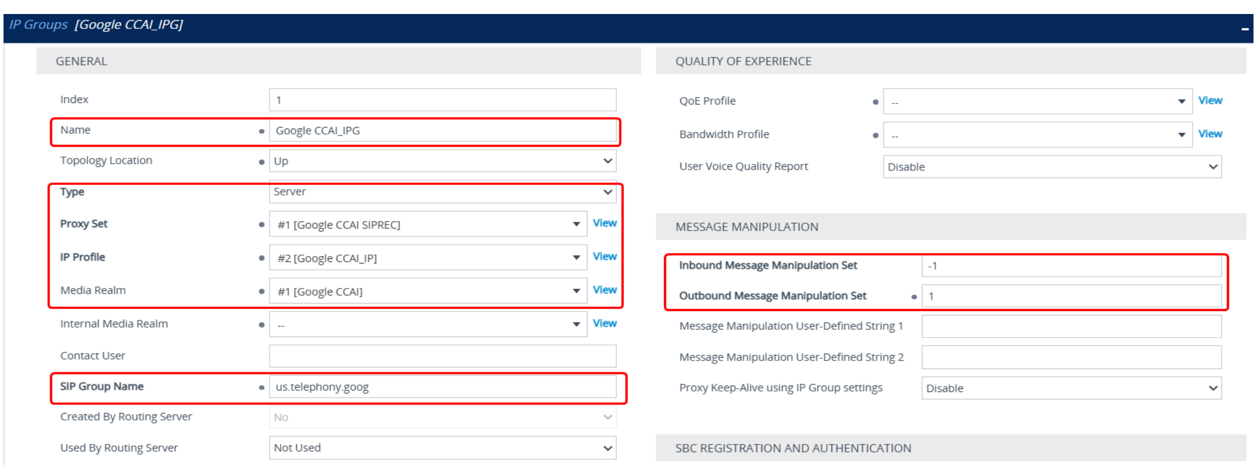


Figure 56: IP Group Configurations of Google CES

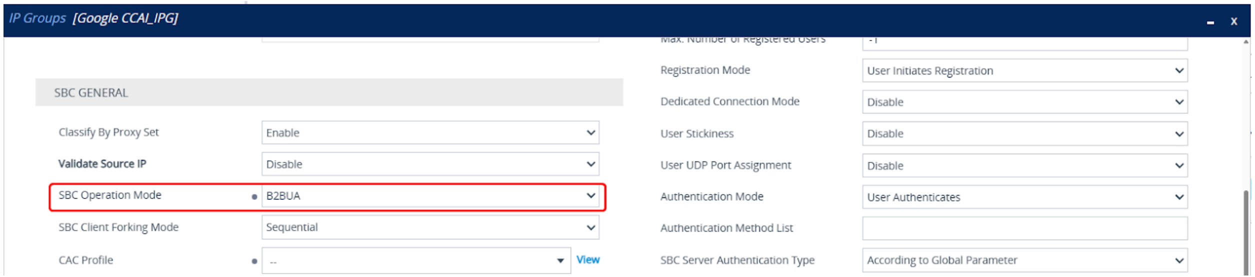


Figure 57: IP Group Configurations of Google CES (Cont.)

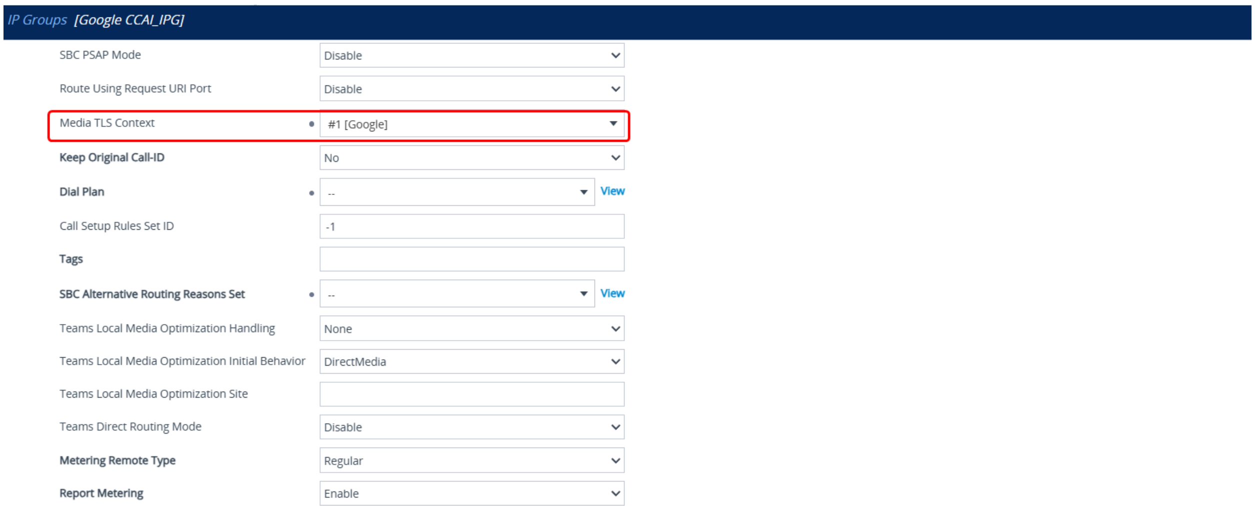


Figure 58: IP Group Configurations of Google CES (Cont.)

- 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.

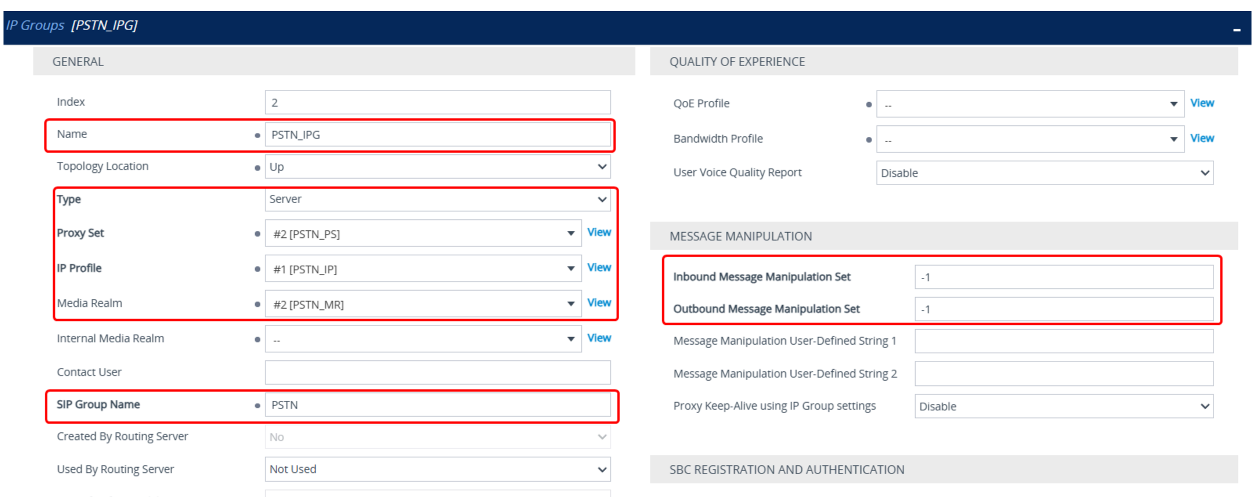


Figure 59: IP Group Configurations of PSTN Gateway

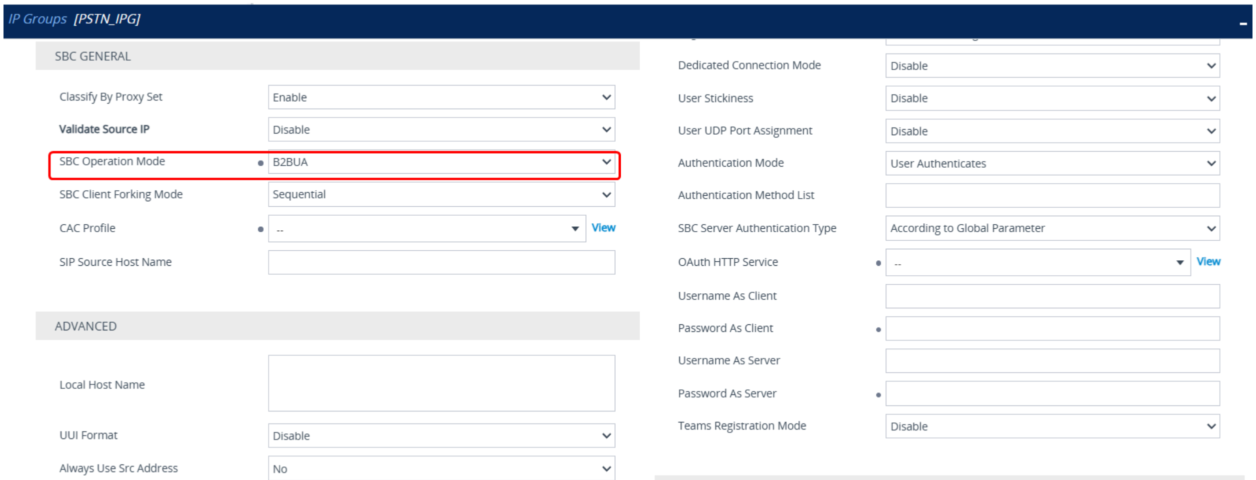


Figure 60: IP Group Configurations of PSTN Gateway (Cont.)

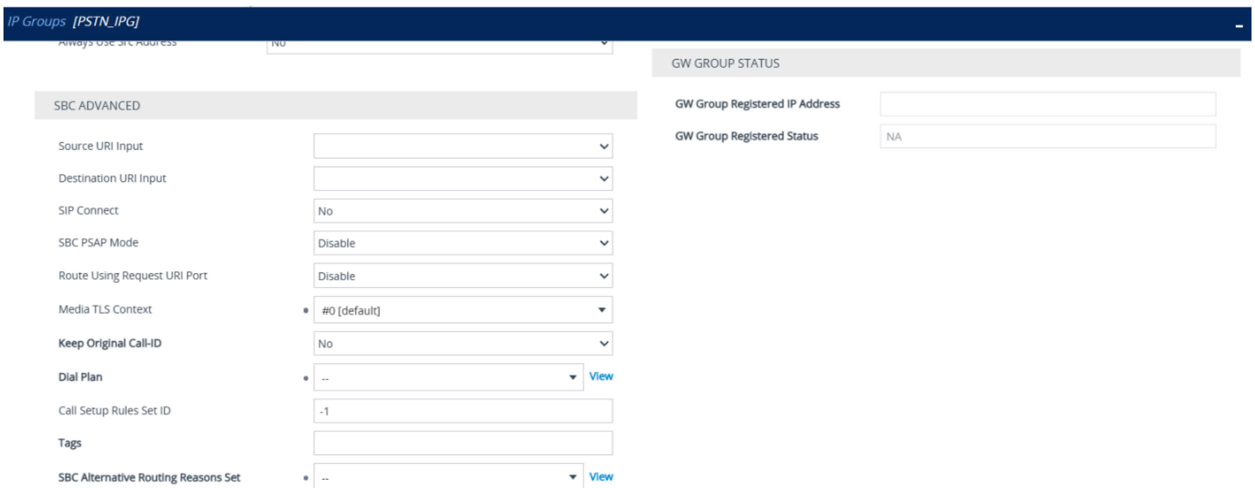


Figure 61: IP Group Configurations of PSTN Gateway (Cont.)

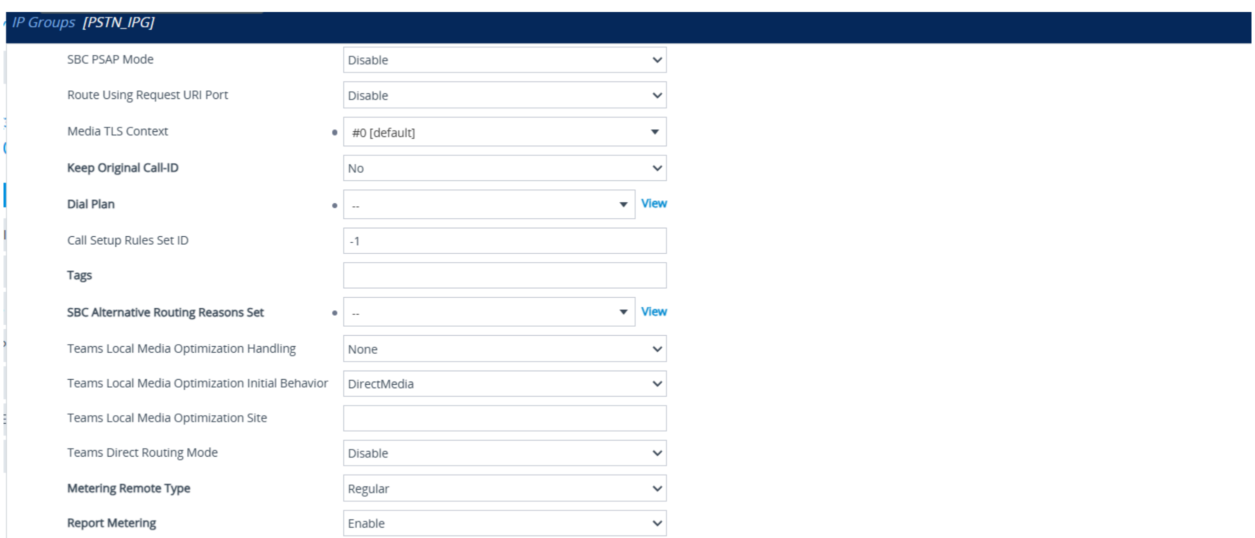


Figure 62: IP Group Configurations of PSTN Gateway (Cont.)

7.4.9 Configure Media Security

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

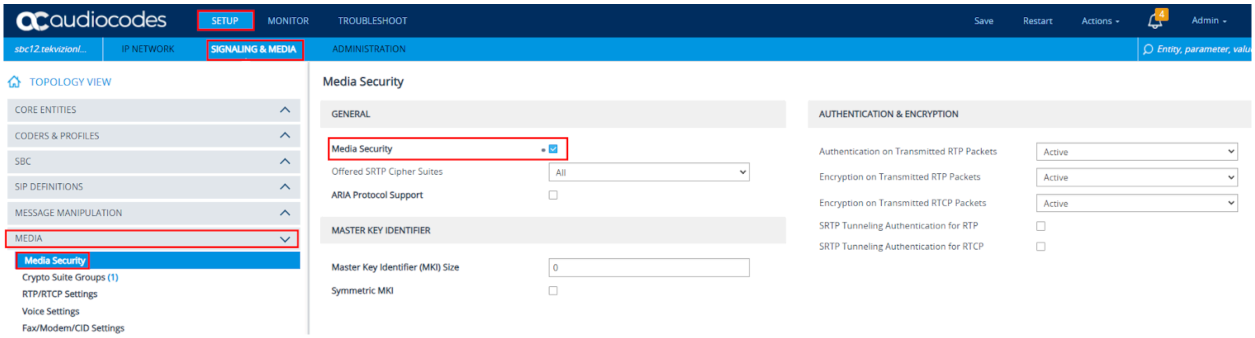


Figure 63: Media Security Configuration

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.

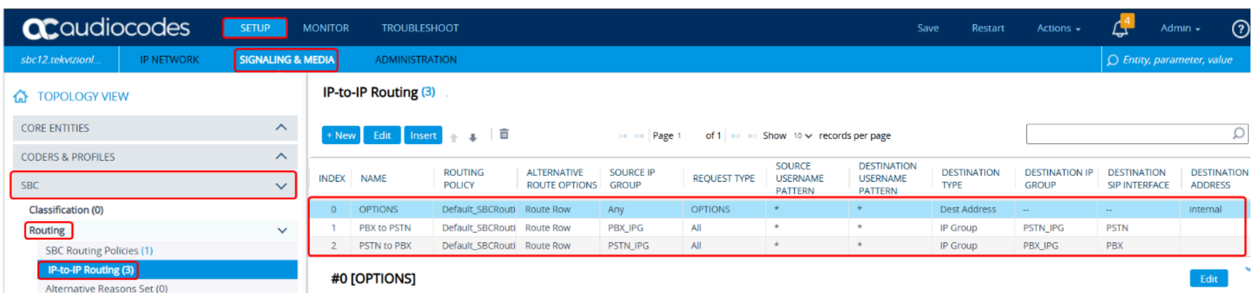


Figure 64: IP to IP Routing

7.4.11 Configure SIP Recording

- Navigate to **SETUP** menu □ **SIGNALING & MEDIA** tab □ **SIP RECORDING** folder □ **SIP Recording Settings**
- Configure Recording Server (SRS) Destination Username as Pilot number of Google CES SIPREC number as shown below.

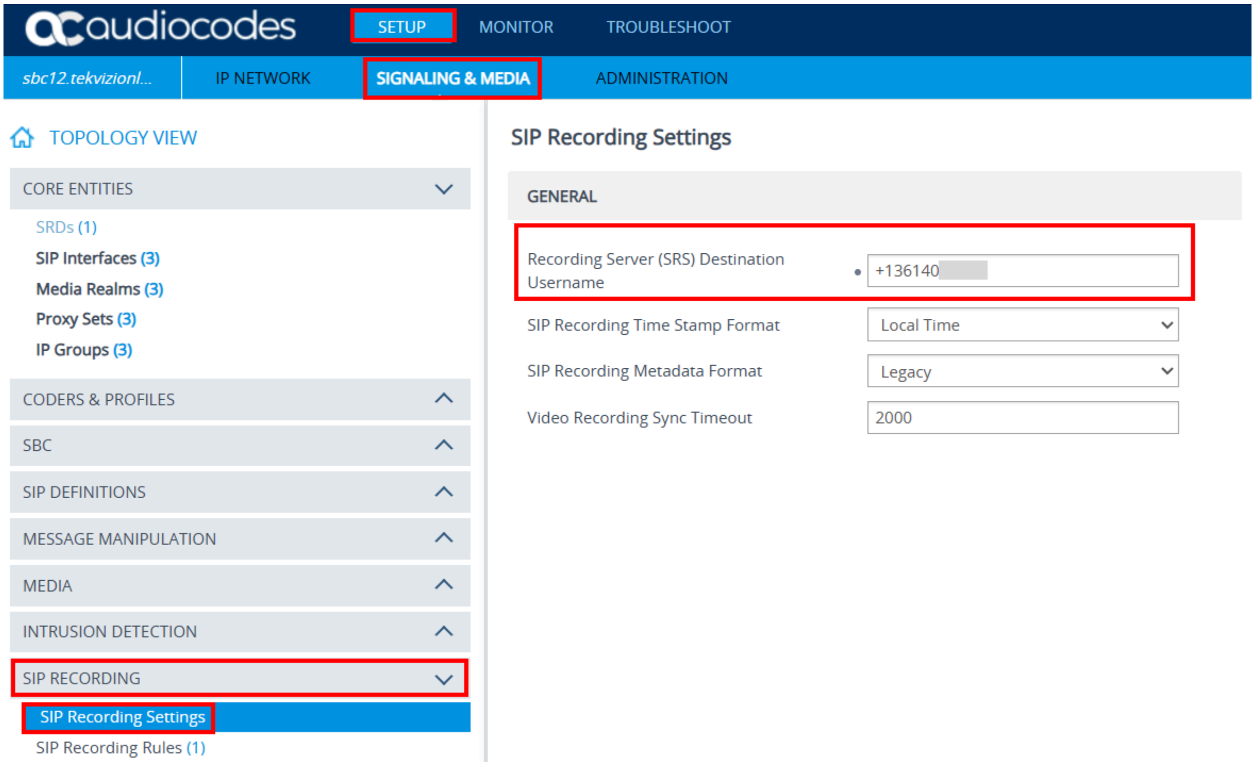


Figure 65: SIP Recording Settings

- Navigate to **SETUP** menu □ **SIGNALING & MEDIA** tab □ **SIP RECORDING** folder □ **SIP Recording Rules**

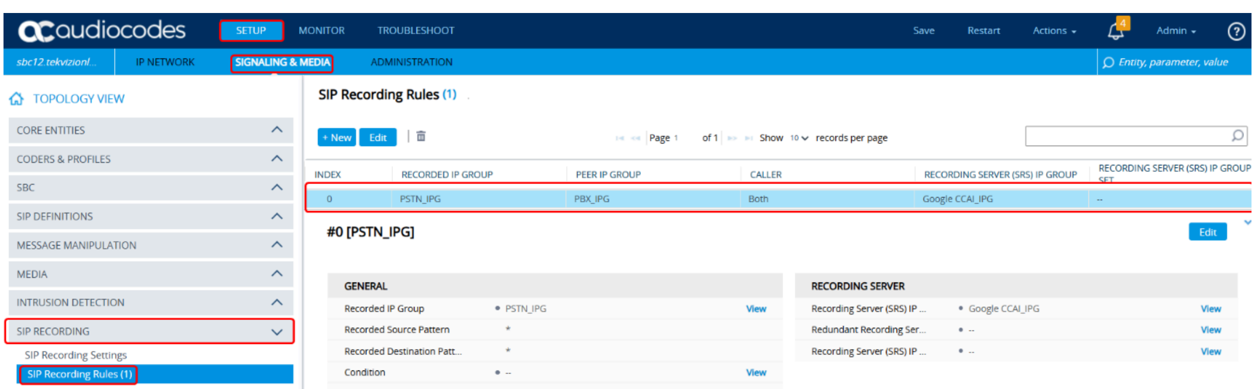


Figure 66: SIP Recording Rules

- Create SIP recording rules as shown below.

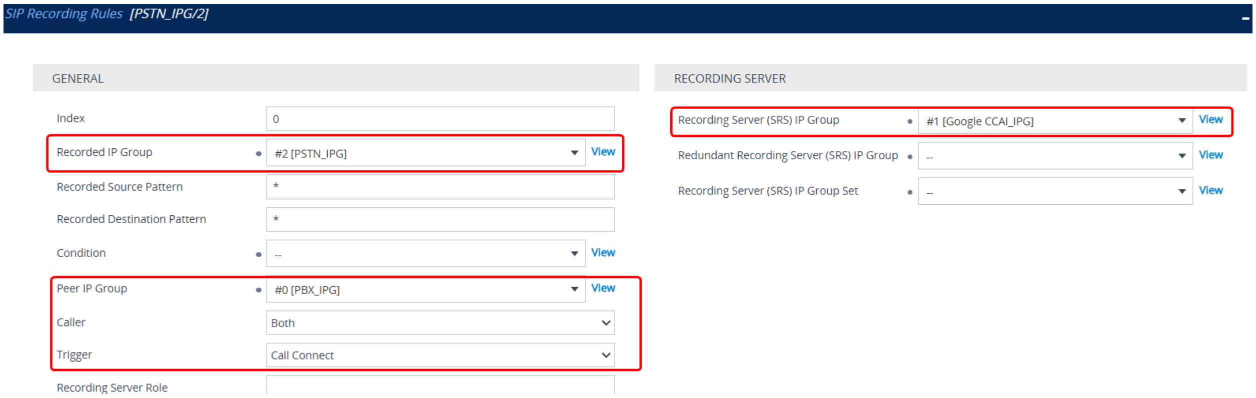


Figure 67: SIP Recording Rules (Cont.)

7.4.12 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.

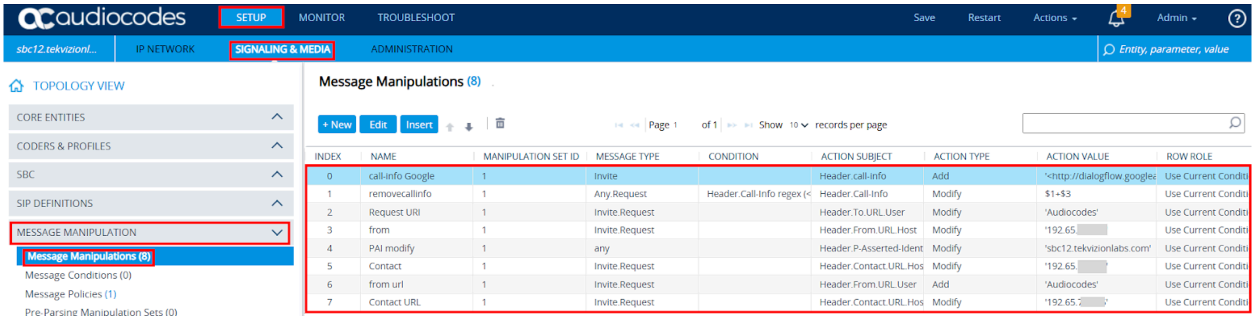


Figure 68: Message Manipulation towards Google CES

- 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.**

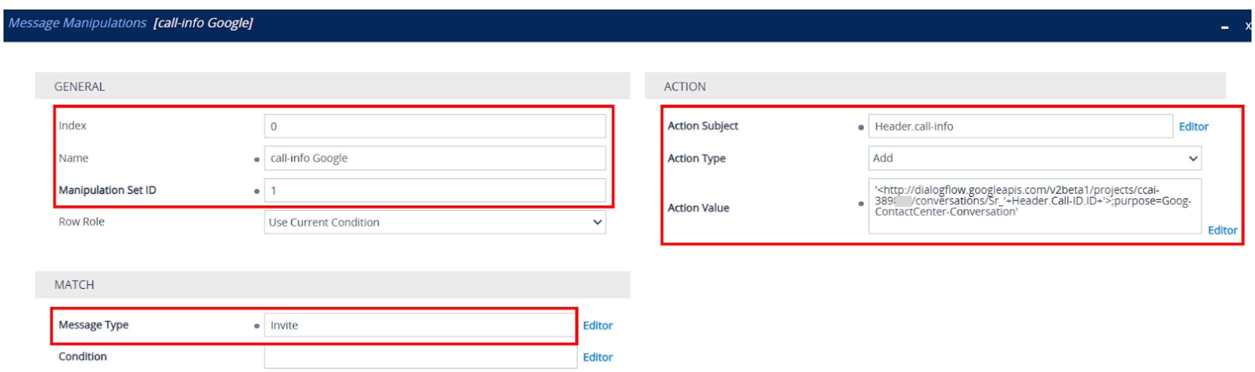


Figure 69: Message Manipulation: Call Info towards Google CES

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

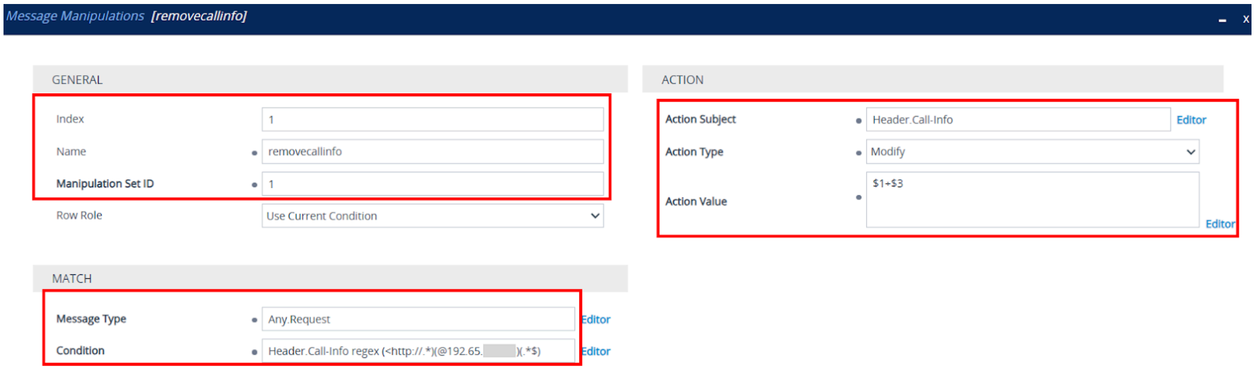


Figure 70: Message Manipulation: Call Info Modification towards Google CES

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

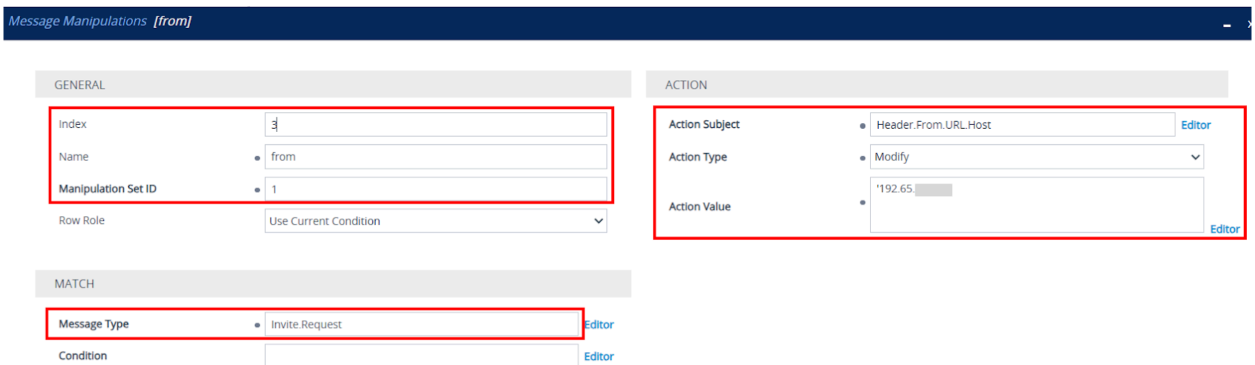


Figure 71: Message Manipulation: From Header host Part Modification towards Google CES

- Below header rule is created to change from header user part towards Google CES as 'AudioCodes'

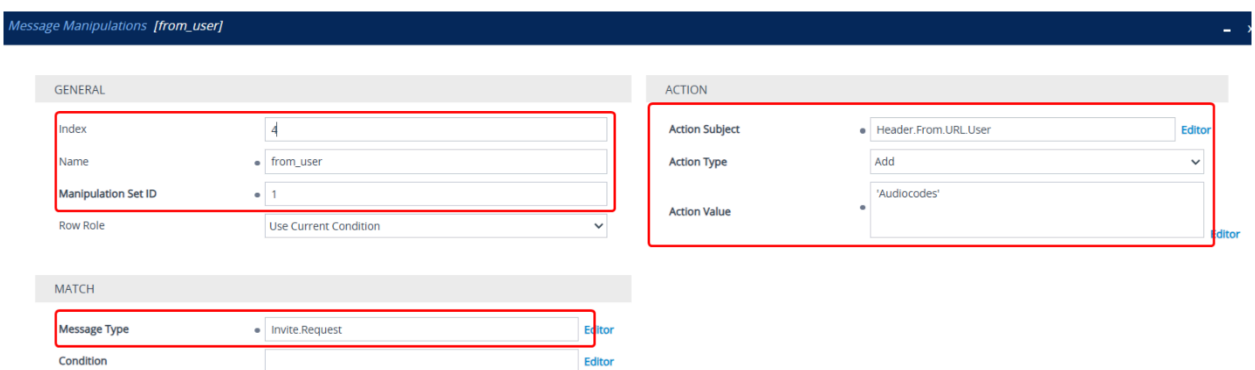


Figure 72: Message Manipulation: From Header User Part Modification towards Google CES

- Below header rule is created to change P-Asserted Identity host part towards Google CES as 'sbcX.Y.com'

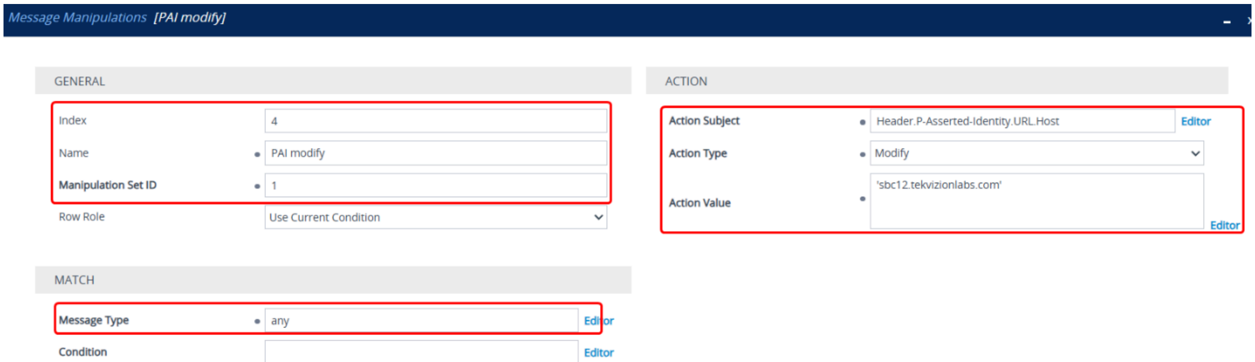


Figure 73: Message Manipulation: PAI Host Part Modification towards Google CES

- Below header rule is created to change To User part towards Google CES as 'AudioCodes'

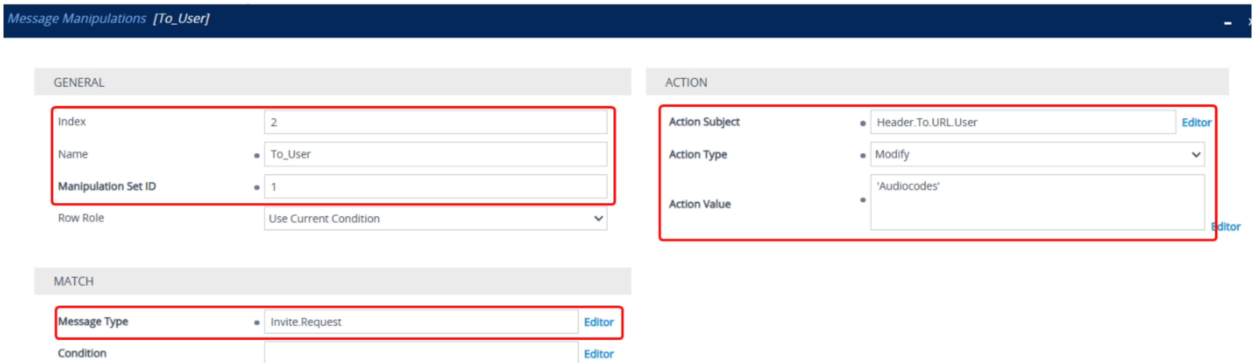


Figure 74: Message Manipulation: From User Part Modification towards Google CES

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

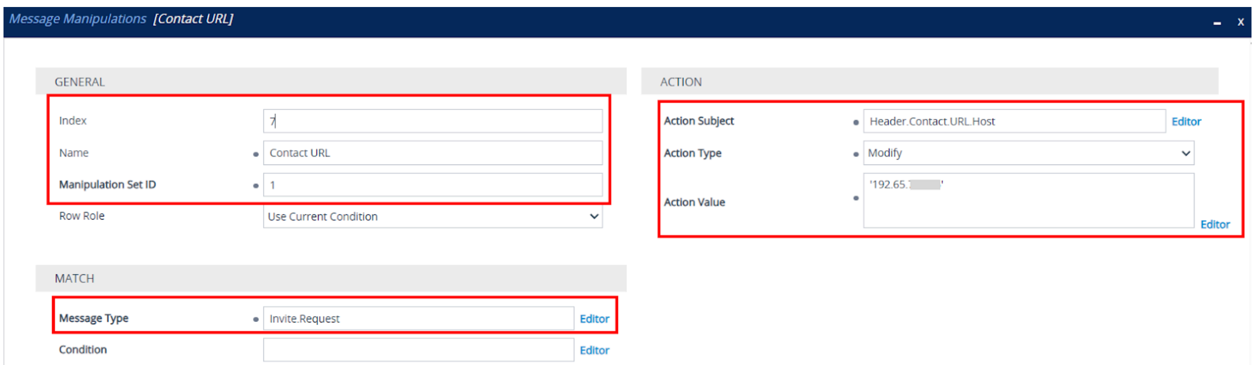


Figure 75: Message Manipulation: Contact User Part Modification towards Google CES

7.4.13 Configure Message Manipulation Rules (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 CES uses the participant labels provided in the Call-info header.
- <u>Call-info header with participant roles:</u>

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

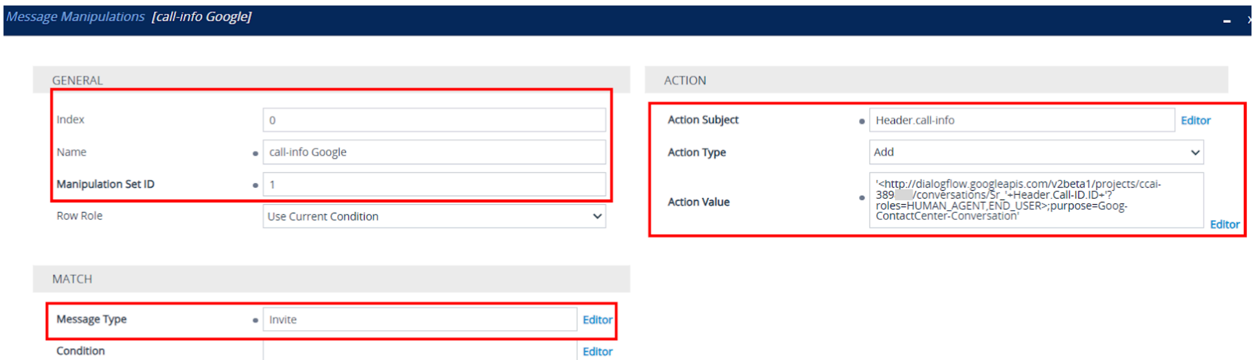


Figure 76: Message Manipulation: Call Info Modification (participation label) towards Google CES

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: 1452649026192025138 4@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:I1IjZjIMiUk11RE00FQyHkGoHuC6THFMk+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 77: 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: 185111343926820251303 2@192.65.
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:7JjlVVOjEOONIo9gRZztFnaLGIS gSqFlafy6751dZYMyq+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.

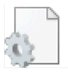
For SIPREC, there can be single media lines with *a=sendonly*, for SIP GTP there will be a multiple media line with *a=sendrecv*
Encrypted SRTP and the allocated port range should be used (Port: 16384-32767) else CES will not receive audio.

It must be a supported crypto suite by Google.

Figure 78: 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
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	TLS certificates have been verified, and a successful TLS connection has been established.
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	
3	SBC Configuration Verification	TCP link is persistent. Establish call, 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	every 900 secs the SBC should refresh the SIP session.	PASSED	Update message sent to Google CES every 900 secs.

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		Update or 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	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
8	SBC Configuration Verification	DTLS for Media Encryption. Configure the DTLS parameters for crypto negotiation for the BYOT trunk, certificate for DTLS must be self-signed by the SBC.	Validate the crypto is successfully negotiated and media is encrypted.	NOT SUPPORTED	
Inbound					
9	Inbound	SIP OPTIONS. SBC send SIP options every 60 seconds	Verify SBC sends SIP OPTIONS every 60 seconds and responded 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	Verify Call is established with audio and transcripts from both participants Verify call is disconnected properly	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		proper disconnect			
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	
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	UPDATE message is sent from SBC every 900 seconds without SDP
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	No audio was recorded during call hold is activated and when hold made inactive and recording continues.

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
16	Inbound	Inbound call hold scenarios. call starts out as active for both participants and making the Deactivate/Activate Conversation via grpc api.	Validate if media should not be present when activated and conversation starts to happen after deactivation. Confirm Signaling events and validate media and transcripts	PASSED	Recording is not present when deactivate conversation is started and recording resumed after activate conversation is initiated.
17	Inbound	Update. Validate that update sent prior to call establishment do not contain SDP	Validate that update prior to call establishment do not contain SDP as expected	PASSED	UPDATE message is sent from SBC every 900 seconds without SDP
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	UPDATE with SDP is not supported.
19	Inbound	re-invites. Ensure re-invites that modify session include SDP	Ensure re-invites that modify session include SDP	PASSED	UPDATE message is sent to Google CES as part of hold and resume 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	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
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	
23	Inbound	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	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
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 sdps properly, once call is move to hold active validate media and transcripts	PASSED	
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	UPDATE message is sent from AudioCodes SBC to Google CES every 15min (900 seconds)
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 conducted on 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_XXXX816?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 7 out of 10 attempts.
32	Inbound	DTLS test		NOT SUPPORTED	
33	Inbound	Conference TEST	Determine requirements, record all legs	PASSED	
34	Inbound	Validate Call recording	Verify call recording is recorded throughout the call	PASSED	