

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



## Table of Contents

1	Audience	3
1.1	Introduction.....	3
1.1.1	TekVizionLabs	3
2	SIP Trunking Network Components	4
3	Hardware Components	5
4	Software Requirements	5
5	Certified AudioCodes VE SBC Version	5
6	Features	5
6.1	Features Tested for Google CES Call Recording.....	5
6.2	Features Not Tested for Google CES Call Recording.....	5
6.3	Caveats and Limitations.....	5
6.4	Failed Test Case.....	5
7	Configuration	6
7.1	Configuration Checklist.....	6
7.2	IP Address Worksheet.....	7
7.3	Google CES API Configuration.....	8
7.4	AudioCodes VE SBC Configuration.....	9
7.4.1	Network Interface IP	9
7.4.2	Configure TLS Context for Google CES	10
7.4.3	Configure Media Realms	14
7.4.4	Configure SIP Signaling Interfaces	15
7.4.5	Configure Proxy Sets and Proxy Address	17
7.4.6	Configure Coders	21
7.4.7	Configure IP Profiles	23
7.4.8	Configure IP Groups	29
7.4.9	Configure Media Security	33
7.4.10	Configure IP to IP Call Routing	33
7.4.11	Configure SIP Recording	34
7.4.12	Configure Message Manipulation Rules	36
7.4.13	Configure Message Manipulation Rules (Participation Label)	39
8	SIP INVITE To GOOGLE CES	40
8.1	SIP INVITE for SIPREC call	40
8.2	SIP INVITE for GTP call	40
9	AudioCodes VE SBC Running configuration	41
10	Summary of Tests and Results	42

# 1 Audience

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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.

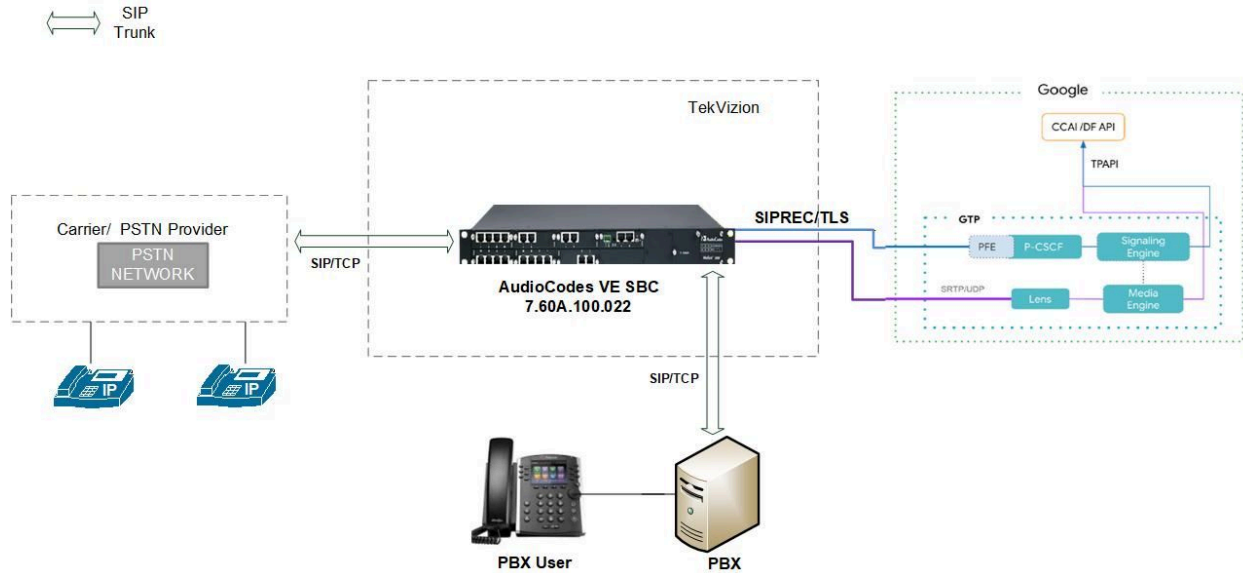
TekVizion is a systems integrator specifically dedicated to the telecommunications industry. Our core services include consulting/solution design, interoperability/Verification testing, integration, custom software development and solution support services. Our services help service providers achieve 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

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- AudioCodes VE SBC running on ESXi host version 6.7.0

### 4 Software Requirements

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- AudioCodes VE SBC Software version: 7.60A.100.022

### 5 Certified AudioCodes VE SBC Version

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**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

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#### 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
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#### 6.4 Failed Test Case

- None

## 7 Configuration

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### 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	<a href="#">Section 7.4.1</a>
Step 2	Configure TLS Context for Google CES	<a href="#">Section 7.4.2</a>
Step 3	Configure Media Realms	<a href="#">Section 7.4.3</a>
Step 4	Configure SIP Signaling Interfaces	<a href="#">Section 7.4.4</a>
Step 5	Configure Proxy Sets and Proxy Address	<a href="#">Section 7.4.5</a>
Step 6	Configure Coders	<a href="#">Section 7.4.6</a>
Step 7	Configure IP Profiles	<a href="#">Section 7.4.7</a>
Step 8	Configure IP Groups	<a href="#">Section 7.4.8</a>
Step 9	Configure Media Security	<a href="#">Section 7.4.9</a>
Step 10	Configure IP to IP Call Routing	<a href="#">Section 7.4.10</a>
Step 11	Configure SIP Recording	<a href="#">Section 7.4.11</a>
Step 12	Configure Message Manipulation Rules	<a href="#">Section 7.4.12</a>
Step 13	Configure Message Manipulation Rules (Participation label)	<a href="#">Section 7.4.13</a>

## 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**

<b>Component</b>	<b>IP Address</b>
<b>Google CES</b>	
Signaling	us.telephony.goog:5672
Media	74.125.X.X
<b>OnPrem PBX</b>	
LAN IP Address	10.80.X.X
<b>AudioCodes VE SBC</b>	
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.



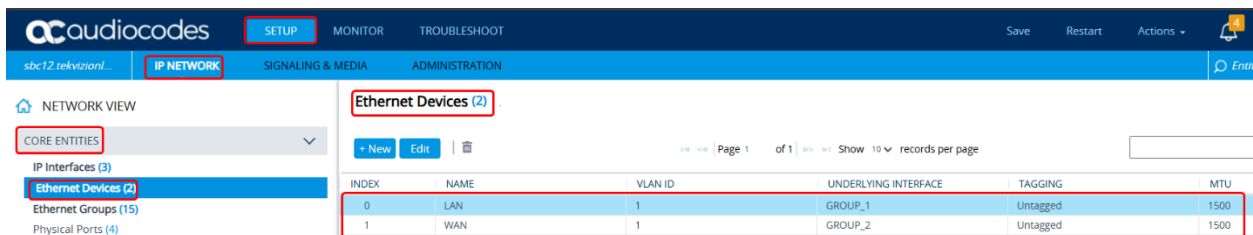
The screenshot shows the 'IP Interfaces (3)' configuration page. The left sidebar has 'CORE ENTITIES' expanded to 'IP Interfaces (3)'. The main table lists three interfaces:

INDEX	NAME	APPLICATION TYPE	INTERFACE MODE	IP ADDRESS	PREFIX LENGTH	DEFAULT GATEWAY	PRIMARY DNS	SECONDARY DNS	ETHERNET DEVICE
0	PBX-MGMT	OAMP + Media + Cont	IPv4 Manual	10.80.11.49	24	10.80.11.1	10.85.0.12	0.0.0.0	vlan 1
1	PSTN	Media + Control	IPv4 Manual	10.80.11.238	24	10.80.11.1	10.85.0.12	0.0.0.0	vlan 1
2	Google CCAI	Media + Control	IPv4 Manual	192.65	27	192.65	8.8.8.8	0.0.0.0	vlan2

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.



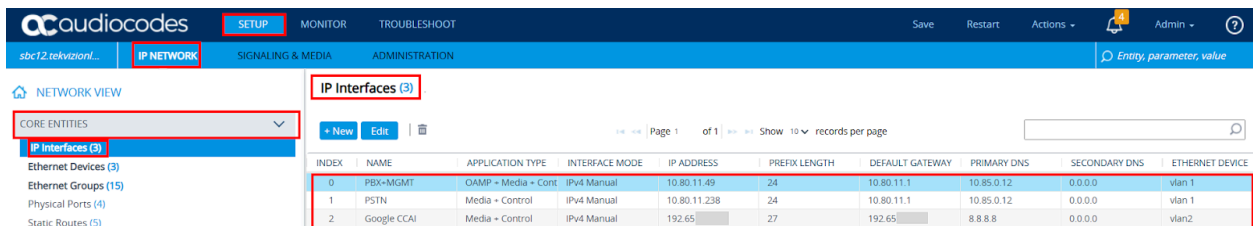
The screenshot shows the 'Ethernet Devices (2)' configuration page. The left sidebar has 'CORE ENTITIES' expanded to 'Ethernet Devices (2)'. The main table lists two devices:

INDEX	NAME	VLAN ID	UNDERLYING INTERFACE	TAGGING	MTU
0	LAN	1	GROUP_1	Untagged	1500
1	WAN	1	GROUP_2	Untagged	1500

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.



The screenshot shows the 'IP Interfaces (3)' configuration page, identical to Figure 2. The left sidebar has 'CORE ENTITIES' expanded to 'IP Interfaces (3)'. The main table lists three interfaces:

INDEX	NAME	APPLICATION TYPE	INTERFACE MODE	IP ADDRESS	PREFIX LENGTH	DEFAULT GATEWAY	PRIMARY DNS	SECONDARY DNS	ETHERNET DEVICE
0	PBX-MGMT	OAMP + Media + Cont	IPv4 Manual	10.80.11.49	24	10.80.11.1	10.85.0.12	0.0.0.0	vlan 1
1	PSTN	Media + Control	IPv4 Manual	10.80.11.238	24	10.80.11.1	10.85.0.12	0.0.0.0	vlan 1
2	Google CCAI	Media + Control	IPv4 Manual	192.65	27	192.65	8.8.8.8	0.0.0.0	vlan2

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

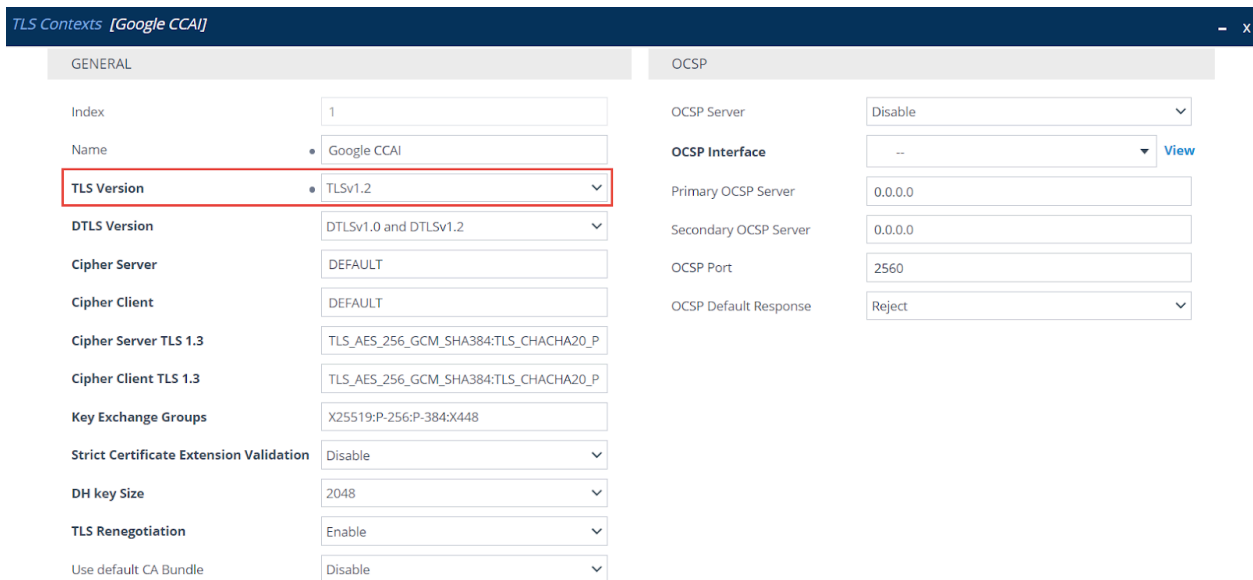
- Navigate to **SETUP** menu  **IP NETWORK** tab  **SECURITY** folder  **TLS Contexts**.
- Configure TLS context for Google CES as shown below.



The screenshot shows the AudioCodes Mediant VE SBC configuration interface. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The left sidebar shows 'CORE ENTITIES' with 'SECURITY' expanded to 'TLS Contexts (2)'. The main content area displays a table of TLS Contexts with the following data:

INDEX	NAME	TLS VERSION	DTLS VERSION	CIPHER SERVER
0	Default	TLSv1.0 TLSv1.1 and TLSv1.2	DTLSv1.0 and DTLSv1.2	DEFAULT
1	Google CCAI	TLSv1.2	DTLSv1.0 and DTLSv1.2	DEFAULT

Figure 5: TLS Context for Google CES



The screenshot shows the configuration details for the 'Google CCAI' TLS Context. The 'GENERAL' tab is active, and the 'TLS Version' field is highlighted with a red box. The configuration includes the following fields:

Field	Value
Index	1
Name	Google CCAI
TLS Version	TLSv1.2
DTLS Version	DTLSv1.0 and DTLSv1.2
Cipher Server	DEFAULT
Cipher Client	DEFAULT
Cipher Server TLS 1.3	TLS_AES_256_GCM_SHA384:TLS_CHACHA20_P
Cipher Client TLS 1.3	TLS_AES_256_GCM_SHA384:TLS_CHACHA20_P
Key Exchange Groups	X25519:P-256:P-384:X448
Strict Certificate Extension Validation	Disable
DH key Size	2048
TLS Renegotiation	Enable
Use default CA Bundle	Disable

The 'OCSP' tab is also visible, showing the following configuration:

Field	Value
OCSP Server	Disable
OCSP Interface	--
Primary OCSP Server	0.0.0.0
Secondary OCSP Server	0.0.0.0
OCSP Port	2560
OCSP Default Response	Reject

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.

The screenshot shows the Audiocodes management console interface. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The left sidebar shows a tree view with 'SECURITY' expanded to 'TLS Contexts (2)'. The main content area displays a table of TLS contexts. The first row is selected, and the 'Change Certificate' link is highlighted. Below the table, the configuration details for the selected context are shown.

INDEX	NAME	TLS VERSION	DTLS VERSION	CIPHER SERVER
0	default	TLSv1.2 and TLSv1.3	DTLSv1.0 and DTLSv1.2	DEFAULT
1	Google	TLSv1.2 and TLSv1.3	DTLSv1.0 and DTLSv1.2	DEFAULT

**#1 [Google]**

**GENERAL**

- Name: Google
- TLS Version: TLSv1.2 and TLSv1.3
- DTLS Version: DTLSv1.0 and DTLSv1.2
- Cipher Server: DEFAULT
- Cipher Client: DEFAULT
- Cipher Server TLS 1.3: TLS\_AES\_256\_GCM\_SHA384:TLS\_CHACHA20\_POLY1305\_SHA256:TLS\_AES\_1...
- Cipher Client TLS 1.3: TLS\_AES\_256\_GCM\_SHA384:TLS\_CHACHA20\_POLY1305\_SHA256:TLS\_AES\_1...
- Key Exchange Groups: X25519:P-256:P-384:X448
- Strict Certificate Extension Vall...: Disable
- DH key Size: 2048
- TLS Renegotiation: Enable
- Use default CA Bundle: Disable
- Security Level: 1

**OCSF**

- OCSF Server: Disable
- OCSF Interface: --
- Primary OCSF Server: --
- Secondary OCSF Server: --
- OCSF Port: 2560
- OCSF Default Response: Reject

Change Certificate >>

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.

The screenshot shows the 'Change Certificates' form for the Google CES TLS context. The form is titled 'CERTIFICATE SIGNING REQUEST / GENERATE SELF-SIGNED CERTIFICATE REQUEST'. It contains several input fields for certificate details and a section for generating a private key.

**CERTIFICATE SIGNING REQUEST / GENERATE SELF-SIGNED CERTIFICATE REQUEST**

- Common Name [CN]: sbc12.tekvisionlabs.com
- Organizational Unit [OU]:
- Company name [O]:
- Locality or city name [L]:
- State [ST]:
- Country code [C]:
- 1st Subject Alternative Name [SAN]: EMAIL
- 2nd Subject Alternative Name [SAN]: EMAIL
- 3rd Subject Alternative Name [SAN]: EMAIL
- 4th Subject Alternative Name [SAN]: EMAIL

**GENERATE NEW PRIVATE KEY**

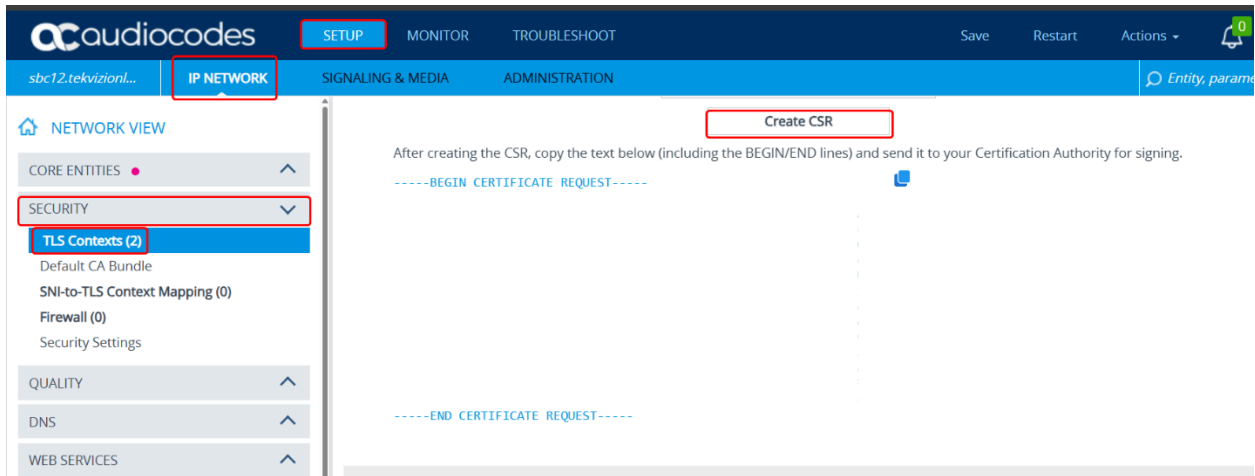
- Private Key Format: RSA
- Private Key Size: 2048

Important: generation of private key is a lengthy operation during which the device service may be affected.

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.

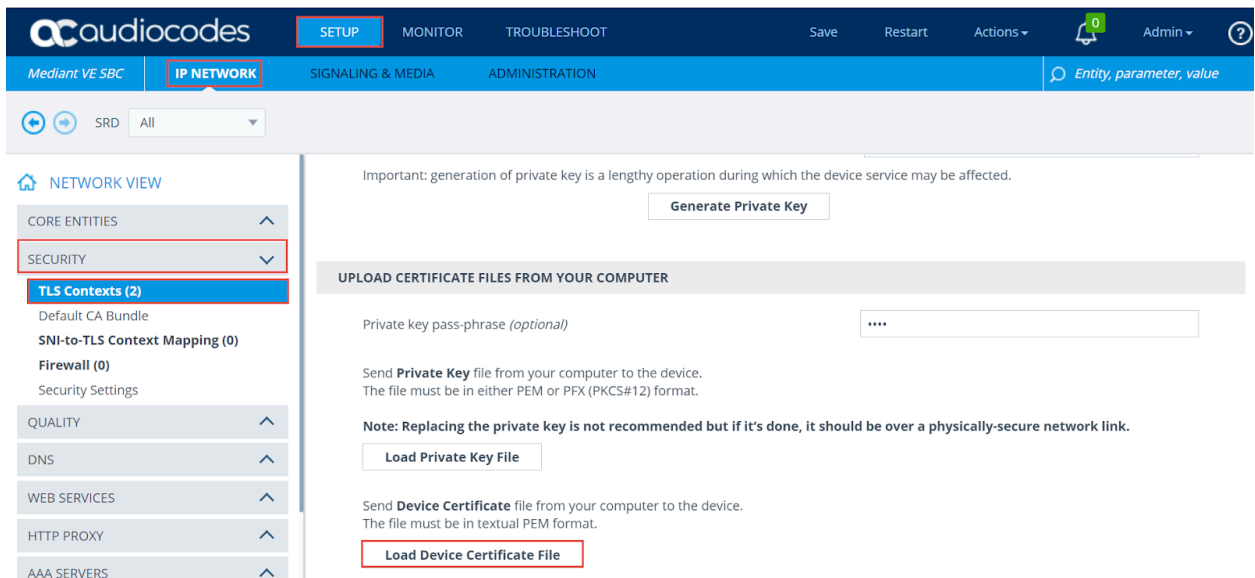


**Figure 9: CSR Generation for Google CES TLS Context (Cont.)**

### 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.

The screenshot shows the Audiocodes management console. The navigation menu includes **SETUP**, **MONITOR**, and **TROUBLESHOOT**. The main menu has **IP NETWORK**, **SIGNALING & MEDIA**, and **ADMINISTRATION**. The left sidebar shows **SECURITY** > **TLS Contexts (2)**. The main content area displays the configuration for a context named "Google CCAI".

GENERAL		OCSP	
Name	Google CCAI	OCSP Server	Disable
TLS Version	TLSv1.2	OCSP Interface	-- <a href="#">View</a>
DTLS Version	DTLSv1.0 and DTLSv1.2	Primary OCSP Server	0.0.0.0
Cipher Server	DEFAULT	Secondary OCSP Ser...	0.0.0.0
Cipher Client	DEFAULT	OCSP Port	2560
Cipher Server TLS 1.3	TLS_AES_256_GCM_SHA384:TLS_CHACHA20_POLY1...	OCSP Default Respo...	Reject
Cipher Client TLS 1.3	TLS_AES_256_GCM_SHA384:TLS_CHACHA20_POLY1...		
Key Exchange Groups	X25519:P-256:P-384:X448		
Strict Certificate Ex...	Disable		
DH key Size	2048		
TLS Renegotiation	Enable		
Use default CA Bundle	Disable		

At the bottom of the configuration page, there are three links: [Certificate Information >>](#), [Change Certificate >>](#), and [Trusted Root Certificates >>](#).

Figure 11: Trusted Root Certificates

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

The screenshot shows the Audiocodes management console with the **Trusted Root Certificates** page selected. The page displays a table of certificates with the following data:

INDEX	SUBJECT	ISSUER	EXPIRES
12	Go Daddy Root Certificate Autho	The Go Daddy Group, Inc.	Fri, 30 May 2031 01:30:00 GMT
13	Go Daddy Secure Certificate Aut	Go Daddy Root Certificate Autho	Sat, 03 May 2031 01:30:00 GMT

At the top right of the table, there are buttons for **Import**, **Export**, and **Remove**. The **Import** button is highlighted with a red box.

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.

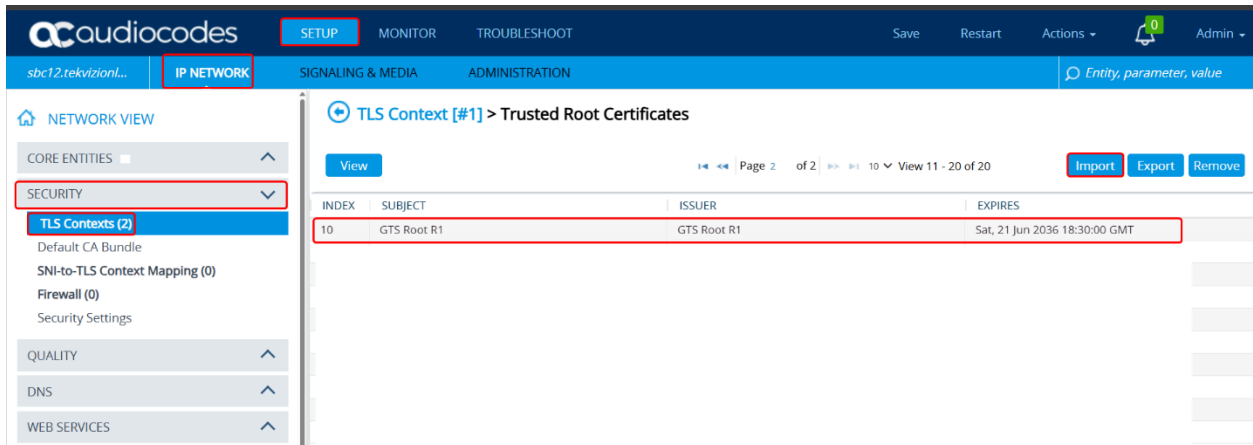


Figure : GTS Root R1 Certificate

#### 7.4.3 Configure Media Realms

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

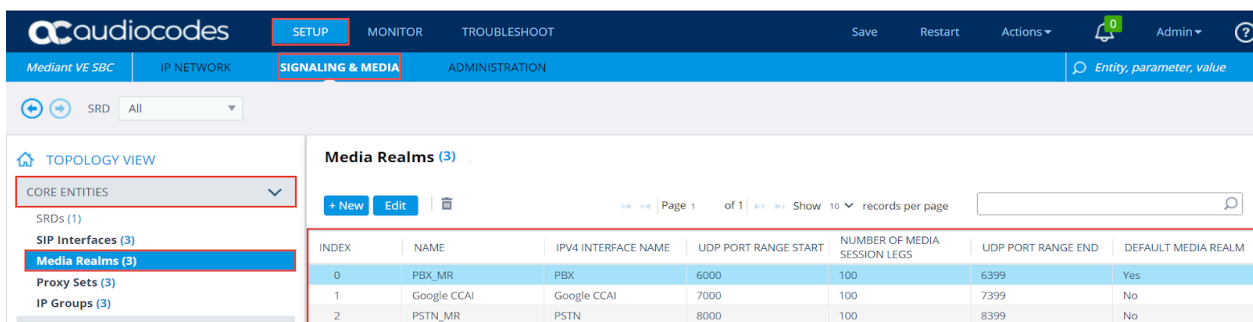


Figure : Configure Media Realms

## 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:

INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULATING PROTOCOL	MEDIA REALM
0	PBX	DefaultSRD (#	PBX	SBC	5060	5060	0	No encapsulation	PBX_MR
1	Google CCAI	DefaultSRD (#	Google CCAI	SBC	0	0	5061	No encapsulation	PSTN_MR
2	PSTN	DefaultSRD (#	PSTN	SBC	5060	5060	0	No encapsulation	PSTN_MR

Figure : SIP Signaling Interfaces

SRD: #0 [DefaultSRD]

**GENERAL**

Index: 0

Name: PBX

Topology Location: Up

Network Interface: #0 [PBX+MGMT]

Application Type: SBC

UDP Port: 5060

TCP Port: 5060

**MEDIA**

Media Realm: #0 [PBX\_MR]

Direct Media: Disable

MSRP TCP Port: 0

MSRP TLS Port: 0

**SECURITY**

TLS Context Name: --

Figure : SIP Signaling Interfaces for OnPrem PBX

**CLASSIFICATION**

Classification Failure Response Type: 500

Pre-classification Manipulation Set ID: -1

**SECURITY**

TLS Mutual Authentication: --

Message Policy: --

User Security Mode: Not Configured

Enable Un-Authenticated Registrations: Not configured

Max. Number of Registered Users: -1

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

## Google CES:

SIP Interfaces [Google CCAI]

Index	1	Media Realm	#1 [Google CCAI] <a href="#">View</a>
Name	Google CCAI	Direct Media	Disable
Topology Location	Up	MSRP TCP Port	0
Network Interface	#2 [Google CCAI] <a href="#">View</a>	MSRP TLS Port	0
Application Type	SBC	<b>SECURITY</b>	
UDP Port	0	TLS Context Name	#1 [Google] <a href="#">View</a>
TCP Port	5060	TLS Mutual Authentication	Enable
TLS Port	5061	Message Policy	-- <a href="#">View</a>
SCTP Port	0	User Security Mode	Not Configured
SCTP Secondary Network Interface	-- <a href="#">View</a>	Enable Un-Authenticated Registrations	Not configured
Additional UDP Ports		Max. Number of Registered Users	-1
Additional UDP Ports Mode	Always Open		
Encapsulating Protocol	No encapsulation		
Enable TCP Keepalive	Enable		

Figure : SIP Signaling Interfaces for Google CES

SIP Interfaces [Google CCAI]

SCTP Secondary Network Interface	-- <a href="#">View</a>	User Security Mode	Not Configured
Additional UDP Ports		Enable Un-Authenticated Registrations	Not configured
Additional UDP Ports Mode	Always Open	Max. Number of Registered Users	-1
Encapsulating Protocol	No encapsulation		
Enable TCP Keepalive	Enable		
Used By Routing Server	Not Used		
Pre-Parsing Manipulation Set	-- <a href="#">View</a>		
CAC Profile	-- <a href="#">View</a>		
<b>CLASSIFICATION</b>			
Classification Failure Response Type	500		
Pre-classification Manipulation Set ID	-1		
Call Setup Rules Set ID	-1		
Classify By Registration DB	Enable		

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



## PSTN Gateway:

The screenshot shows the configuration page for SIP Interfaces for a PSTN gateway. The page is divided into two main sections: GENERAL and MEDIA. The GENERAL section includes fields for Index (2), Name (PSTN), Topology Location (Up), Network Interface (#1 [PSTN]), Application Type (SBC), UDP Port (5060), TCP Port (5060), TLS Port (0), SCTP Port (0), SCTP Secondary Network Interface (--), Additional UDP Ports, and Additional UDP Ports Mode (Always Open). The MEDIA section includes Media Realm (#2 [PSTN\_MR]), Direct Media (Disable), MSRP TCP Port (0), and MSRP TLS Port (0). The SECURITY section includes TLS Context Name (--), TLS Mutual Authentication, Message Policy (--), User Security Mode (Not Configured), and Enable Un-Authenticated Registrations (Not configured).

Figure : SIP Signaling Interfaces for PSTN Gateway

The screenshot shows the continuation of the configuration page for SIP Interfaces for a PSTN gateway. The bottom of the GENERAL section includes SCTP Secondary Network Interface (--), Additional UDP Ports, Additional UDP Ports Mode (Always Open), Encapsulating Protocol (No encapsulation), Enable TCP Keepalive (Disable), Used By Routing Server (Not Used), Pre-Parsing Manipulation Set (--), and CAC Profile (--). The CLASSIFICATION section includes Classification Failure Response Type (500), Pre-classification Manipulation Set ID (-1), Call Setup Rules Set ID (-1), and Classify By Registration DB (Enable).

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.

INDEX	NAME	SRD	SBC IPv4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP MODE
0	PBX_PS	DefaultSRD (#0)	PBX	60		Disable
1	Google CCAI SIPREC	DefaultSRD (#0)	Google CCAI	60		Disable
2	PSTN_PS	DefaultSRD (#0)	PSTN	60		Disable

**Figure : Configurations of Proxy Sets**

### OnPrem PBX:

**KEEP ALIVE**

- Index: 0
- Name: PBX\_PS
- SBC IPv4 SIP Interface: #0 [PBX]
- Proxy Keep-Alive: Using OPTIONS
- Proxy Keep-Alive Time [sec]: 60
- Keep-Alive Failure Responses: [ ]
- Success Detection Retries: 1
- Success Detection Interval: 10
- Failure Detection Retransmissions: -1

**ADVANCED**

- Redundancy Mode: [ ]
- Proxy Hot Swap Mode: Disable
- Proxy Load Balancing Method: Disable
- Min. Active Servers for Load Balancing: 1
- Classification Input: IP Address only
- DNS Resolve Method: [ ]
- Accept DHCP Proxy List: Disable
- TCP/TLS Connection Reuse: Use Global Setting
- TLS Remote Subject Name: [ ]
- Peer Host Name Verification Mode: Use Global Settings
- In-Call Route Mode: Disable
- Reliable Connection Failure Retry: Disable

**Figure : Proxy Set Configuration of OnPrem PBX**

## Google CES:

Proxy Sets [Google CCAI SIPREC]

GENERAL		REDUNDANCY	
Index	1	Redundancy Mode	Homing
Name	Google CCAI SIPREC	Proxy Hot Swap Mode	Enable
SBC IPv4 SIP Interface	#1 [Google CCAI] <a href="#">View</a>	Proxy Load Balancing Method	Random Weights
TLS Context Name	#1 [Google]	Min. Active Servers for Load Balancing	1
KEEP ALIVE		ADVANCED	
Proxy Keep-Alive	Using OPTIONS	Classification Input	IP Address only
Proxy Keep-Alive Time [sec]	60	DNS Resolve Method	SRV
Keep-Alive Failure Responses		Accept DHCP Proxy List	Disable
Success Detection Retries	1	TCP/TLS Connection Reuse	Use Global Setting
Success Detection Interval	10	TLS Remote Subject Name	
Failure Detection Retransmissions	-1	Peer Host Name Verification Mode	Use Global Settings
		In-Call Route Mode	Disable

Figure : Proxy Set Configuration of Google CES

## PSTN Gateway:

Proxy Sets [PSTN\_PS]

GENERAL		REDUNDANCY	
Index	2	Redundancy Mode	
Name	PSTN_PS	Proxy Hot Swap Mode	Disable
SBC IPv4 SIP Interface	#2 [PSTN] <a href="#">View</a>	Proxy Load Balancing Method	Disable
TLS Context Name	--	Min. Active Servers for Load Balancing	1
KEEP ALIVE		ADVANCED	
Proxy Keep-Alive	Using OPTIONS	Classification Input	IP Address only
Proxy Keep-Alive Time [sec]	60	DNS Resolve Method	
Keep-Alive Failure Responses		Accept DHCP Proxy List	Disable
Success Detection Retries	1	TCP/TLS Connection Reuse	Use Global Setting
Success Detection Interval	10	TLS Remote Subject Name	
Failure Detection Retransmissions	-1	Peer Host Name Verification Mode	Use Global Settings
		In-Call Route Mode	Disable
		Reliable Connection Failure Retry	Disable

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**

The screenshot shows the Audiocodes management console interface. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The left sidebar shows a 'TOPOLOGY VIEW' with a tree structure where 'CORE ENTITIES' is expanded to show 'Proxy Sets (3)'. The main content area is divided into several sections: 'GENERAL' (SRD, Name, SBC IP4 SIP Interface, TLS Context Name), 'REDUNDANCY' (Redundancy Mode, Proxy Hot Swap Mode, Proxy Load Balancing Me..., Min. Active Servers for Lo...), 'KEEP ALIVE' (Proxy Keep-Alive, Proxy Keep-Alive Time [sec], Keep-Alive Failure Respon..., Success Detection Retries, Success Detection Interval, Failure Detection Retrans...), 'ADVANCED' (Classification Input, DNS Resolve Method, Accept DHCP Proxy List, TCP/TLS Connection Reuse, TLS Remote Subject Name, Peer Host Name Verificat..., In-Call Route Mode, Reliable Connection Failu...), and a 'PROXY ADDRESS' table with columns 'PROXY ADDRESS' and 'TYPE'. The table contains one entry: '10.64. .:5060' with type 'TCP'. A 'Proxy Address 1 items >>' link is visible at the bottom.

Figure 26: Proxy Address Configuration

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

The screenshot shows the Audiocodes management console interface, similar to Figure 26, but with a modal window open for adding a new proxy address. The modal title is 'Proxy Address [10.80.11.246:5060]'. The 'GENERAL' section of the modal contains the following fields: 'index' (0), 'Proxy Address' (10.80.11.246:5060), 'Transport Type' (TCP), 'Proxy Priority' (0), and 'Proxy Random Weight' (0). The 'Proxy Address' and 'Transport Type' fields are highlighted with a red box.

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

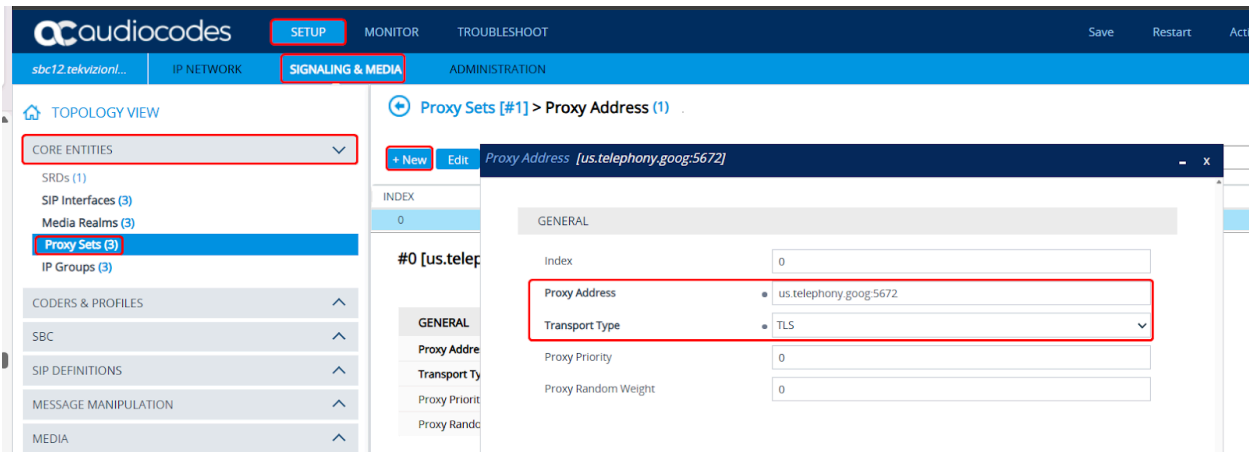


Figure : Proxy Address Configuration of Google CES

- 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

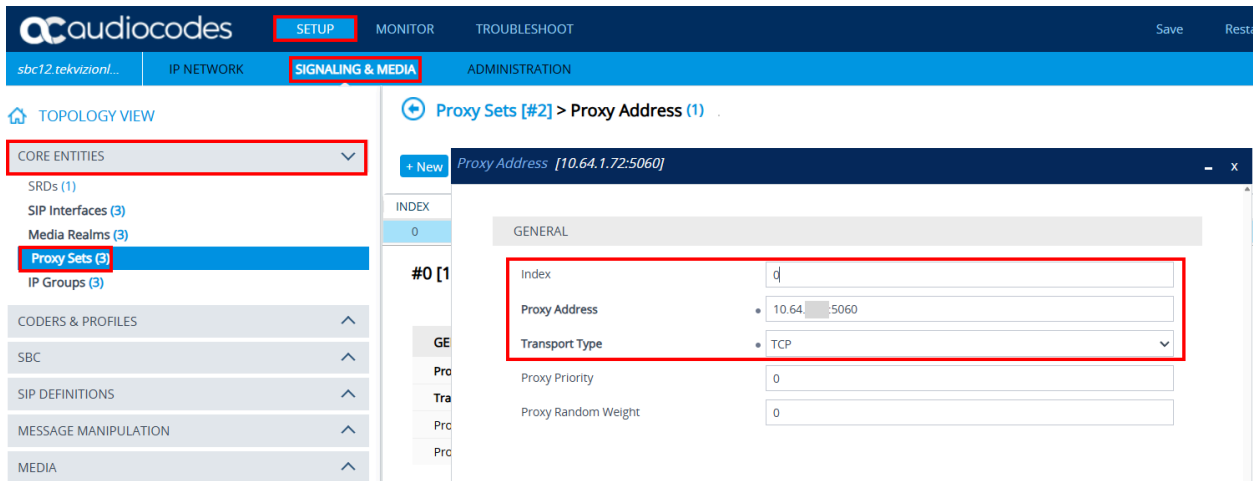


Figure 29: Proxy Address Configuration of PSTN Gateway

## 7.4.6 Configure Coders

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

The screenshot shows the Audiocodes management console. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The left sidebar has 'CORE ENTITIES', 'CODERS & PROFILES', 'IP Profiles (3)', 'Coder Settings', 'Coders Groups (1)', 'Allowed Audio Coders Groups (1)', 'Allowed Video Coders Groups (0)', 'SBC', 'SIP DEFINITIONS', 'MESSAGE MANIPULATION', and 'MEDIA'. The main content area is titled 'Coders Groups (1)'. It features a table with the following data:

INDEX	NAME
0	AudioCodersGroups_0

Below the table, there is a section for '#0[AudioCodersGroups\_0]' with an 'Edit' button. Under the 'GENERAL' tab, the 'Name' field is set to 'AudioCodersGroups\_0'. A link 'Coders Table 4 items >>' is visible at the bottom of the configuration area.

Figure 30: Coders Configurations

The screenshot shows the Audiocodes management console. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The left sidebar has 'CORE ENTITIES', 'CODERS & PROFILES', 'IP Profiles (3)', 'Coder Settings', 'Coders Groups (1)', 'Allowed Audio Coders Groups (1)', and 'Allowed Video Coders Groups (0)'. The main content area is titled 'Coders Groups [#0] > Coders Table (2)'. It features a table with the following data:

INDEX	NAME	PTIME	RATE	PAYLOAD TYPE	SILENCE SUPPRESSION	CODER SPECIFIC
0	G.711U-law	20	64	0	Disable	
1	G.711A-law	20	64	8	Disable	

Below the table, there is a section for '#0 [G.711U-law]' with an 'Edit' button.

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**.

The screenshot displays the Audiocodes management interface. The top navigation bar includes 'audiocodes', 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The main menu shows 'Mediant VE SBC', 'IP NETWORK', 'SIGNALING & MEDIA', and 'ADMINISTRATION'. The left sidebar is titled 'TOPOLOGY VIEW' and contains several categories: 'CORE ENTITIES', 'CODERS & PROFILES', 'IP Profiles (3)', 'Coder Settings', 'Coders Groups (1)', 'Allowed Audio Coders Groups (1)', 'Allowed Video Coders Groups (0)', 'SBC', 'SIP DEFINITIONS', and 'MESSAGE MANIPULATION'. The 'Allowed Audio Coders Groups (1)' section is active, showing a table with one record:

INDEX	NAME
0	G711

Below the table, the configuration details for '#0[G711]' are shown, including a 'GENERAL' section with the 'Name' field set to 'G711'. A link at the bottom indicates 'Allowed Audio Coders 1 items >>'.

**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.

The screenshot displays the Audiocodes management interface. The top navigation bar includes 'audiocodes', 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The main menu shows 'Mediant VE SBC', 'IP NETWORK', 'SIGNALING & MEDIA', and 'ADMINISTRATION'. The left sidebar is titled 'TOPOLOGY VIEW' and contains several categories: 'CORE ENTITIES', 'CODERS & PROFILES', 'IP Profiles (3)', 'Coder Settings', 'Coders Groups (1)', 'Allowed Audio Coders Groups (1)', 'Allowed Video Coders Groups (0)', 'SBC', 'SIP DEFINITIONS', and 'MESSAGE MANIPULATION'. The 'Allowed Audio Coders Groups (1)' section is active, showing a table with one record:

INDEX	CODER	USER-DEFINED CODER
0	G.711 U-law	

Below the table, the configuration details for '#0' are shown, including a 'GENERAL' section with the 'Name' field set to 'G.711 U-law'.

**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:

The screenshot shows the Audiocodes management console. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. Below it, the 'SIGNALING & MEDIA' tab is selected, and the 'IP NETWORK' sub-tab is active. The left sidebar shows a tree view with 'CORE ENTITIES', 'CODERS & PROFILES', and 'IP Profiles (3)'. The main content area displays a table of IP Profiles:

INDEX	NAME
0	PBX_IP
1	PSTN_IP
2	Google CCAI_IP

Figure 34: IP Profile Configurations

The screenshot shows the configuration page for the IP Profile 'PBX\_IP'. The page is divided into several sections:

- GENERAL:** Index (0), Name (PBX\_IP), Created by Routing Server (No), Used By Routing Server (Not Used).
- MEDIA SECURITY:** SBC Media Security Mode (Not Secured), Symmetric MKI (Disable), MKI Size (0), SBC Enforce MKI Size (Don't enforce), SBC Media Security Method (SDES), Reset SRTP Upon Re-key (Disable).
- SBC SIGNALING:** PRACK Mode (Transparent), P-Asserted-Identity Header Mode (As Is), Diversion Header Mode (As Is), History-Info Header Mode (As Is), Session Expires Mode (Transparent), SIP UPDATE Support (Supported), Remote re-INVITE (Supported), Remote Delayed Offer Support (Supported), MSRP re-INVITE/UPDATE (Supported), MSRP Offer Setup Role (ActPass), MSRP Empty Message Format (Default), Remote Representation Mode (According to Operation Mode).

Figure 35: IP Profile Configurations of OnPrem PBX

The screenshot shows the continuation of the IP Profile configuration page for 'PBX\_IP'. The sections include:

- Remote Multiple Invites Mode:** Remote Early Media RTP Detection Mode (By Media), Remote RFC 3960 Support (Not Supported), Remote Can Play Ringback (Yes), Generate RTP (None).
- SBC MEDIA:** SDP Subsequent Responses Mode (Handle All), Mediation Mode (RTP Mediation), Extension Coders Group (---), Allowed Audio Coders (#0 [G711]), Allowed Coders Mode (Restriction).
- SBC FORWARD AND TRANSFER:** Remote REFER Mode (Regular), Remote Replaces Mode (Standard), Play RBT To Transferee (No), Remote 3xx Mode (Transparent), Send Header for Transfer (None).
- Other settings:** User Registration Time (0), NAT UDP Registration Time (-1), NAT TCP Registration Time (-1), UnRegister on WebSocket Disconnect (Enable).

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



IP Profiles [PBX\_IP]

Allowed Video Coders	...	View
Allowed Media Types		
Direct Media Tag		
RFC 2833 Mode	As Is	
RFC 2833 DTMF Payload Type	0	
Alternative DTMF Method	As Is	
Send Multiple DTMF Methods	Disable	
Receive Multiple DTMF Methods	Disable	
Adapt RFC2833 BW to Voice coder BW	Disabled	
SDP Prime Answer	Remote Answer	
Preferred PTIME	0	
Use Silence Suppression	Transparent	
RTP Redundancy Mode	As Is	
RTCP Mode	Transparent	

<b>SBC HOLD</b>	
Remote Hold Format	Transparent
Reliable Held Tone Source	Yes
Play Held Tone	No

<b>SBC FAX</b>	
Fax Coders Group	...
Fax Mode	As Is
Fax Offer Mode	All coders
Fax Answer Mode	Single coder
Remote Renegotiate on Fax Detection	Transparent
Fax Rerouting Mode	Disable

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

IP Profiles [Google CCAL\_IP]

Jitter Compensation	Disable
ICE Mode	Disable
SDP Handle RTCP	Don't Care
RTCP Mux	Not Supported
RTCP Feedback	Feedback Off
Re-number MID	Disable
Voice Quality Enhancement	Disable
Switch Coder Upon Voice Quality	Disable
Max Opus Bandwidth	0
Generate No-Op Packets	Disable
Enhanced PLC	Disable
SBC Multiple Coders	Not Supported
SBC Allow Only Negotiated PT	Disable
Add Media IP Change Header	Disable

<b>MEDIA</b>	
Broken Connection Mode	Disconnect
No RTP Mode	Disconnect
Media IP Version Preference	Only IPv4
RTP Redundancy Depth	Disable

<b>LOCAL TONES</b>	
Local Ringback Tone Index	-1
Local Held Tone Index	-1

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

IP Profiles [Google CCAL\_IP]

Remove CSRC	Disable
SBC Precondition	Not Supported
BFCP IP from Audio Media	According to Global Parameter
Remove EXTMAP	Disable

<b>QUALITY OF SERVICE</b>	
RTP IP DiffServ	46
RTP Video DiffServ	-1
Signaling DiffServ	24
Data DiffServ	0

<b>JITTER BUFFER</b>	
Dynamic Jitter Buffer Minimum Delay [msec]	10
Dynamic Jitter Buffer Optimization Factor	10

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

## PSTN Gateway:

IP Profiles [PSTN\_IP]

GENERAL		SBC SIGNALING	
Index	1	PRACK Mode	Transparent
Name	PSTN_IP	P-Asserted-Identity Header Mode	As Is
Created by Routing Server	No	Diversion Header Mode	As Is
Used By Routing Server	Not Used	History-Info Header Mode	As Is
<b>SBC Media Security Mode</b> Not Secured		Session Expires Mode	Transparent
Symmetric MKI	Disable	SIP UPDATE Support	Supported
MKI Size	0	Remote re-INVITE	Supported
SBC Enforce MKI Size	Don't enforce	Remote Delayed Offer Support	Supported
SBC Media Security Method	SDES	MSRP re-INVITE/UPDATE	Supported
		MSRP Offer Setup Role	ActPass
		MSRP Empty Message Format	Default

Figure 40: IP Profile Configurations of PSTN Gateway

IP Profiles [PSTN\_IP]

Remote Multiple Early Dialogs	According to Operation Mode	<b>SBC REGISTRATION</b>	
Remote Multiple Answers Mode	Disable	User Registration Time	0
<b>Remote Early Media RTP Detection Mode</b>	By Signaling	NAT UDP Registration Time	-1
Remote RFC 3960 Support	Not Supported	NAT TCP Registration Time	-1
Remote Can Play Ringback	Yes	UnRegister on WebSocket Disconnect	Enable
Generate RTP	None	<b>SBC FORWARD AND TRANSFER</b>	
<b>SBC MEDIA</b>		Remote REFER Mode	Regular
SDP Subsequent Responses Mode	Handle All	Remote Replaces Mode	Standard
Mediation Mode	RTP Mediation	Play RBT To Transferee	No
Extension Coders Group	--	Remote 3xx Mode	Transparent
<b>Allowed Audio Coders</b>	#0 [G711]	Send Header for Transfer	None
Allowed Coders Mode	Restriction		

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

IP Profiles [PSTN\_IP]

Allowed Video Coders	--	<b>SBC HOLD</b>	
Allowed Media Types		Remote Hold Format	Transparent
Direct Media Tag		Reliable Held Tone Source	Yes
<b>RFC 2833 Mode</b>	As Is	Play Held Tone	No
RFC 2833 DTMF Payload Type	0	<b>SBC FAX</b>	
Alternative DTMF Method	As Is	Fax Coders Group	--
Send Multiple DTMF Methods	Disable	Fax Mode	As Is
Receive Multiple DTMF Methods	Disable	Fax Offer Mode	All coders
Adapt RFC2833 BW to Voice coder BW	Disabled	Fax Answer Mode	Single coder
SDP Ptime Answer	Remote Answer	Remote Renegotiate on Fax Detection	Transparent
Preferred PTime	0	Fax Rerouting Mode	Disable
Use Silence Suppression	Transparent		
RTP Redundancy Mode	As Is		
RTCP Mode	Transparent		

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

IP Profiles [PSTN\_IP]

Jitter Compensation	Disable		
ICE Mode	Disable		
SDP Handle RTCP	Don't Care		
RTCP Mux	Not Supported		
RTCP Feedback	Feedback Off		
Re-number MID	Disable		
Voice Quality Enhancement	Disable		
Switch Coder Upon Voice Quality	Disable		
Max Opus Bandwidth	0		
Generate No-Op Packets	Disable		
Enhanced PLC	Disable		
SBC Multiple Coders	Not Supported		
SBC Allow Only Negotiated PT	Disable		
Add Media IP Change Header	Disable		

MEDIA	
Broken Connection Mode	Disconnect
No RTP Mode	Disconnect
Media IP Version Preference	Only IPv4
RTP Redundancy Depth	Disable

LOCAL TONES	
Local Ringback Tone Index	-1
Local Held Tone Index	-1

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

IP Profiles [PSTN\_IP]

SBC Precondition	Not Supported
BFCP IP from Audio Media	According to Global Parameter
Remove EXTMAP	Disable

QUALITY OF SERVICE	
RTP IP DiffServ	46
RTP Video DiffServ	-1
Signaling DiffServ	24
Data DiffServ	0

JITTER BUFFER	
Dynamic Jitter Buffer Minimum Delay [msec]	10
Dynamic Jitter Buffer Optimization Factor	10
Jitter Buffer Max Delay [msec]	300

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

Google CES:

IP Profiles [Google CCA\_IP]

GENERAL	
Index	2
Name	Google CCA_IP
Created by Routing Server	No
Used By Routing Server	Not Used

MEDIA SECURITY	
SBC Media Security Mode	Secured
Symmetric MKI	Disable

SBC SIGNALING	
PRACK Mode	Transparent
P-Asserted-Identity Header Mode	As Is
Diversion Header Mode	As Is
History-Info Header Mode	As Is
Session Expires Mode	Supported
SIP UPDATE Support	Supported
Remote re-INVITE	Supported
Remote Delayed Offer Support	Supported

Figure 45: IP Profile configurations of Google CES

IP Profiles [Google CCAL\_IP]

SBC Remove Crypto Lifetime in SDP	Yes	Keep Incoming Routing Headers	According to Operation Mode
SBC Remove Unknown Crypto	No	Keep User-Agent Header	According to Operation Mode
Crypto Suites Group	#0 [WAN]	Use Initial Incoming INVITE for Re-INVITE	Disable
Encryption on RTCP Packets	As Is	Handle X-Detect	No

**SBC EARLY MEDIA**

Remote Early Media	Supported
Remote Multiple 18x	Supported
Remote Early Media Response Type	Transparent
Remote Multiple Early Dialogs	According to Operation Mode

ISUP Body Handling	Transparent
ISUP Variant	Itu92
Max Call Duration [min]	0
Broken Signaling Connection Mode	Ignore
Disconnect In-Dialog Subscribe Failure	Enable

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

IP Profiles [Google CCAL\_IP]

Remote Early Media RTP Detection Mode	By Signaling	User Registration Time	0
Remote RFC 3960 Support	Not Supported	NAT UDP Registration Time	-1
Remote Can Play Ringback	Yes	NAT TCP Registration Time	-1
Generate RTP	None	UnRegister on WebSocket Disconnect	Enable

**SBC MEDIA**

SDP Subsequent Responses Mode	Handle All
Mediation Mode	RTP Mediation
Extension Coders Group	..
Allowed Audio Coders	#0 [G711]
Allowed Coders Mode	Restriction
Allowed Video Coders	..
Allowed Media Types	
Direct Media Tag	

**SBC FORWARD AND TRANSFER**

Remote REFER Mode	Regular
Remote Replaces Mode	Standard
Play RBT To Transferee	No
Remote 3xx Mode	Transparent
Send Header for Transfer	None

**SBC HOLD**

Remote Hold Format	Transparent
--------------------	-------------

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

IP Profiles [Google CCAL\_IP]

Allowed Video Coders	..
Allowed Media Types	
Direct Media Tag	
RFC 2833 Mode	As Is
RFC 2833 DTMF Payload Type	0
Alternative DTMF Method	As Is
Send Multiple DTMF Methods	Disable
Receive Multiple DTMF Methods	Disable
Adapt RFC2833 BW to Voice coder BW	Disabled
SDP Ptime Answer	Remote Answer
Preferred PTime	0
Use Silence Suppression	Transparent
RTP Redundancy Mode	As Is
RTCP Mode	Transparent

**SBC HOLD**

Remote Hold Format	Transparent
Reliable Held Tone Source	Yes
Play Held Tone	No

**SBC FAX**

Fax Coders Group	..
Fax Mode	As Is
Fax Offer Mode	All coders
Fax Answer Mode	Single coder
Remote Renegotiate on Fax Detection	Transparent
Fax Rerouting Mode	Disable

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

IP Profiles [Google CCAI\_IP]

Jitter Compensation	Disable	MEDIA	
ICE Mode	Disable	Broken Connection Mode	Disconnect
SDP Handle RTP	Don't Care	No RTP Mode	Disconnect
RTCP Mux	Not Supported	Media IP Version Preference	Only IPv4
RTCP Feedback	Feedback Off	RTP Redundancy Depth	Disable
Re-number MID	Disable		
Voice Quality Enhancement	Disable	LOCAL TONES	
Switch Coder Upon Voice Quality	Disable	Local Ringback Tone Index	-1
Max Opus Bandwidth	0	Local Held Tone Index	-1
Generate No-Op Packets	Disable		
Enhanced PLC	Disable		
SBC Multiple Coders	Not Supported		
SBC Allow Only Negotiated PT	Disable		
Add Media IP Change Header	Disable		

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

IP Profiles [Google CCAI\_IP]

Remove CSRC	Disable
SBC Precondition	Not Supported
BFCP IP from Audio Media	According to Global Parameter
Remove EXTMAP	Disable
QUALITY OF SERVICE	
RTP IP DiffServ	46
RTP Video DiffServ	-1
Signaling DiffServ	24
Data DiffServ	0
JITTER BUFFER	
Dynamic Jitter Buffer Minimum Delay [msec]	10
Dynamic Jitter Buffer Optimization Factor	10

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.

The screenshot shows the Audiocodes management console interface. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The left sidebar shows 'CORE ENTITIES' with 'IP Groups (3)' selected. The main area displays a table of IP Group configurations:

INDEX	NAME	SRD	TYPE	SBC OPERATION MODE	PROXY SET	IP PROFILE	MEDIA REALM	SIP GROUP NAME	CLASSIFY BY PROXY SET	INBOUND MESSAGE MANIPULATION SET	OUTBOUND MESSAGE MANIPULATION SET
0	PBX_IPG	DefaultSRD (#)	Server	B2BUA	PBX_PS	PBX_IP	PBX_MR	10.80	Enable	-1	-1
1	Google CCAI_IPG	DefaultSRD (#)	Server	B2BUA	Google CCAI SIPRI	Google CCAI_IP	Google CCAI	us.telephony.goo	Enable	-1	1
2	PSTN_IPG	DefaultSRD (#)	Server	B2BUA	PSTN_PS	PSTN_IP	PSTN_MR	PSTN	Enable	-1	-1

**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

The screenshot shows the configuration form for the IP Group 'PBX\_IPG'. The 'GENERAL' tab is active, and several fields are highlighted with red boxes:

- Index:** 0
- Name:** PBX\_IPG
- Type:** Server
- Proxy Set:** #0 [PBX\_PS]
- IP Profile:** #0 [PBX\_IP]
- Media Realm:** #0 [PBX\_MR]
- SIP Group Name:** 10.80

Other visible fields include 'Topology Location' (Up), 'Internal Media Realm' (--), and 'Contact User'.

**Figure 52: IP Group configurations of OnPrem PBX**

The screenshot shows the configuration form for the IP Group 'PBX\_IPG', focusing on the 'SBC REGISTRATION AND AUTHENTICATION' section. The 'SBC Operation Mode' is highlighted with a red box and set to 'B2BUA'.

Other visible settings include:

- Created By Routing Server:** No
- Used By Routing Server:** Not Used
- Proxy Set Connectivity:** Connected
- Classify By Proxy Set:** Enable
- Validate Source IP:** Disable
- SBC Client Forking Mode:** Sequential
- Max. Number of Registered Users:** -1
- Registration Mode:** User Initiates Registration
- Dedicated Connection Mode:** Disable
- User Stickiness:** Disable
- User UDP Port Assignment:** Disable
- Authentication Mode:** User Authenticates

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

IP Groups [PBX\_IPG]

**ADVANCED**

Local Host Name

UII Format

Always Use Src Address

Password As Client

Username As Server

Password As Server

Teams Registration Mode

**SBC ADVANCED**

Source URI Input

Destination URI Input

SIP Connect

SBC PSAP Mode

Route Using Request URI Port

Media TLS Context

**GW GROUP STATUS**

GW Group Registered IP Address

GW Group Registered Status

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

IP Groups [PBX\_IPG]

SBC PSAP Mode

Route Using Request URI Port

Media TLS Context

Keep Original Call-ID

Dial Plan  [View](#)

Call Setup Rules Set ID

Tags

SBC Alternative Routing Reasons Set  [View](#)

Teams Local Media Optimization Handling

Teams Local Media Optimization Initial Behavior

Teams Local Media Optimization Site

Teams Direct Routing Mode

Metering Remote Type

Report Metering

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.

IP Groups [Google CCAI\_IPG]

**GENERAL**

Index

Name

Topology Location

Type

Proxy Set  [View](#)

IP Profile  [View](#)

Media Realm  [View](#)

Internal Media Realm  [View](#)

Contact User

SIP Group Name

Created By Routing Server

Used By Routing Server

Proxy Set Connectivity

**QUALITY OF EXPERIENCE**

QoE Profile  [View](#)

Bandwidth Profile  [View](#)

User Voice Quality Report

**MESSAGE MANIPULATION**

Inbound Message Manipulation Set

Outbound Message Manipulation Set

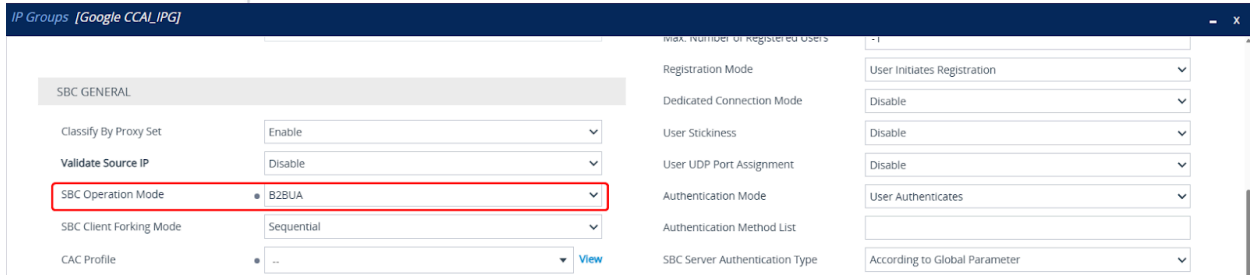
Message Manipulation User-Defined String 1

Message Manipulation User-Defined String 2

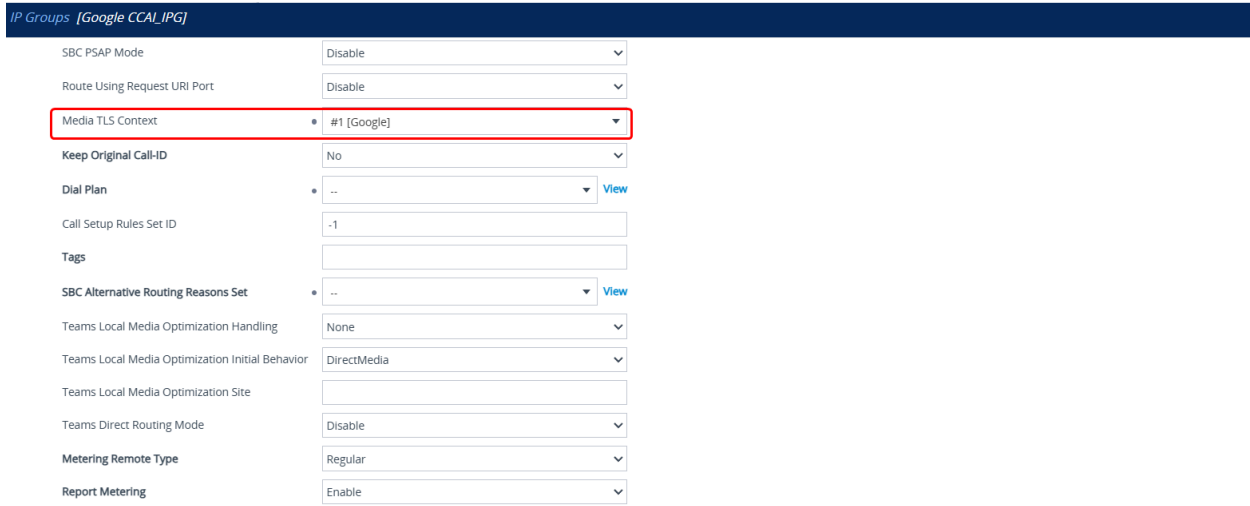
Proxy Keep-Alive using IP Group settings

**SBC REGISTRATION AND AUTHENTICATION**

**Figure 56: IP Group Configurations of Google CES**

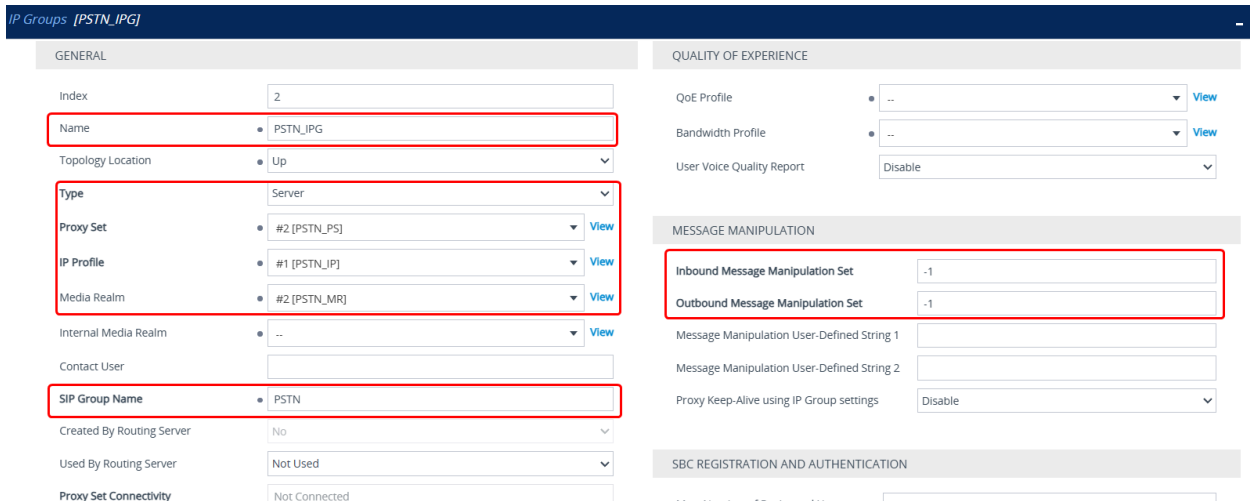


**Figure 57: IP Group Configurations of Google CES (Cont.)**



**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**



IP Groups [PSTN\_IPG]

SBC GENERAL			
Classify By Proxy Set	Enable	Dedicated Connection Mode	Disable
Validate Source IP	Disable	User Stickiness	Disable
SBC Operation Mode	B2BUA	User UDP Port Assignment	Disable
SBC Client Forking Mode	Sequential	Authentication Mode	User Authenticates
CAC Profile	..	Authentication Method List	
SIP Source Host Name		SBC Server Authentication Type	According to Global Parameter
		OAuth HTTP Service	..
		Username As Client	
		Password As Client	
		Username As Server	
		Password As Server	
		Teams Registration Mode	Disable

ADVANCED	
Local Host Name	
UII Format	Disable
Always Use Src Address	No

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

IP Groups [PSTN\_IPG]

Always Use Src Address: No

SBC ADVANCED		GW GROUP STATUS	
Source URI Input		GW Group Registered IP Address	
Destination URI Input		GW Group Registered Status	NA
SIP Connect	No		
SBC PSAP Mode	Disable		
Route Using Request URI Port	Disable		
Media TLS Context	#0 [default]		
Keep Original Call-ID	No		
Dial Plan	..		
Call Setup Rules Set ID	-1		
Tags			
SBC Alternative Routing Reasons Set	..		

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

IP Groups [PSTN\_IPG]

SBC PSAP Mode	Disable
Route Using Request URI Port	Disable
Media TLS Context	#0 [default]
Keep Original Call-ID	No
Dial Plan	..
Call Setup Rules Set ID	-1
Tags	
SBC Alternative Routing Reasons Set	..
Teams Local Media Optimization Handling	None
Teams Local Media Optimization Initial Behavior	DirectMedia
Teams Local Media Optimization Site	
Teams Direct Routing Mode	Disable
Metering Remote Type	Regular
Report Metering	Enable

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.

The screenshot shows the 'Media Security' configuration page. The 'Media Security' checkbox is checked and highlighted with a red box. Below it, 'Offered SRTP Cipher Suites' is set to 'All', 'ARIA Protocol Support' is unchecked, and 'Master Key Identifier (MKI) Size' is set to 0. The 'Symmetric MKI' checkbox is also unchecked. On the right, under 'AUTHENTICATION & ENCRYPTION', 'Authentication on Transmitted RTP Packets', 'Encryption on Transmitted RTP Packets', and 'Encryption on Transmitted RTCP Packets' are all set to 'Active'. 'SRTP Tunneling Authentication for RTP' and 'SRTP Tunneling Authentication for RTCP' are both unchecked.

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.

The screenshot shows the 'IP-to-IP Routing' configuration page. A table lists three routing rules:

INDEX	NAME	ROUTING POLICY	ALTERNATIVE ROUTE OPTIONS	SOURCE IP GROUP	REQUEST TYPE	SOURCE USERNAME PATTERN	DESTINATION USERNAME PATTERN	DESTINATION TYPE	DESTINATION IP GROUP	DESTINATION SIP INTERFACE	DESTINATION ADDRESS
0	OPTIONS	Default_SBCRoute	Route Row	Any	OPTIONS	*	*	Dest Address	--	--	internal
1	PBX to PSTN	Default_SBCRoute	Route Row	PBX_IPG	All	*	*	IP Group	PSTN_IPG	PSTN	
2	PSTN to PBX	Default_SBCRoute	Route Row	PSTN_IPG	All	*	*	IP Group	PBX_IPG	PBX	

Below the table, there is a section for '#0 [OPTIONS]' with an 'Edit' button.

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.

The screenshot shows the Audiocodes management interface. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. Below it, the 'SIGNALING & MEDIA' tab is selected. The left sidebar shows a 'TOPOLOGY VIEW' with a tree structure where 'SIP RECORDING' is expanded and 'SIP Recording Settings' is selected. The main content area is titled 'SIP Recording Settings' and has a 'GENERAL' section. A red box highlights the 'Recording Server (SRS) Destination Username' field, which contains the value '+136140'. Other fields include 'SIP Recording Time Stamp Format' (Local Time), 'SIP Recording Metadata Format' (Legacy), and 'Video Recording Sync Timeout' (2000).

Figure 65: SIP Recording Settings

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

The screenshot shows the Audiocodes management interface for 'SIP Recording Rules'. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. Below it, the 'SIGNALING & MEDIA' tab is selected. The left sidebar shows a 'TOPOLOGY VIEW' where 'SIP RECORDING' is expanded and 'SIP Recording Rules (1)' is selected. The main content area is titled 'SIP Recording Rules (1)' and shows a table with one rule. The rule is highlighted with a red box. Below the table, the rule details are shown in a 'GENERAL' and 'RECORDING SERVER' section.

INDEX	RECORDED IP GROUP	PEER IP GROUP	CALLER	RECORDING SERVER (SRS) IP GROUP	RECORDING SERVER (SRS) IP GROUP SET
0	PSTN_IPG	PBX_IPG	Both	Google CCAI_IPG	--

**#0 [PSTN\_IPG]** [Edit](#)

**GENERAL**

- Recorded IP Group: \* PSTN\_IPG [View](#)
- Recorded Source Pattern: \*
- Recorded Destination Patt...: \*
- Condition: \* -- [View](#)

**RECORDING SERVER**

- Recording Server (SRS) IP ...: \* Google CCAI\_IPG [View](#)
- Redundant Recording Ser...: \* -- [View](#)
- Recording Server (SRS) IP ...: \* -- [View](#)

Figure 66: SIP Recording Rules

- Create SIP recording rules as shown below.

SIP Recording Rules [PSTN\_IPG/2]

GENERAL		RECORDING SERVER	
Index	0	Recording Server (SRS) IP Group	#1 [Google CCAL_IPG] <a href="#">View</a>
Recorded IP Group	#2 [PSTN_IPG] <a href="#">View</a>	Redundant Recording Server (SRS) IP Group	-- <a href="#">View</a>
Recorded Source Pattern	*	Recording Server (SRS) IP Group Set	-- <a href="#">View</a>
Recorded Destination Pattern	*		
Condition	-- <a href="#">View</a>		
Peer IP Group	#0 [PBX_IPG] <a href="#">View</a>		
Caller	Both		
Trigger	Call Connect		
Recording Server Role			

**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.

INDEX	NAME	MANIPULATION SET ID	MESSAGE TYPE	CONDITION	ACTION SUBJECT	ACTION TYPE	ACTION VALUE	ROW ROLE
0	call-info Google	1	Invite		Header:call-info	Add	'<http://dialogflow.google	Use Current Condi
1	removecallinfo	1	Any Request	Header:Call-Info regex (<	Header:Call-Info	Modify	'\$1+\$3	Use Current Condi
2	Request URI	1	Invite Request		Header:To URL User	Modify	'Audiocodes'	Use Current Condi
3	from	1	Invite Request		Header:From URL Host	Modify	'192.65	Use Current Condi
4	PAI modify	1	any		Header:P-Asserted-Ident	Modify	'sbc12.tekvizionlabs.com'	Use Current Condi
5	Contact	1	Invite Request		Header:Contact URL Hos	Modify	'192.65	Use Current Condi
6	from url	1	Invite Request		Header:From URL User	Add	'Audiocodes'	Use Current Condi
7	Contact URL	1	Invite Request		Header:Contact URL Hos	Modify	'192.65	Use Current Condi

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 Google 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.`

Message Manipulations [call-info Google]

**GENERAL**

Index: 0

Name: call-info Google

Manipulation Set ID: 1

Row Role: Use Current Condition

**MATCH**

Message Type: Invite

Condition: [Empty]

**ACTION**

Action Subject: Header:call-info

Action Type: Add

Action Value: '<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.

Message Manipulations: [removecallinfo]

GENERAL		ACTION	
Index	1	Action Subject	Header.Call-Info
Name	removecallinfo	Action Type	Modify
Manipulation Set ID	1	Action Value	\$1-\$3
Row Role	Use Current Condition		

MATCH	
Message Type	Any Request
Condition	Header.Call-Info regex (<http://*>@192.65.[0-9].*)

**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'

Message Manipulations: [from]

GENERAL		ACTION	
Index	3	Action Subject	Header.From.URL.Host
Name	from	Action Type	Modify
Manipulation Set ID	1	Action Value	*192.65.[0-9].*
Row Role	Use Current Condition		

MATCH	
Message Type	Invite Request
Condition	

**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'

Message Manipulations: [from\_user]

GENERAL		ACTION	
Index	4	Action Subject	Header.From.URL.User
Name	from_user	Action Type	Add
Manipulation Set ID	1	Action Value	'AudioCodes'
Row Role	Use Current Condition		

MATCH	
Message Type	Invite Request
Condition	

**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'

GENERAL		ACTION	
Index	4	Action Subject	Header: P-Asserted-Identity.URL.Host <a href="#">Editor</a>
Name	PAI modify	Action Type	Modify
Manipulation Set ID	1	Action Value	'sbc12.tekvizionlabs.com' <a href="#">Editor</a>
Row Role	Use Current Condition		

MATCH	
Message Type	any <a href="#">Editor</a>
Condition	<a href="#">Editor</a>

**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'

Message Manipulations: [To\_User]

GENERAL		ACTION	
Index	2	Action Subject	Header.To.URL.User <a href="#">Editor</a>
Name	To_User	Action Type	Modify
Manipulation Set ID	1	Action Value	'Audiocodes' <a href="#">Editor</a>
Row Role	Use Current Condition		

MATCH	
Message Type	Invite.Request <a href="#">Editor</a>
Condition	<a href="#">Editor</a>

**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'

Message Manipulations: [Contact URL]

GENERAL		ACTION	
Index	7	Action Subject	Header.Contact.URL.Host <a href="#">Editor</a>
Name	Contact URL	Action Type	Modify
Manipulation Set ID	1	Action Value	'192.65.X.X' <a href="#">Editor</a>
Row Role	Use Current Condition		

MATCH	
Message Type	Invite.Request <a href="#">Editor</a>
Condition	<a href="#">Editor</a>

**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.
- 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
```

Message Manipulations [call-info Google]

**GENERAL**

Index: 0

Name: call-info Google

Manipulation Set ID: 1

Row Role: Use Current Condition

**MATCH**

Message Type: invite

Condition:

**ACTION**

Action Subject: Header: call-info

Action Type: Add

Action Value: <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[redacted]@us.telephony.goog;user=phone SIP/2.0
Via: SIP/2.0/TLS 192.65.[redacted]:5061;alias;branch=z9hG4bKac1515878973
Max-Forwards: 70
From: <sip:Audiocodes@192.65.[redacted];user=phone>;tag=1c1451797144
To: <sip:Audiocodes@us.telephony.goog;user=phone>
Call-ID: 14526490261920251384@192.65.[redacted]
CSeq: 1 INVITE
Contact: <sip:192.65.[redacted]:5061;transport=tls>;src
Supported: timer,replaces,resource-priority,sdp-anat
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO
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-3898XX/conversations/Sr_145[redacted];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.[redacted]
s=SBC-Call
c=IN IP4 192.65.[redacted]
t=0
m=audio 7064 RTP/SAVP 0 101
c=IN IP4 192.65.[redacted]
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:rtQOzfvjF5y/Ajk0b0DySxQmA/Sh+6GTjko78ik
a=crypto:3 AES_256_CM_HMAC_SHA1_80 inline:WpppEpYoE4mUe7JfXKdDCzcpUMP1fGsXfyeZErq1q82u61/qRd+cdQ612fCwg==
a=crypto:4 AES_256_CM_HMAC_SHA1_32 inline:1IiZjZIMiUklrE00FqYkHGoHuc6TFMk+w9t1f1xU+ZwPcx6KDU5QNMd87G0w==
m=audio 7068 RTP/SAVP 0 101
c=IN IP4 192.65.[redacted]
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+C6C9TdTCHzN+0fYwuEqW+hcGOi
a=crypto:2 AES_CM_128_HMAC_SHA1_32 inline:3Q96EGBNFmJiwxj6wXxlFLFhUV9Yt0asgUGz+D
a=crypto:3 AES_256_CM_HMAC_SHA1_80 inline:QbNdrW3B+ElV54j6+vwfijFfww0W7WC/kLkKjo7wdfTieZm/nJvVDk5tjRA==
a=crypto:4 AES_256_CM_HMAC_SHA1_32 inline:0/kJ2cCn2hm6jwR6Db6HJQpk3L0FV88Eieq03RsitgeappqrnFimeZ/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[redacted]@us.telephony.goog SIP/2.0
Via: SIP/2.0/TLS 192.65.[redacted]:5061;alias;branch=z9hG4bKac210501
Max-Forwards: 69
From: "Pradeep Gopal" <sip:214550[redacted]@192.65.[redacted]>;tag=1c1363300463
To: <sip:+131494[redacted]@us.telephony.goog>
Call-ID: 1851113439268202513032@192.65.[redacted]
CSeq: 1 INVITE
Contact: <sip:214550[redacted]@192.65.[redacted]:5061;transport=tls>
Supported: 100rel,timer,replaces,resource-priority,sdp-anat
Allow: INVITE,OPTIONS,BYE,CANCEL,ACK,PRACK,UPDATE,REFER,INFO
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:214550[redacted]@192.65.[redacted]>
Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898XX/conversations/Sr_18[redacted]32>;purpose=Goog-ContactCenter-Conversation
Date: Tue, 26 Nov 2025 13:00:55 GMT
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 633
Cisco-Guid: 2295206282-2175996[redacted]
Timestamp: 1756213255
v=0
o=CiscoSystemsSIP-GW-UserAgent 658476784 1219741378 IN IP4 192.65.[redacted]
s=SIP Call
c=IN IP4 192.65.[redacted]
t=0
m=audio 7192 RTP/SAVP 0 101
c=IN IP4 192.65.[redacted]
a=ptime:101 0-16
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:gV51G4J4rFnAkwm6t6U9Hd90yEsbBVwNrIkQgBVQ
a=crypto:2 AES_CM_128_HMAC_SHA1_32 inline:3HvTldAZ/quJlW23wchll14yLx[redacted]
a=crypto:3 AES_256_CM_HMAC_SHA1_80 inline:7Jj1VVOjEONIo9gRZztFnaLGISUgSgFlafy6751dZYMyq+LGRB5BSyH0d8djw==
a=crypto:4 AES_256_CM_HMAC_SHA1_32 inline:UvPi02AA0v13J3R0yBEU8mx5/Hrojmrqs/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
<b>SBC Configuration Verification</b>					
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	
<b>Inbound</b>					
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/p articipants 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/p articipants	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	<p>When the roles are set to "HUMAN AGENT" and "END USER,"            (Call-Info&lt;<a ;purpose='Google-ContactCenter-Conversation"' href="http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898XX/conversations/Sr_XXX816?roles=HUMAN_AGENT,END_USER">http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898XX/conversations/Sr_XXX816?roles=HUMAN_AGENT,END_USER";purpose=Google-ContactCenter-Conversation</a>) the transcript shows the first media stream with the participation role as "HUMAN AGENT," followed by "END USER."</p> <p>The transcript indicates that HUMAN AGENT was listed first, followed by the END USER, in 7 out of 10 attempts.</p>
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	