



# **Amazon Chime Voice Connector**

## **SIPREC Configuration Guide:**

### **Genesys Cloud and AudioCodes Mediant Virtual Edition (VE) SBC**

**May 2022**

## Document History

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# 1 Audience

This document is intended for technical staff and Value-Added Resellers (VAR) with installation and operational responsibilities. This configuration guide provides steps for configuring **SIPREC** using **Genesys Cloud** and **AudioCodes Mediant VE SBC** to connect to **Amazon Chime Voice Connector** for streaming audio to Amazon Kinesis Video Streams (KVS)

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## 1.1 Amazon Chime Voice Connector

Amazon Chime Voice Connector is a pay-as-you-go service that enables companies to make or receive secure phone calls over the internet or AWS Direct Connect using their existing telephone system or session border controller (SBC). The service has no upfront fees, elastically scales based on demand, supports calling both landline and mobile phone numbers in over 100 countries, and gives customers the option to enable inbound calling, outbound calling, or both.

Amazon Chime Voice Connector uses the industry-standard Session Initiation Protocol (SIP). Amazon Chime Voice Connector does not require dedicated data circuits. A company can use their existing Internet connection or AWS Direct Connect public virtual interface for SIP connectivity to AWS. Voice connectors can be configured in minutes using the AWS Management Console or Amazon Chime API. Amazon Chime Voice Connector offers cost-effective rates for inbound and outbound calls. Calls into Amazon Chime meetings, as well as calls to other Amazon Chime Voice Connector customers are at no additional cost. With Amazon Chime Voice Connector, companies can reduce their voice calling costs without having to replace their on-premises phone system.

## 2 SIP Trunking Network Components

The network for SIP Trunk reference configuration is illustrated below and is representative of **Genesys Cloud** using **AudioCodes Mediant VE SBC** with **Amazon Chime Voice Connector**.

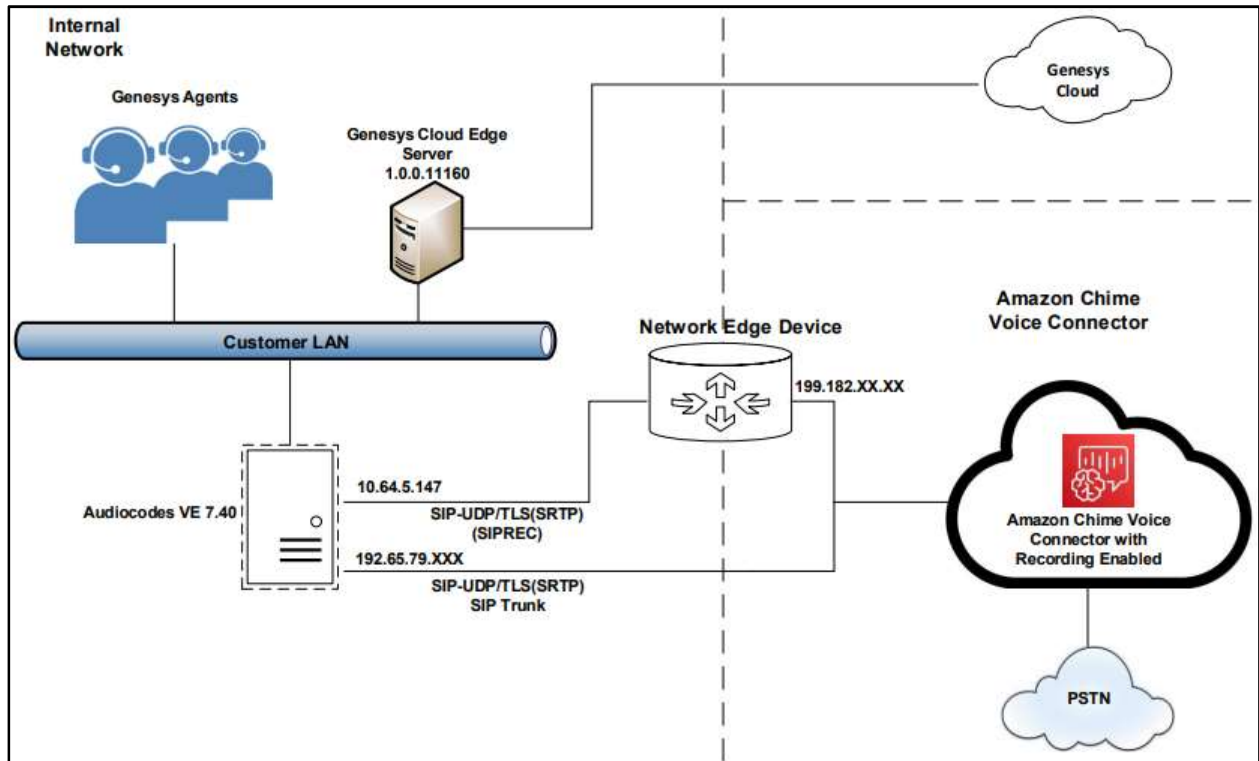


Figure 1: Network Topology

## **2.1 Hardware Components**

- Genesys Cloud Edge Server
- AudioCodes Mediant VE SBC running on UCS VMware ESXi 6.5 server

## **2.2 Software Requirements**

- Genesys Cloud Edge Server – 1.0.0.11160
- AudioCodes Mediant VE SBC version 7.40A.250.004

# **3 Features**

## **3.1 Features Supported**

- SIPREC

## **3.2 Features Not Supported**

- Making conference call when a call is originated from queue in Genesys Cloud

## **3.3 Features Not Tested**

- None

## **3.4 Caveats and Limitations**

- None

## 4 Configuration

The specific values listed in this guide are used in the lab configuration described in this document and are for illustrative purposes only. You must obtain and use the appropriate values for your deployment. Encryption is always recommended if supported.

### 4.1 Configuration Checklist

In this section we present an overview of the steps that are required to **Genesys Cloud (Edge Server)** and **AudioCodes Mediant VE SBC** for SIPREC with **Amazon Chime Voice Connector**.

Table 1 – Configuration Steps

Steps	Description	Reference
Step 1	Genesys Cloud Configuration	<a href="#">Section 4.2</a>
Step 2	AudioCodes Mediant VE SBC configuration	<a href="#">Section 4.3</a>
Step 3	Amazon Chime Voice Connector Configuration	<a href="#">Amazon Chime Voice Connector</a>

## 4.2 Genesys Cloud Configuration

This section with screen shots taken from Genesys Cloud used for the interoperability testing gives a general overview of the Genesys Cloud configuration.

### 4.2.1 Login and Version

1. Access Genesys Cloud URL / application and enter the required credentials to perform the configuration.

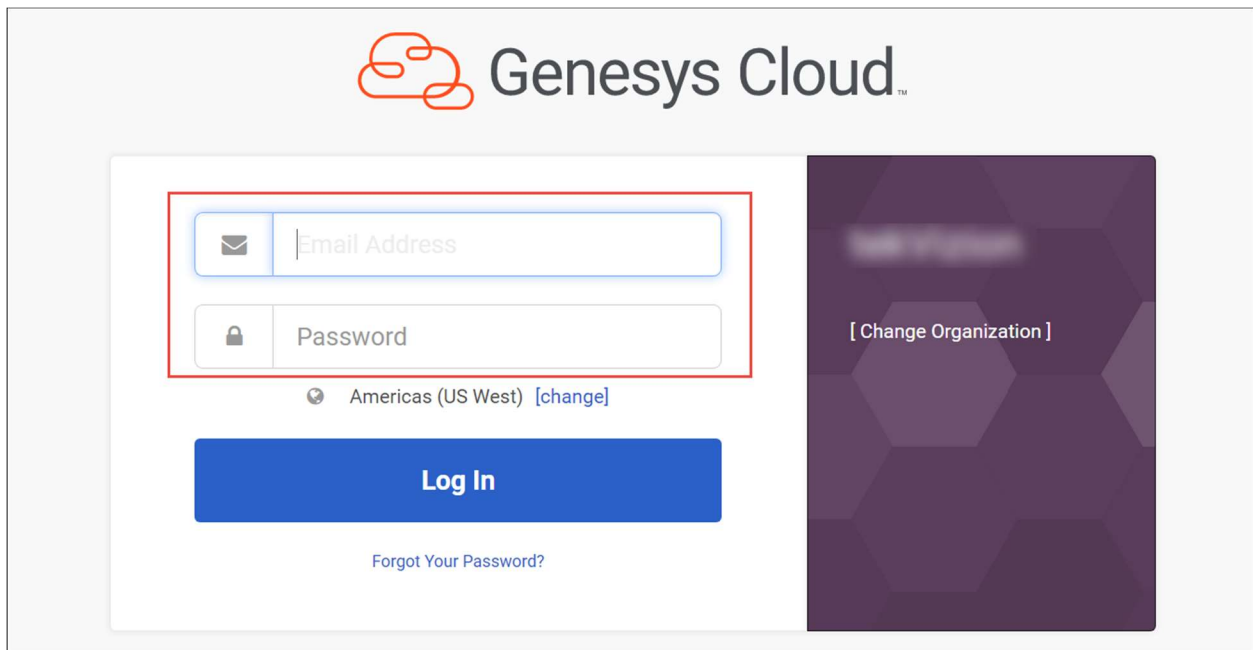


Figure 2: Genesys Cloud Login

## 4.2.2 Add Phone

1. Click on **Admin** then choose **Phone Management > Base settings**
2. Click on **Add Base Settings** to select the phone model.
3. Enter the Base settings Name and Choose the required phone model in **Phone Make and Model** field.
4. Save the Base settings.
5. To add phone, go to **Phone Management > Phones**.
6. Click on **Add Phone**.
7. Enter the Phone name and choose the proper **base settings** and **Site**.

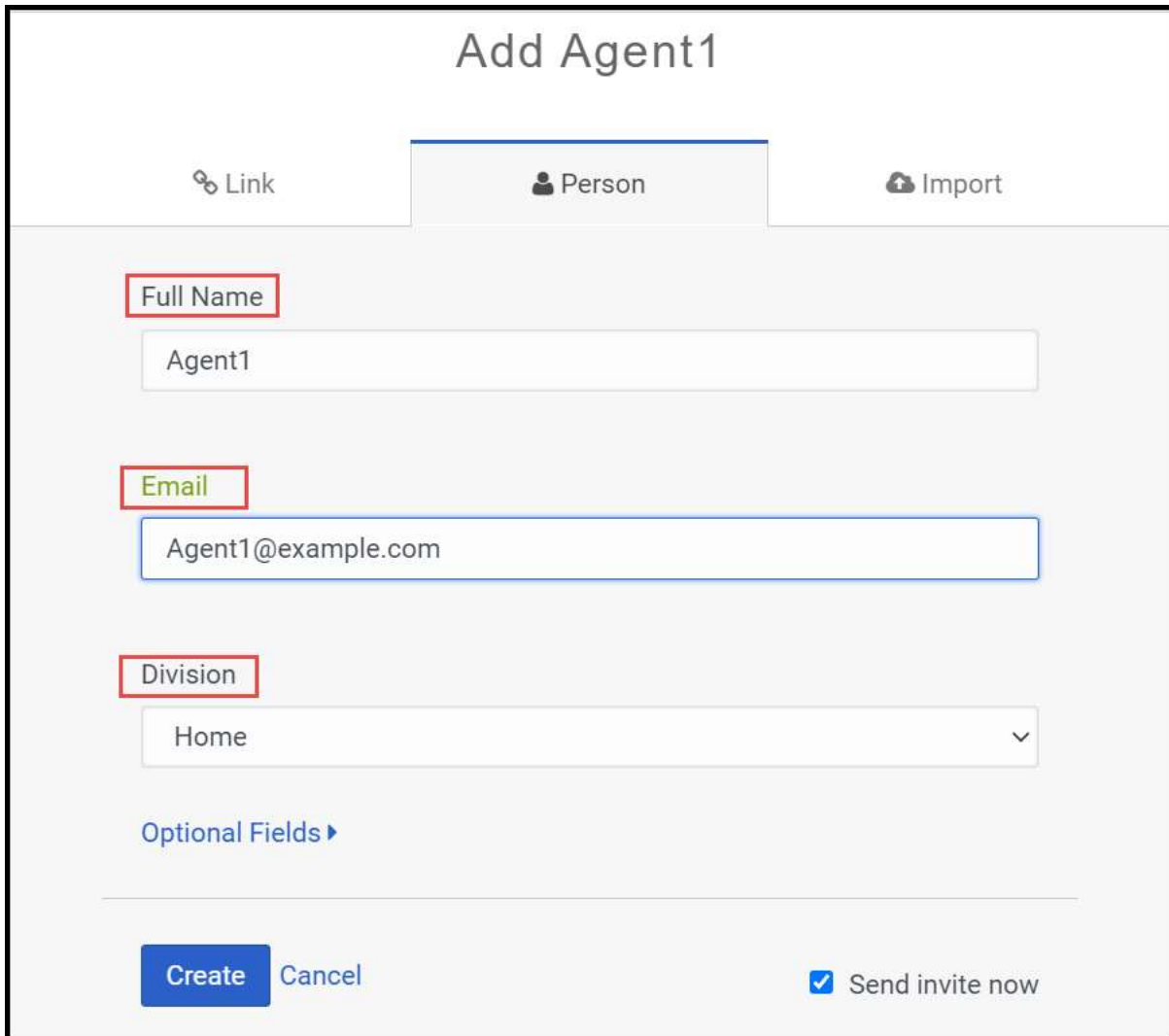
The screenshot displays the Genesys Cloud interface for adding a new phone. The top navigation bar includes 'Admin', which is highlighted. The breadcrumb trail shows 'Telephony / Phone Management / Phones / Create Phone'. The left sidebar contains various management options, with 'Phone Management' highlighted. The main form area is titled 'Create Phone' and contains the following fields:

- Topology:** A dropdown menu showing 'Phone' with a warning icon.
- Phone Name:** A text input field containing 'Agent1\_webrtc'.
- Base Settings:** A dropdown menu showing 'Webrtc' with a warning icon.
- Site:** A dropdown menu showing 'New\_site' with a warning icon.
- Person:** A dropdown menu showing 'Agent1'.
- Phone Configuration:** A section with two expandable options: 'General' and 'Media'.

Figure 3: Add Phone

### 4.2.3 Add User

1. Click on **Admin**, Under **People & Permission** choose **People**
2. Click on **Add Person**
3. Provide Agent's **Full Name, Email, Division** and click **Create**.



The screenshot shows a form titled "Add Agent1" with three tabs: "Link", "Person", and "Import". The "Person" tab is selected. The form contains three required fields: "Full Name" (containing "Agent1"), "Email" (containing "Agent1@example.com"), and "Division" (a dropdown menu with "Home" selected). Below these fields is an "Optional Fields" section with a right-pointing arrow. At the bottom, there is a "Create" button, a "Cancel" button, and a checked checkbox labeled "Send invite now".

Figure 4: Add User

4. Select the created people and go to the **Roles** tab, ensure the following are chosen
  - a. Employee
  - b. User [Role for basic agent functions]

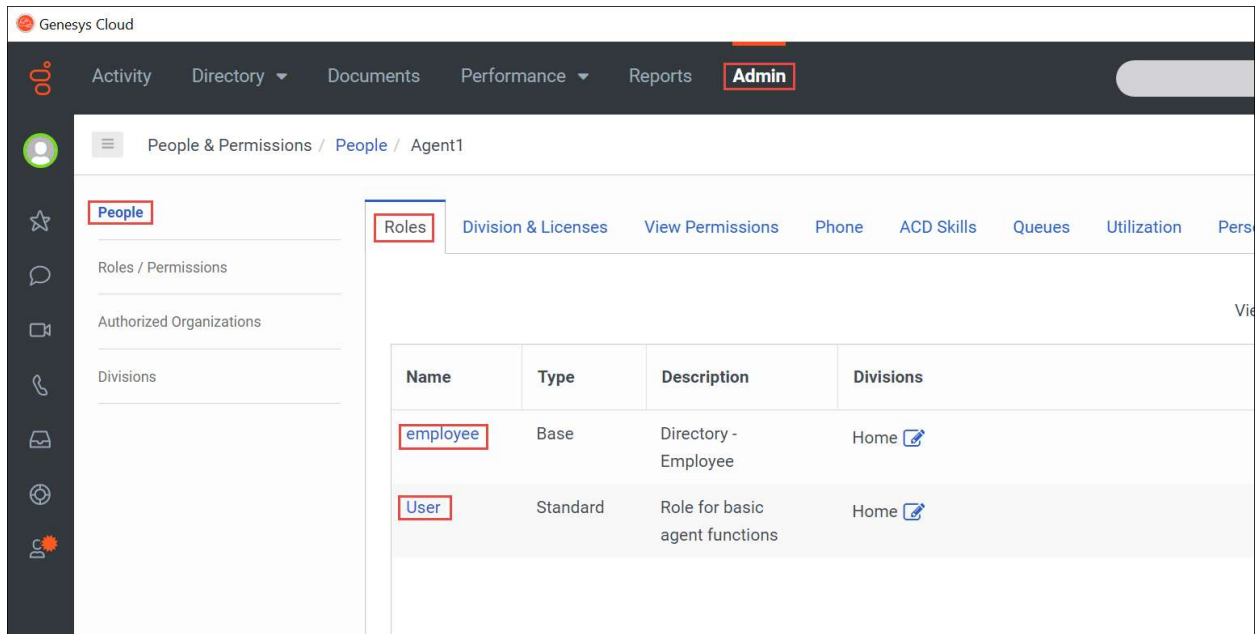


Figure 5: Add User Contd.,

## 5. Choose the **License type** as **Cloud CX3**

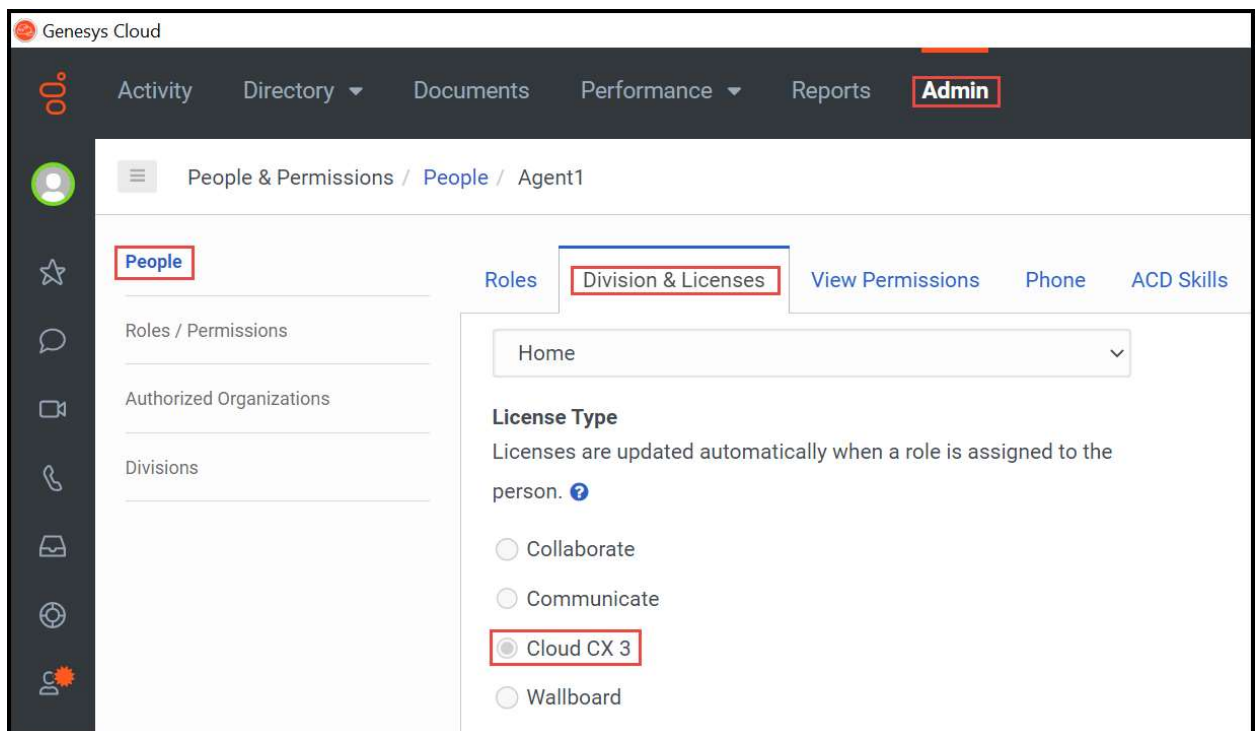


Figure 6: Add User Contd.,

6. Choose the Phone created previously and associate it with the user in the **Default phone** field

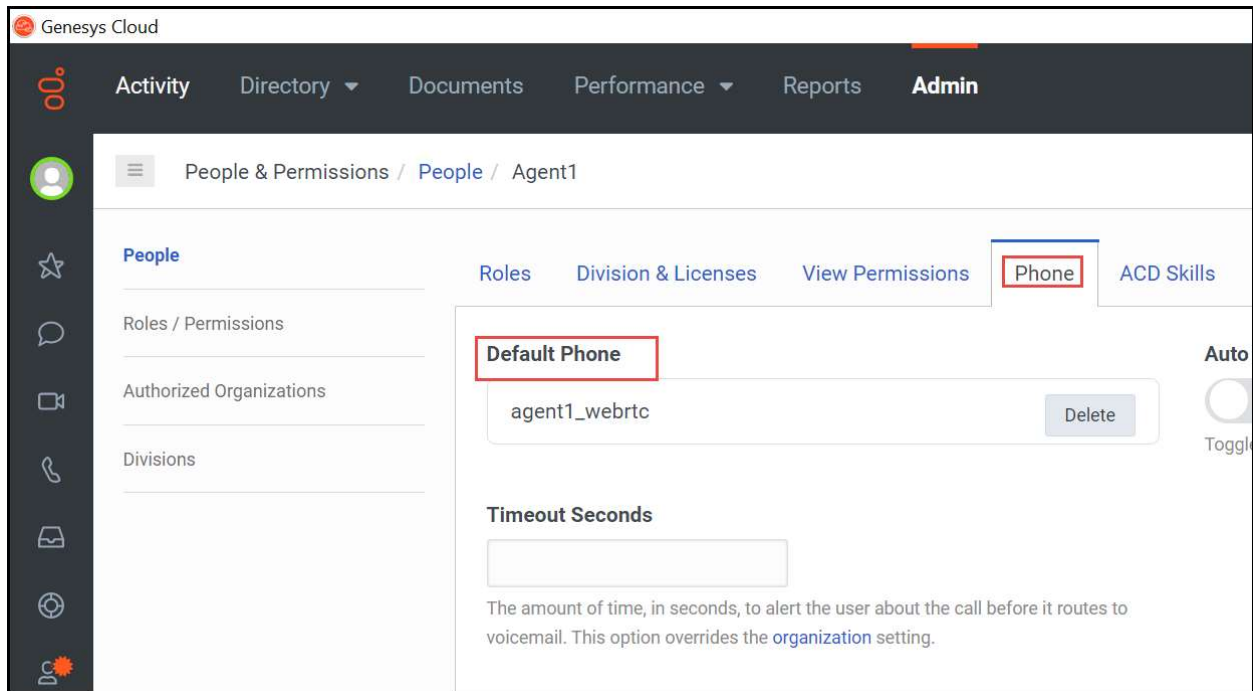


Figure 7: Add User Contd.,

## 4.2.4 SIP Trunk

1. Click on **Admin**, Under **Telephony** choose **Trunks**
2. In the **External Trunk** tab, Click on **Create New**
3. Enter the **External Trunk Name**, **Protocol**, **Listen Port**

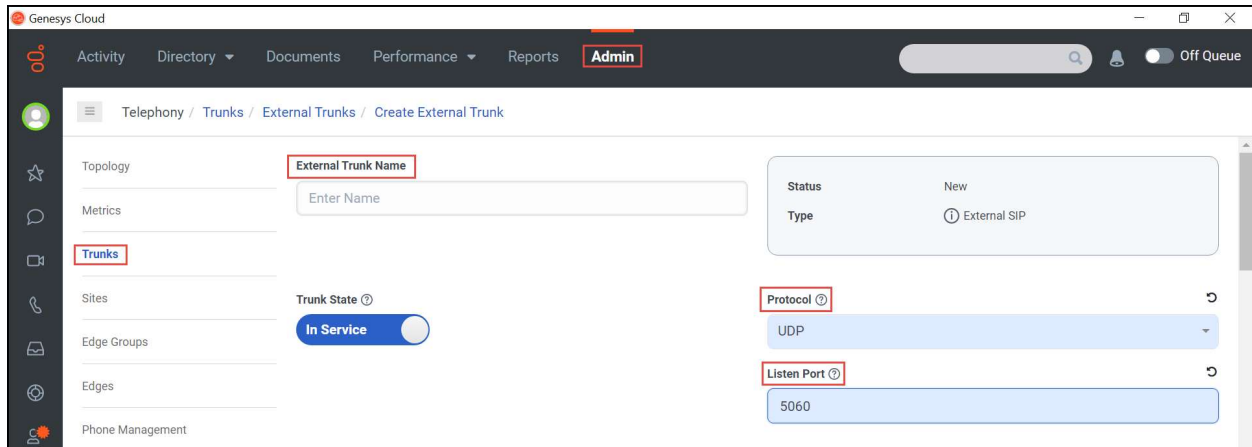


Figure 8: SIP Trunk

In the **Outbound** section enter the **SIP Servers or Proxies** address (LAN side IP address of AudioCodes VE SBC)



Figure 9: SIP Trunk Contd.,

4. In the **SIP Access Control** Section enter the SBC LAN side IP address in the **Allow the Following Addresses** field and leave the rest of the fields to default

**SIP Access Control** ⓘ

Use Source Address ⓘ  
 Yes

Allow the Following Addresses ⓘ  
 10.64.5.147  
 Add an IP or CIDR address +

Always Deny the Following Addresses ⓘ  
 No exceptions  
 Add an IP or CIDR address +

Allow All

Figure 10: SIP Trunk Contd.,

In the **External Trunk Configuration**, Leave the values to default in all sections except Media.

- In the **Media** section, Choose the Codec in **Preferred Codec List**, **DTMF Payload** and **DTMF Method** and leave the rest to default values.

**Preferred Codec List** ⓘ

audio/PCMU  
 audio/PCMA

Select a Codec

**DTMF Payload** ⓘ  
 101

**DTMF Method** ⓘ  
 RTP Events

**SRTP Cipher Suite List** ⓘ  
 AES\_CM\_128\_HMAC\_SHA1\_80

Select a Cipher Suite

Show legacy SRTP ciphers

**Disconnect on Idle RTP** ⓘ  
 Enabled

**Ringback** ⓘ  
 Enabled

Figure 11: SIP Trunk Contd.,

## 4.2.5 Dial Plan

1. Click on **Admin**, Under **Telephony** choose **Sites**
2. In the **Number Plans** tab, Click on **New Number Plan**
3. Enter the **Number Plan Name**, **Match Type**, **Match Expression**, **Normalized Number Expression** and **Classification**

In the **Outbound Routes** tab, Click on **New Outbound Route** and enter the **Outbound Route Name**, **Classification** (required Number Plan ) and select the **External Trunk** (SIP trunk as created earlier)

The screenshot shows the 'New Outbound Route' configuration page in the AWS Management Console. The form is titled '+ New Outbound Route' and has a 'Default Outbound Route' button on the left. The main form fields are:

- Outbound Route Name:** Default Outbound Route
- Description:** (Empty text box)
- State:** Enabled (Toggle switch)
- Classifications:** Emergency x, National x, International x, Network x
- Distribution Pattern:** Sequential (Selected), Random (Unselected)
- External Trunks:** testtrunk (Selected)

At the bottom left, there are two buttons: 'Save Outbound Routes' and 'Cancel'.

Figure 12: Dial Plan

## 4.2.6 Create Queue

1. Click on **Admin**, Under **Contact Center** choose **Queues**
2. Click on **Create Queue** and provide Queue name, Division and save it.

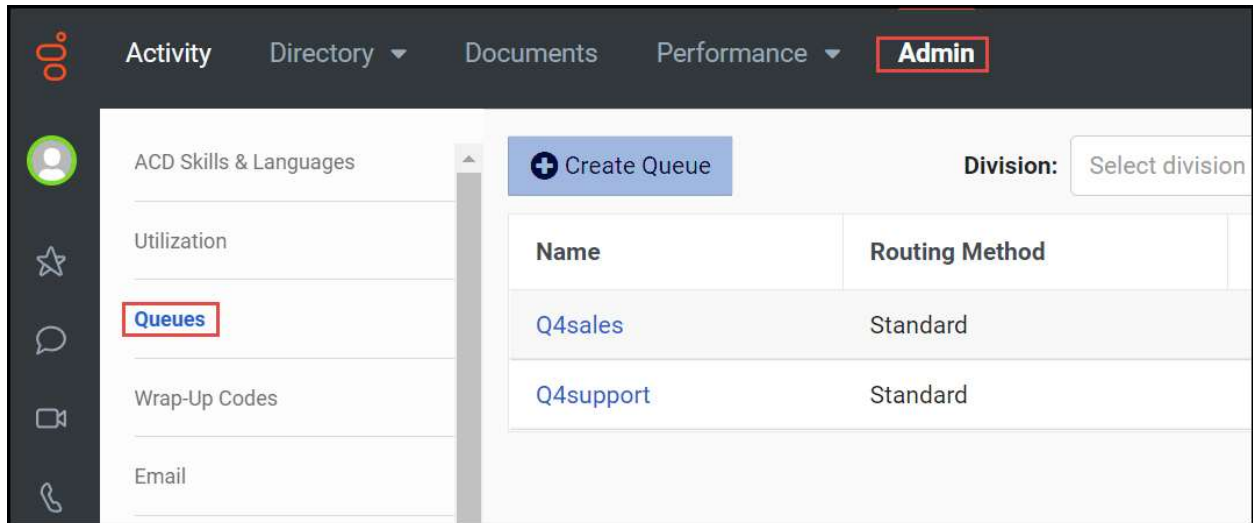


Figure 13: Create Queue

3. Click on above created Queue and select **After call work** in the **General** Tab.

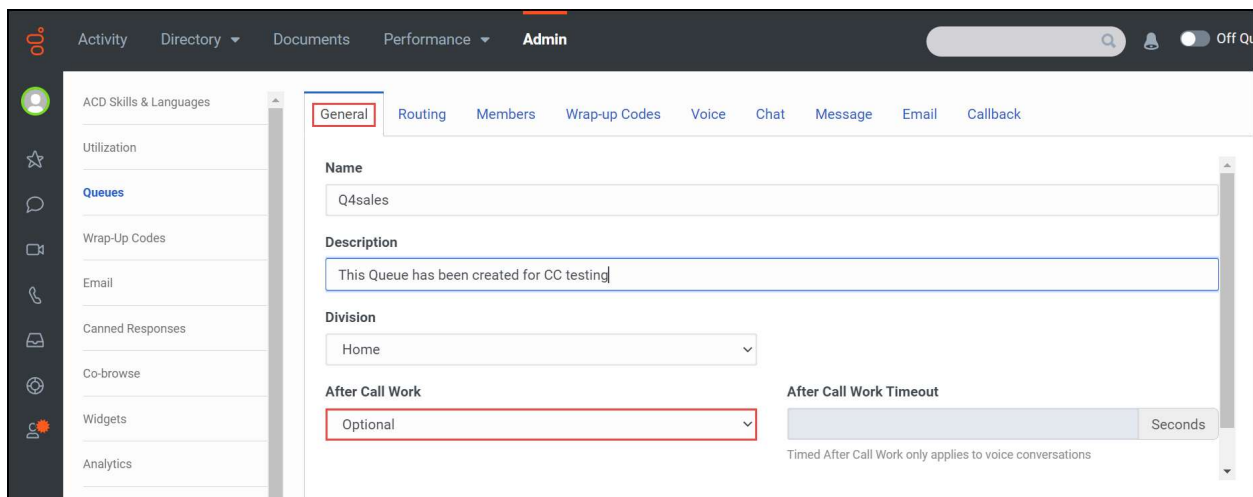


Figure 14: Create Queue Contd.,

4. Click on **Routing** tab and select **Routing method** and **evaluation method**.

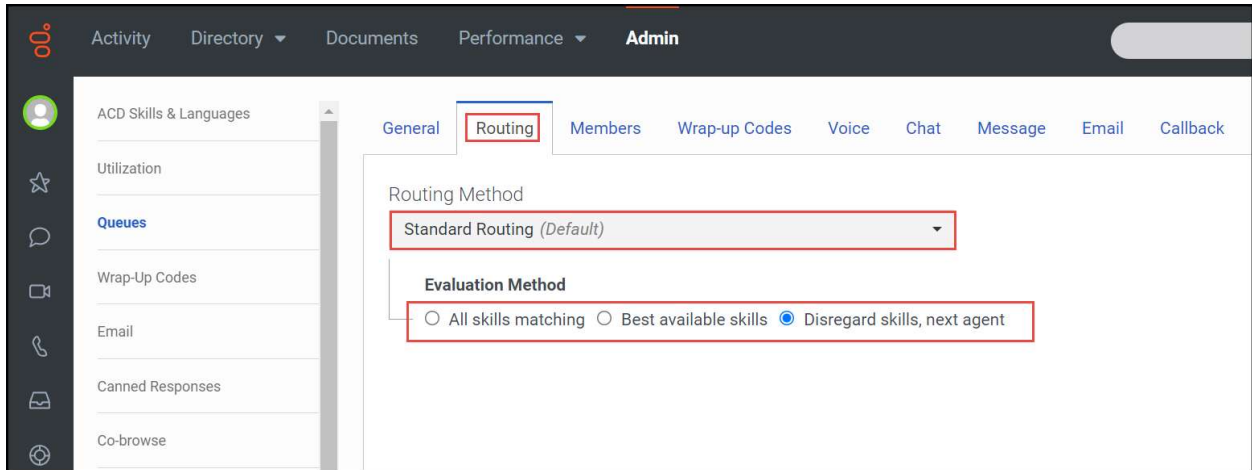


Figure 15: Create Queue Contd.,

5. Click on **Members** tab and add the Agents to this Queue by using **Select new members** search box.

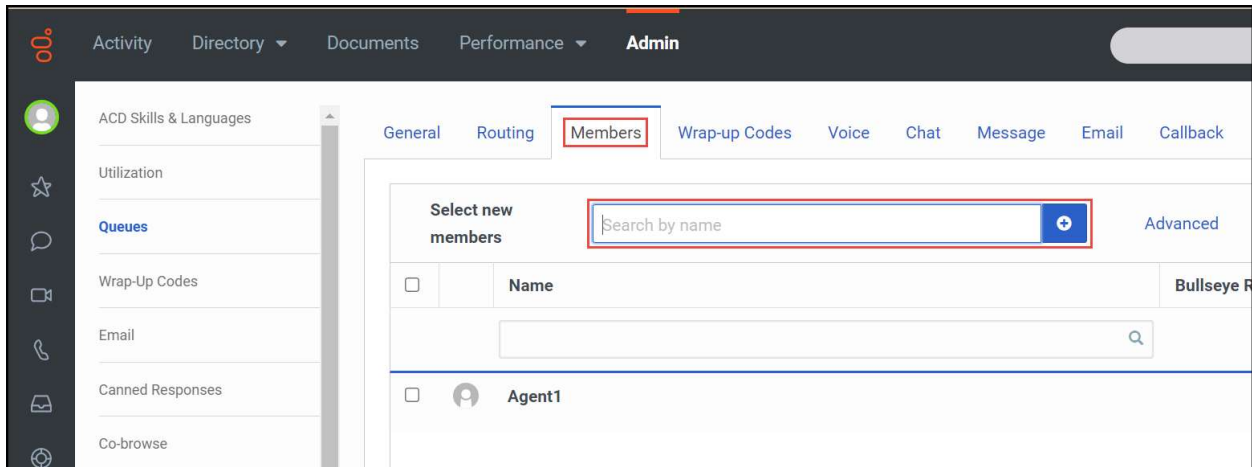


Figure 16: Create Queue Contd.,

## 4.2.7 Create Call Flow

1. Click on **Admin**, Under **Architect** choose **Architect**.
2. The Genesys Cloud architect page will be loaded, provide credentials if needed.
3. Click **Add** to create Inbound call flow.

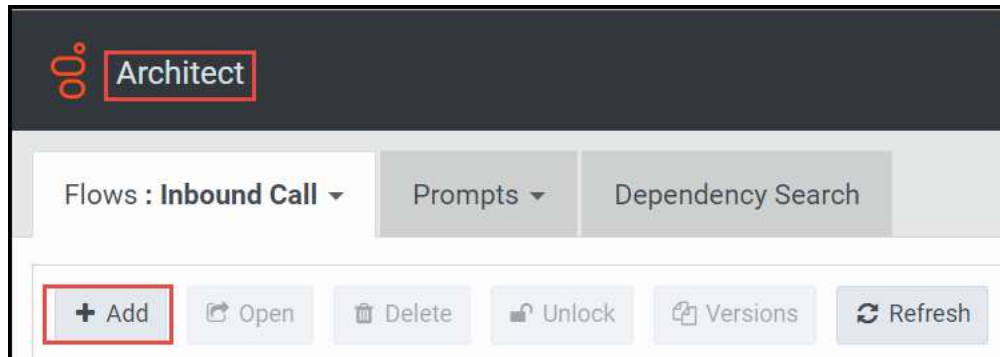


Figure 17: Create Call Flow

4. Provide the Call flow **Name, Language, Division** and click **Create Flow**.
5. Click the above created Call flow and design the callflow as per your need.
6. In the design, transfer the call to ACD Queue to route the call to the skilled Agents.

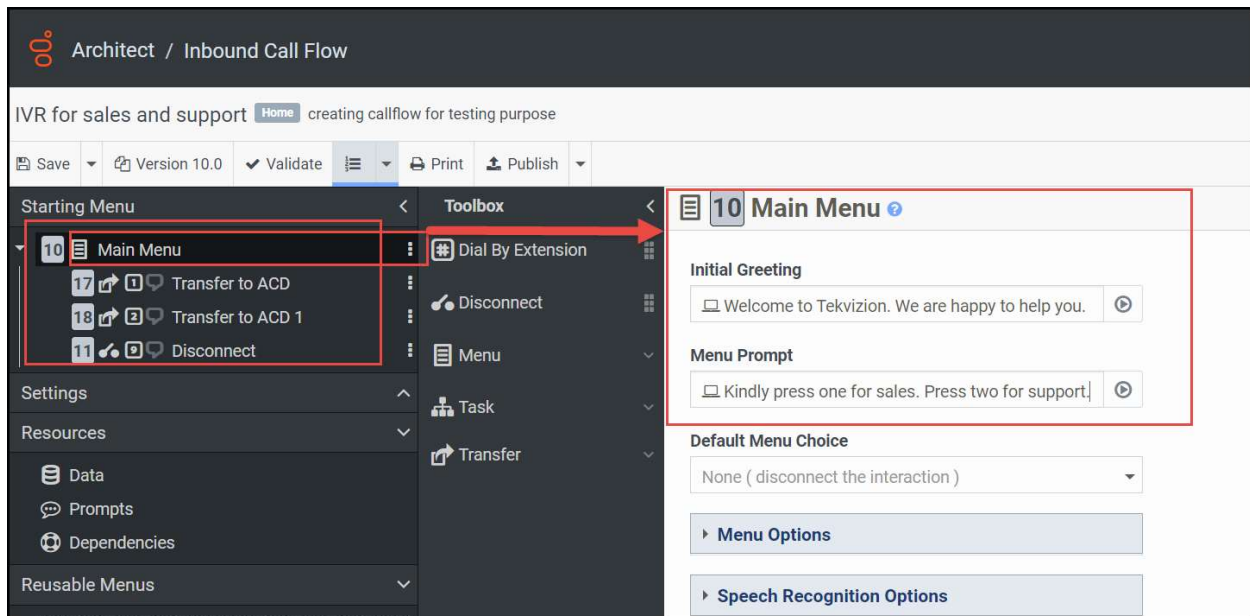


Figure 18: Create Call Flow Contd.,

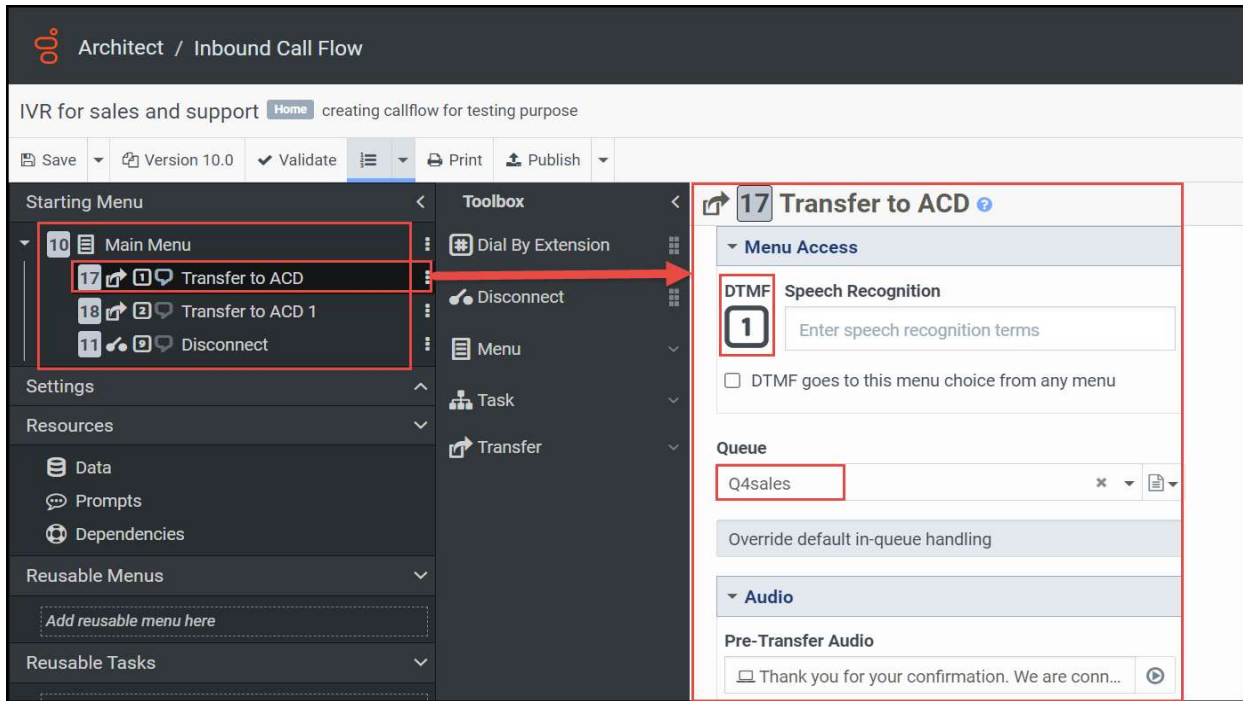


Figure 19: Create Call Flow Contd.,

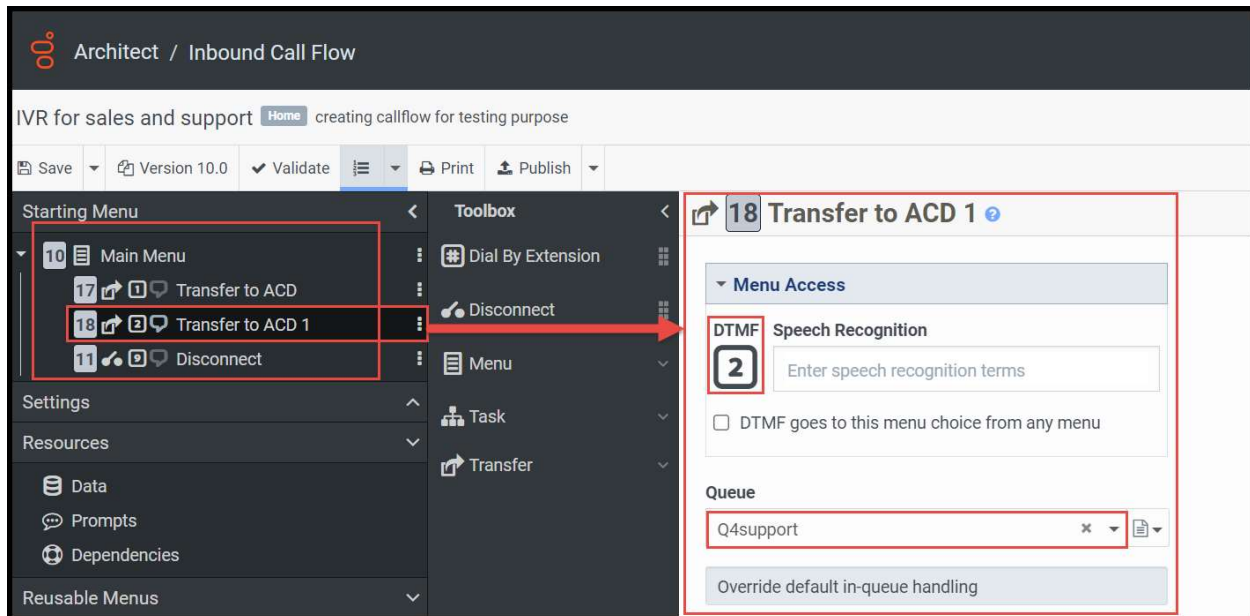


Figure 20: Create Call Flow Contd.,

- After completing the IVR design **click Save, Validate and Publish** to publish the call flow.

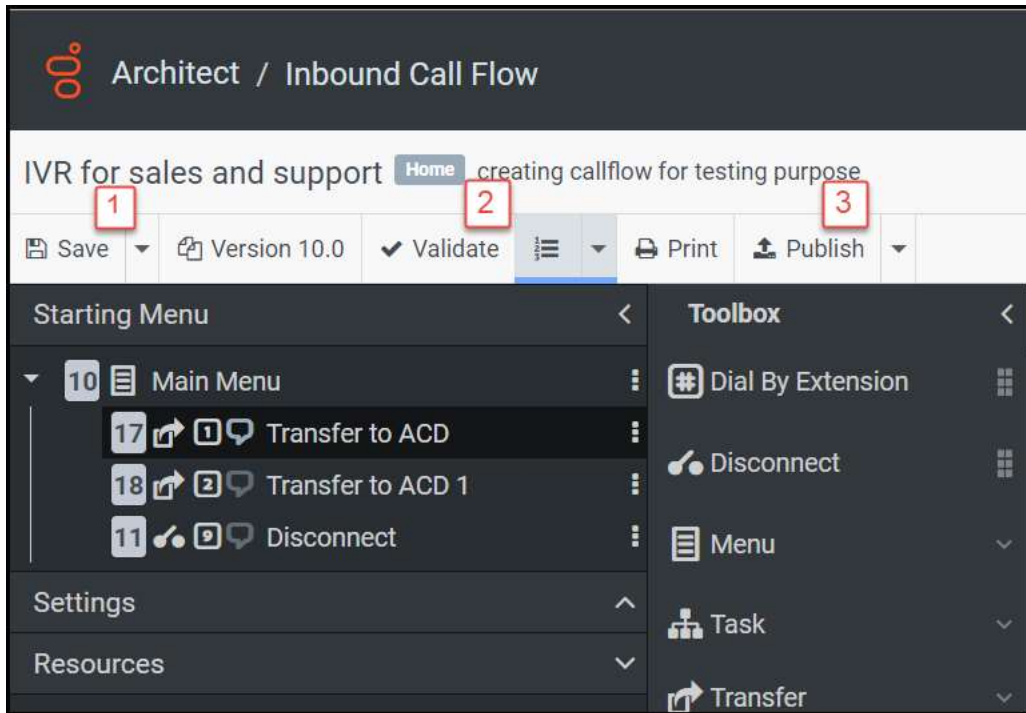


Figure 21: Create Call Flow Contd.,

#### 4.2.8 Create Call Routing

1. Click on **Admin**, Under **Routing** choose **Call Routing**
2. Click **Add Call Route**

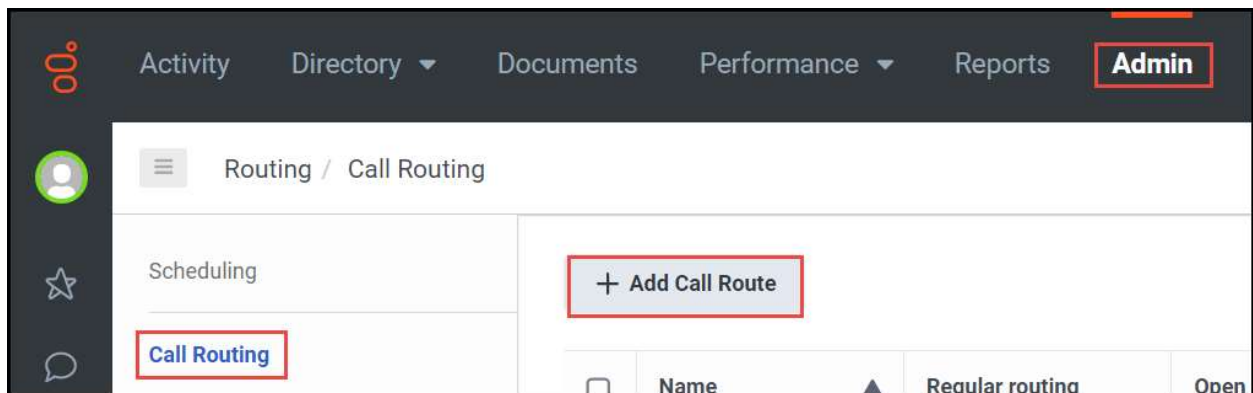


Figure 22: Create Call Routing

3. Provide Call Route **Name, Division**.
4. Add the inbound call flow to **Regular Routing**.
5. Add the DID from the DID number range.

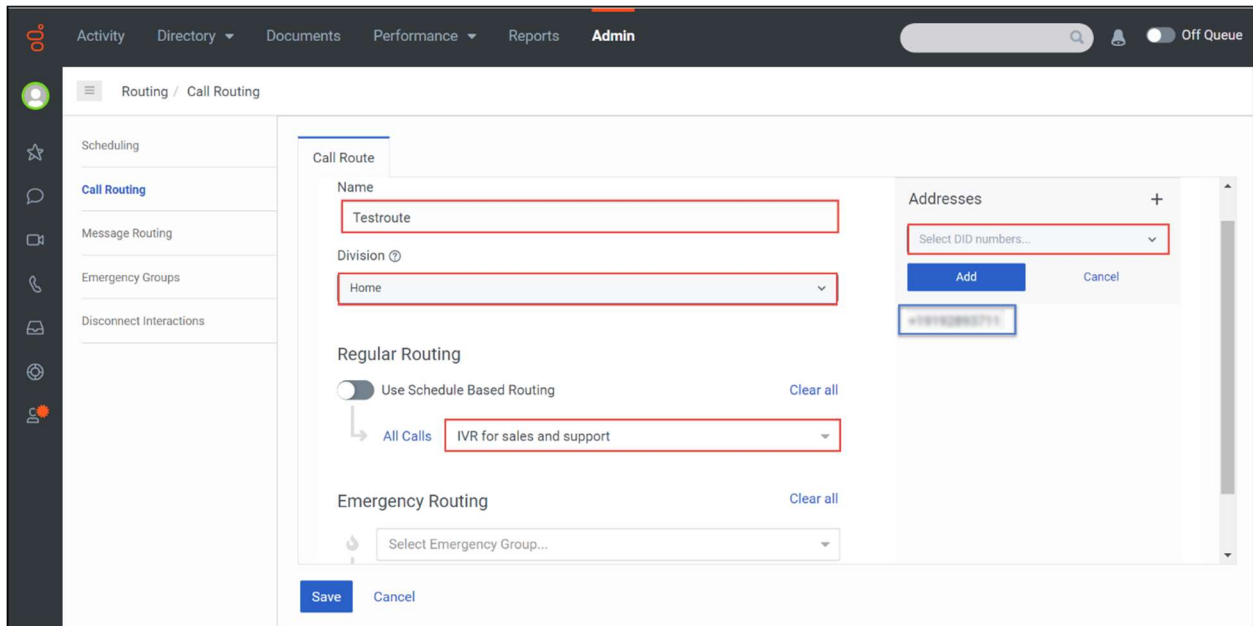


Figure 23: Create Call Flow Contd.,

#### 4.2.9 Adding Trunk to Edge

1. Click on **Admin**, Under Telephony choose **Edges**

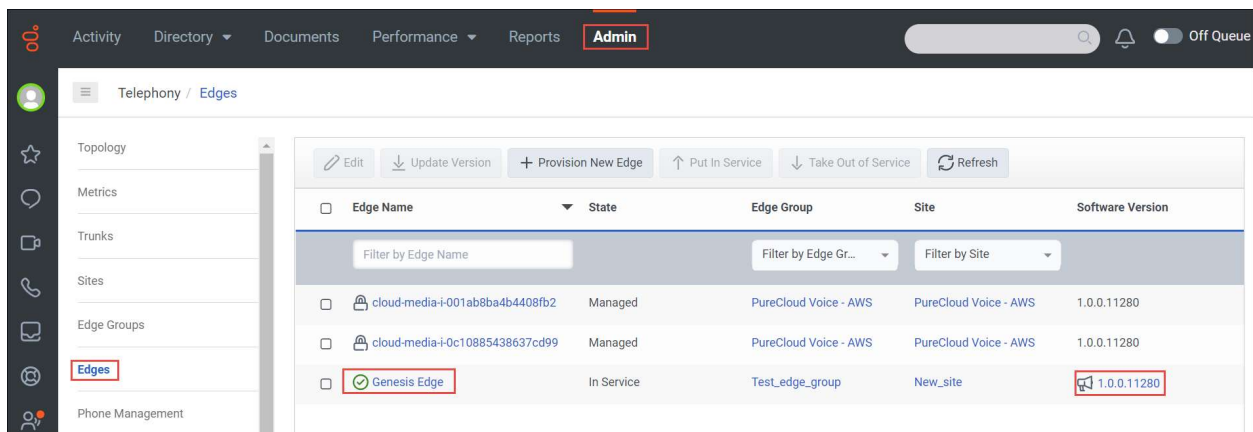


Figure 24: Genesys Cloud Edge Server Version

2. Click on the Edge which is going to be use.
3. In **General** tab, select the site name.

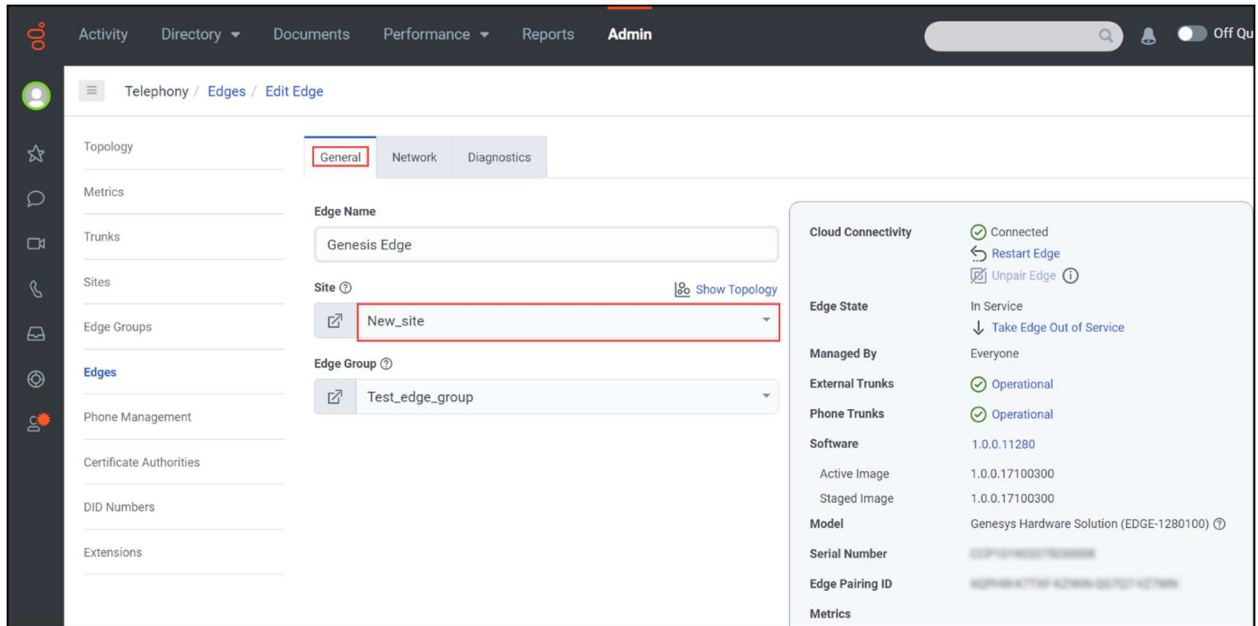


Figure 25: Edge General Info

4. In **Network** tab, add the created **External SIP trunk**.

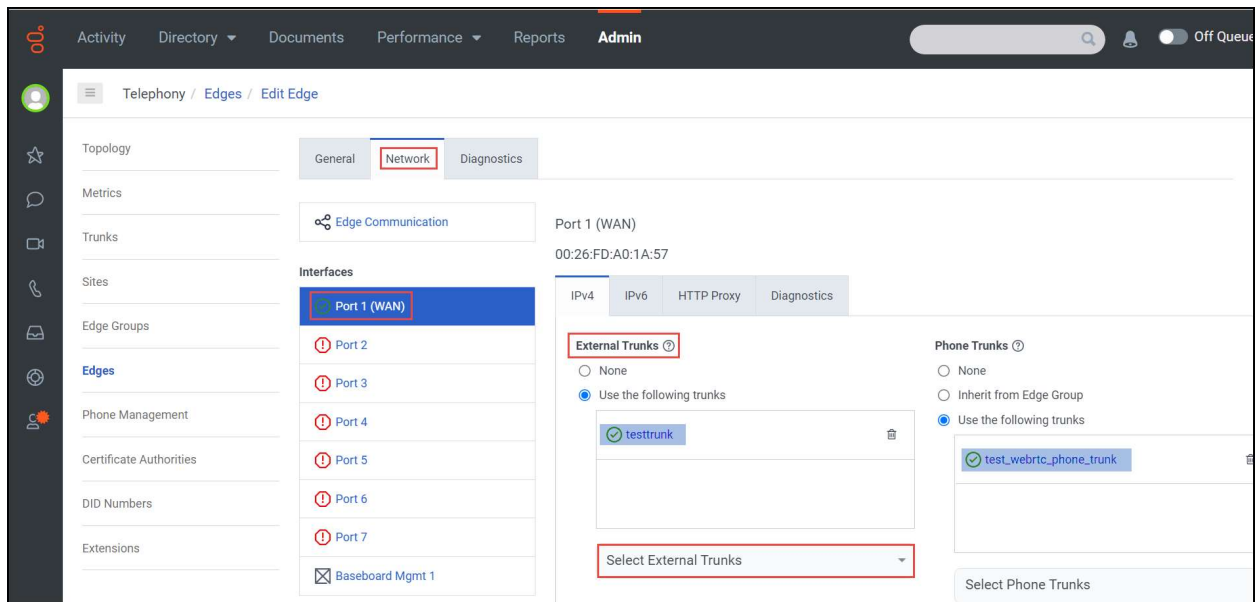
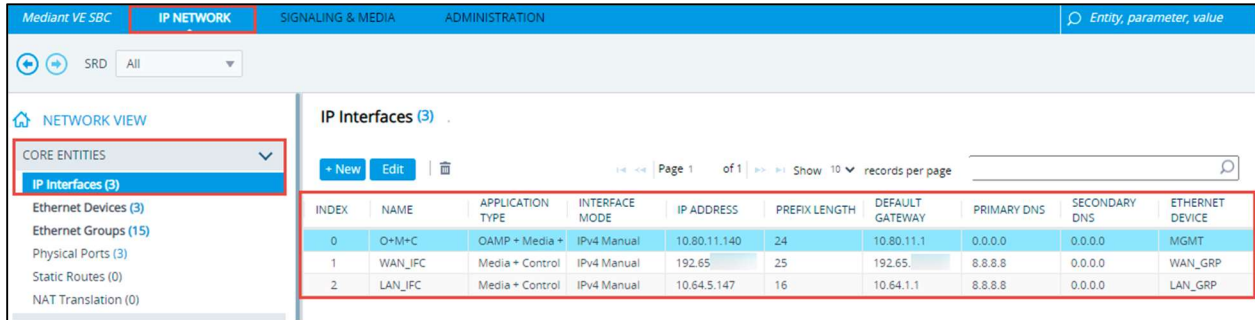


Figure 26: Adding Trunk to Edge

## 4.3 AudioCodes SBC configuration

### 4.3.1 Network IP Interface configuration

Navigate to **SETUP, IP NETWORK** and expand **CORE ENTITIES**. Click **IP Interfaces** and the below figure shows the interfaces that are been used.



The screenshot shows the 'IP NETWORK' section of the Mediant VE SBC configuration. The 'CORE ENTITIES' menu is expanded to show 'IP Interfaces (3)'. The main area displays a table of IP Interfaces with the following data:

INDEX	NAME	APPLICATION TYPE	INTERFACE MODE	IP ADDRESS	PREFIX LENGTH	DEFAULT GATEWAY	PRIMARY DNS	SECONDARY DNS	ETHERNET DEVICE
0	O+M+C	OAMP + Media +	IPv4 Manual	10.80.11.140	24	10.80.11.1	0.0.0.0	0.0.0.0	MGMT
1	WAN_IFC	Media + Control	IPv4 Manual	192.65	25	192.65	8.8.8.8	0.0.0.0	WAN_GRP
2	LAN_IFC	Media + Control	IPv4 Manual	10.64.5.147	16	10.64.1.1	8.8.8.8	0.0.0.0	LAN_GRP

Figure 27: IP Interfaces

### 4.3.2 Media Realm configuration

Two media realms are created, one is associated to Genesys Cloud & Amazon Chime Voice Connector for SIPREC (Voice Recording). Another is associated with Amazon Chime Voice Connector for SIP Trunk. To configure media realm, navigate to **SETUP** and select **SIGNALING & MEDIA**. Expand **CORE ENTITIES** and select **Media Realms**.



The screenshot shows the 'SIGNALING & MEDIA' section of the Mediant VE SBC configuration. The 'CORE ENTITIES' menu is expanded to show 'Media Realms (2)'. The main area displays a table of Media Realms with the following data:

INDEX	NAME	IPv4 INTERFACE NAME	UDP PORT RANGE START	NUMBER OF MEDIA SESSION LEGS	UDP PORT RANGE END	DEFAULT MEDIA REALM
0	MR_Genesys	LAN_IFC	6000	14883	65531	No
1	MR_ACVC	WAN_IFC	6000	14883	65531	No

Figure 28: Media Realms Table

Click **+New** button to add new media realms. Enter the **name** of the Media Realm, **Port Range Start value** and **Number of Media Session Legs**. Select the appropriate **IPv4 Interface Name** for Genesys Cloud & Amazon Chime Voice Connector(SIPREC)

GENERAL		QUALITY OF EXPERIENCE	
Index	0	QoE Profile	-- View
Name	MR_Genesys	Bandwidth Profile	-- View
Topology Location	Down		
IPv4 Interface Name	#2 [LAN_IFC] View		
UDP Port Range Start	6000		
Number Of Media Session Legs	14883		
UDP Port Range End	65531		
TCP Port Range Start	0		
TCP Port Range End	0		
Default Media Realm	No		
Used By Routing Server	Not Used		

Figure 29: Media Realm for Genesys Cloud

Enter the **name** of the Media Realm, **Port Range Start value** and **Number of Media Session Legs**. Select the appropriate **IPv4 Interface Name** for Amazon Chime Voice Connector.

GENERAL		QUALITY OF EXPERIENCE	
Index	1	QoE Profile	-- View
Name	MRWan	Bandwidth Profile	-- View
Topology Location	Up		
IPv4 Interface Name	#1 [WAN-ITSP] View		
UDP Port Range Start	7000		
Number Of Media Session Legs	100		
UDP Port Range End	7999		
TCP Port Range Start	0		
TCP Port Range End	0		
Default Media Realm	No		
Used By Routing Server	Not Used		

Figure 30: Media Realm for Amazon Chime Voice Connector (SIP Trunk)

### 4.3.3 SRD Configuration

To configure SRD, navigate to **SETUP** and select **SIGNALING & MEDIA**. Expand **CORE ENTITIES** and select **SRDs**.

INDEX	NAME	SHARING POLICY	SBC OPERATION MODE	SBC ROUTING POLICY	MAX. NUMBER OF REGISTERED USERS	USER SECURITY MODE
0	DefaultSRD (#0)	Shared	B2BUA	Default_SBCRoutingPc	-1	Accept All

Figure 31: Default SRD

The default SRD configuration is used.

GENERAL		REGISTRATION	
Index	0	Max. Number of Registered Users	-1
Name	DefaultSRD	User Security Mode	Accept All
Sharing Policy	Shared	Enable Un-Authenticated Registrations	Enable
SBC Operation Mode	B2BUA		
SBC Routing Policy	#0 [Default_SBCRoutingPolicy]		
Used By Routing Server	Not Used		
Dial Plan	--		
CAC Profile	--		

Figure 32: Default SRD table details

### 4.3.4 SIP Interface configuration

Navigate to **SETUP** and select **SIGNALING & MEDIA**. Expand **CORE ENTITIES** and select **SIP Interfaces**. Two SIP Interfaces are created, one is for Genesys Cloud & Amazon Chime Voice Connector(SIPREC), and other is for Amazon Chime Voice Connector(SIP Trunk).

INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULATION PROTOCOL	MEDIA REALM
0	SI_Genesys	DefaultSRD	LAN_IFC	SBC	5060	5060	5061	No encapsulation	MR_Genesys
1	SI_ACVC	DefaultSRD	WAN_IFC	SBC	5060	5060	5061	No encapsulation	MR_ACVC

Figure 33: SIP Interfaces

**Network Interface, Media Realm, Application Type, SRD** and **Port numbers** are associated to Genesys Cloud Interface and the remaining parameters are set to default.

SRD: #0 [DefaultSRD]

**GENERAL**

Index: 0

Name: SI\_Genesys

Topology Location: Down

Network Interface: #2 [LAN\_IFC] [View](#)

Application Type: SBC

UDP Port: 5060

TCP Port: 5060

TLS Port: 5061

SCTP Port: 0

SCTP Secondary Network Interface: -- [View](#)

**MEDIA**

Media Realm: #0 [MR\_Genesys] [View](#)

Direct Media: Disable

**SECURITY**

TLS Context Name: #0 [default] [View](#)

TLS Mutual Authentication: --

Message Policy: -- [View](#)

User Security Mode: Not Configured

Enable Un-Authenticated Registrations: Not configured

Max. Number of Registered Users: -1

Figure 34: SIP Interface for Genesys Cloud

SCTP Port	0	Enable Un-Authenticated Registrations	Not configured
SCTP Secondary Network Interface	--	View	Max. Number of Registered Users
Additional UDP Ports			-1
Additional UDP Ports Mode	Always Open		
Encapsulating Protocol	No encapsulation		
Enable TCP Keepalive	Disable		
Used By Routing Server	Not Used		
Pre-Parsing Manipulation Set	--	View	
CAC Profile	--	View	
<b>CLASSIFICATION</b>			
Classification Failure Response Type	500		
Pre-classification Manipulation Set ID	-1		
Call Setup Rules Set ID	-1		
		Cancel	APPLY

Figure 35: SIP Interface for Genesys Cloud Contd.,

**Network Interface, Media Realm, Application Type, SRD and Port numbers** are associated to Amazon Chime Voice Connector SIP Interface and the remaining parameters are set to default.

SIP Interfaces [SL\_ACVC]

SRD #0 [DefaultSRD]

GENERAL	MEDIA	SECURITY
Index	Media Realm	TLS Context Name
1	#1 [MR_ACVC]	#0 [default]
Name	Direct Media	TLS Mutual Authentication
SL_ACVC	Disable	
Topology Location		Message Policy
Down		--
Network Interface		User Security Mode
#1 [WAN_IFC]		Not Configured
Application Type		Enable Un-Authenticated Registrations
SBC		Not configured
UDP Port		Max. Number of Registered Users
5060		-1
TCP Port		
5060		
TLS Port		
5061		
SCTP Port		
0		
SCTP Secondary Network Interface		
--		

Figure 36: SIP Interface for Amazon Chime Voice Connector (SIP Trunk)

Figure 37: SIP Interface for Amazon Chime Voice Connector (SIP Trunk) Contd.,

### 4.3.5 Proxy Sets configuration

Navigate to **SETUP** and select **SIGNALING & MEDIA**. Expand **CORE ENTITIES** and select **Proxy Sets**. Destination address or FQDN is configured in Proxy Sets. Three Proxy Sets are created, one for Genesys Cloud, Amazon Chime Voice Connector (SIPREC) & Amazon Chime Voice Connector(SIP Trunk).

INDEX	NAME	SRD	SBC IPV4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP
0	PS_Genesys	DefaultSRD (#0)	SI_Genesys	60		Disable
1	PS_ACVC	DefaultSRD (#0)	SI_ACVC	60		Disable
2	PS_AWS_SIPREC	DefaultSRD (#0)	SI_Genesys	60		Disable

Figure 38: Proxy Sets table

Select SRD, **SBC IPv4 SIP Interface** and leave the rest to default for Genesys Cloud Proxy Set.

Proxy Sets [PS\_Genesys]

SRD #0 [DefaultSRD]

**GENERAL**

Index 0

Name PS\_Genesys

SBC IPv4 SIP Interface #0 [SI\_Genesys] View

TLS Context Name -- View

**REDUNDANCY**

Redundancy Mode

Proxy Hot Swap Disable

Proxy Load Balancing Method Disable

Min. Active Servers for Load Balancing 1

**KEEP ALIVE**

Proxy Keep-Alive Disable

Proxy Keep-Alive Time [sec] 60

Keep-Alive Failure Responses

Success Detection Retries 1

Success Detection Interval 10

**ADVANCED**

Classification Input IP Address only

DNS Resolve Method

Accept DHCP Proxy List Disable

Cancel APPLY

Figure 39: Proxy Set table for Genesys Cloud

Click on **Proxy Address 1 items** link in bottom to add **Proxy Address** and **Transport Type**.

Proxy Address

**GENERAL**

Index 0

Proxy Address

Transport Type UDP

Proxy Priority 0

Proxy Random Weight 0

Cancel APPLY

Figure 40 Proxy Address for Genesys Cloud

Select **SRD**, **SBC IPv4 SIP Interface** and leave the rest to default for Amazon Chime Voice Connector (SIP Trunk) Proxy Set.

The screenshot shows the 'Proxy Sets [PS\_ACVC]' configuration window. At the top, the SRD dropdown is set to '#0 [DefaultSRD]'. The 'GENERAL' tab is active, showing the following settings:

- Index: 1
- Name: PS\_ACVC
- SBC IPv4 SIP Interface: #1 [SI\_ACVC] (highlighted with a red box)
- TLS Context Name: --

The 'REDUNDANCY' tab shows:

- Redundancy Mode: (empty)
- Proxy Hot Swap: Disable
- Proxy Load Balancing Method: Disable
- Min. Active Servers for Load Balancing: 1

The 'KEEP ALIVE' tab shows:

- Proxy Keep-Alive: Disable
- Proxy Keep-Alive Time [sec]: 60
- Keep-Alive Failure Responses: (empty)
- Success Detection Retries: 1
- Success Detection Interval: 10

The 'ADVANCED' tab shows:

- Classification Input: IP Address only
- DNS Resolve Method: (empty)
- Accept DHCP Proxy List: Disable

Buttons for 'Cancel' and 'APPLY' are at the bottom.

Figure 41: Proxy Set Table for Amazon Chime Voice Connector (SIP Trunk)

Click on **Proxy Address 1 items** link in bottom to add **Proxy Address** and **Transport Type**.

The screenshot shows the 'Proxy Address' configuration window. The 'GENERAL' tab is active, showing the following settings:

- Index: 0
- Proxy Address: gd (highlighted with a red box)
- Transport Type: UDP (highlighted with a red box)
- Proxy Priority: 0
- Proxy Random Weight: 0

Buttons for 'Cancel' and 'APPLY' are at the bottom.

Figure 42: Proxy Address for Amazon Chime Voice Connector (SIP Trunk)

Select **SRD**, **SBC IPv4 SIP Interface** and leave the rest to default for Amazon Chime Voice Connector (SIPREC) Proxy Set.

Proxy Sets [PS\_AWS\_SIPREC]  
 IP: 10.80.11.140  
 Version: 7.40A.250.004

SRD: #0 [DefaultSRD]

**GENERAL**

Index: 2  
 Name: PS\_AWS\_SIPREC  
 SBC IPv4 SIP Interface: #0 [SI\_Genesisys] [View](#)  
 TLS Context Name: -- [View](#)

**REDUNDANCY**

Redundancy Mode: [Dropdown]  
 Proxy Hot Swap: Disable  
 Proxy Load Balancing Method: Disable  
 Min. Active Servers for Load Balancing: 1

**KEEP ALIVE**

Proxy Keep-Alive: Disable  
 Proxy Keep-Alive Time [sec]: 60  
 Keep-Alive Failure Responses: [Text Field]  
 Success Detection Retries: 1  
 Success Detection Interval: 10

**ADVANCED**

Classification Input: IP Address only  
 DNS Resolve Method: [Dropdown]  
 Accept DHCP Proxy List: Disable

Cancel APPLY

Figure 43: Proxy Set Table for Amazon Chime Voice Connector (SIPREC)

Click on **Proxy Address 1 items** link in bottom to add **Proxy Address** and **Transport Type**.

Proxy Address

**GENERAL**

Index: 0  
 Proxy Address: gd [Redacted]  
 Transport Type: UDP  
 Proxy Priority: 0  
 Proxy Random Weight: 0

Cancel APPLY

Figure 44: Proxy Address for Amazon Chime Voice Connector (SIPREC) Contd.,

### 4.3.6 IP Group Table Configuration

Navigate to **SETUP** and select **SIGNALING & MEDIA**. Expand **CORE ENTITIES** and select **IP Groups**. IP Groups are configured for denoting source and destination in IP-to-IP routing rules. IP Groups created for Genesys Cloud, Amazon Chime Voice Connector (SIPREC) and Amazon Chime Voice Connector (SIP Trunk).

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	IPG_Genesys	DefaultS	Server	Not Configu	PS_Genesys	IPProf_Geni	MR_Genesys		Enable	-1	11
1	IPG_ACVC	DefaultS	Server	Not Configu	PS_ACVC	IPProf_ACVC	MR_ACVC		Enable	-1	10
2	IPG_AWS_SI	DefaultS	Server	Not Configu	PS_AWS_SIF	IPProf_AWS	MR_Genesys		Enable	-1	12

Figure 45: IP Group Table

Enter the name of the IP Groups for Genesys Cloud and associate **Proxy Set, IP Profile, Media Realm, SRD** and the remaining parameters are set to default.

IP: 10.80.11.140  
Version: 7.40A.250.004

SRD: #0 [DefaultSRD]

**GENERAL**

Index: 0

Name: IPG\_Genesys

Topology Location: Down

Type: Server

Proxy Set: #0 [PS\_Genesys]

IP Profile: #0 [IPProf\_Genesys]

Media Realm: #0 [MR\_Genesys]

Internal Media Realm: --

Contact User:

SIP Group Name:

**QUALITY OF EXPERIENCE**

QoE Profile: --

Bandwidth Profile: --

User Voice Quality Report: Disable

**MESSAGE MANIPULATION**

Inbound Message Manipulation Set: -1

Outbound Message Manipulation Set: 11

Message Manipulation User-Defined String 1:

Message Manipulation User-Defined String 2:

Proxy Keep-Alive using IP Group settings: Disable

Figure 46: IP Group Table for Genesys Cloud

Internal Media Realm	--	<a href="#">View</a>	Message Manipulation User-Defined String 1	<input type="text"/>
Contact User	<input type="text"/>		Message Manipulation User-Defined String 2	<input type="text"/>
<b>SIP Group Name</b>	<input type="text"/>		Proxy Keep-Alive using IP Group settings	Disable
Created By Routing Server	No		<b>SBC REGISTRATION AND AUTHENTICATION</b>	
Used By Routing Server	Not Used		Max. Number of Registered Users	-1
Proxy Set Connectivity	NA		Registration Mode	User Initiates Registration
<b>SBC GENERAL</b>			User Stickiness	Disable
Classify By Proxy Set	Enable		User UDP Port Assignment	Disable
<b>Validate Source IP</b>	Disable		Authentication Mode	User Authenticates
SBC Operation Mode	Not Configured		Authentication Method List	<input type="text"/>
SBC Client Forking Mode	Sequential		SBC Server Authentication Type	According to Global Parameter
CAC Profile	--	<a href="#">View</a>	OAuth HTTP Service	--
				<a href="#">View</a>

Figure 47: IP Group table for Genesys Cloud Contd.,

CAC Profile	--	<a href="#">View</a>	OAuth HTTP Service	--	<a href="#">View</a>
SIP Source Host Name	<input type="text"/>		Username As Client	• Admin	<input type="text"/>
<b>ADVANCED</b>			Password As Client	• .	<input type="text"/>
Local Host Name	<input type="text"/>		Username As Server	<input type="text"/>	
UUI Format	Disable		Password As Server	<input type="text"/>	
Always Use Src Address	No		<b>GW GROUP STATUS</b>		
<b>SBC ADVANCED</b>			GW Group Registered IP Address	<input type="text"/>	
Source URI Input	<input type="text"/>		GW Group Registered Status	Not Registered	
Destination URI Input	<input type="text"/>				
SIP Connect	No				
SBC PSAP Mode	Disable				

Figure 48: IP Group table for Genesys Cloud Contd.,

SIP Connect	No	▼
SBC PSAP Mode	Disable	▼
Route Using Request URI Port	Disable	▼
Media TLS Context	#0 [default]	▼ <a href="#">View</a>
Keep Original Call-ID	No	▼
Dial Plan	--	▼ <a href="#">View</a>
Call Setup Rules Set ID	-1	
Tags		
SBC Alternative Routing Reasons Set	--	▼ <a href="#">View</a>
Teams Local Media Optimization Handling	None	▼
Teams Local Media Optimization Initial Behavior	DirectMedia	▼
Teams Local Media Optimization Site		
Teams Direct Routing Mode	Disable	▼

Cancel [APPLY](#)

Figure 49: IP Group table for Genesys Cloud Contd.,

Enter the name of the IP Groups for Amazon Chime Voice Connector (SIP Trunk) and associate **SRD, Proxy Set, IP Profile, Media Realm** and the remaining parameters are set to default.

IP Groups [IPG\_ACVC]

SRD #0 [DefaultSRD] ▼

GENERAL	QUALITY OF EXPERIENCE
Index: 1	QoE Profile: -- ▼ <a href="#">View</a>
Name: IPG_ACVC	Bandwidth Profile: -- ▼ <a href="#">View</a>
Topology Location: Down ▼	User Voice Quality Report: Disable ▼
Type: Server ▼	
Proxy Set: #1 [PS_ACVC] ▼ <a href="#">View</a>	
IP Profile: #1 [IPProf_ACVC] ▼ <a href="#">View</a>	
Media Realm: #1 [MR_ACVC] ▼ <a href="#">View</a>	
Internal Media Realm: -- ▼ <a href="#">View</a>	
Contact User:	
SIP Group Name:	
Created By Routing Server: N/A	
	MESSAGE MANIPULATION
	Inbound Message Manipulation Set: -1
	Outbound Message Manipulation Set: 10
	Message Manipulation User-Defined String 1:
	Message Manipulation User-Defined String 2:
	Proxy Keep-Alive using IP Group settings: Disable ▼

Figure 50: IP Group Table for Amazon Chime Voice Connector (SIP Trunk)

SIP Group Name	<input type="text"/>	Proxy Keep-Alive using IP Group settings	Disable
Created By Routing Server	No	<b>SBC REGISTRATION AND AUTHENTICATION</b>	
Used By Routing Server	Not Used	Max. Number of Registered Users	-1
Proxy Set Connectivity	NA	Registration Mode	User Initiates Registration
<b>SBC GENERAL</b>		User Stickiness	Disable
Classify By Proxy Set	Enable	User UDP Port Assignment	Disable
Validate Source IP	Disable	Authentication Mode	User Authenticates
SBC Operation Mode	Not Configured	Authentication Method List	<input type="text"/>
SBC Client Forking Mode	Sequential	SBC Server Authentication Type	According to Global Parameter
CAC Profile	-- <a href="#">View</a>	OAuth HTTP Service	-- <a href="#">View</a>
SIP Source Host Name	<input type="text"/>	Username As Client	• Admin
<b>ADVANCED</b>		Password As Client	• .
		Username As Server	<input type="text"/>
		Password As Server	<input type="text"/>

Figure 51: IP Group Table for Amazon Chime Voice Connector (SIP Trunk) Contd.,

<b>ADVANCED</b>		Local Host Name	<input type="text"/>
UUI Format	Disable	GW Group Registered IP Address	<input type="text"/>
Always Use Src Address	No	GW Group Registered Status	Not Registered
<b>SBC ADVANCED</b>		Source URI Input	<input type="text"/>
Destination URI Input	<input type="text"/>	SIP Connect	No
SBC PSAP Mode	Disable	Route Using Request URI Port	Disable
Media TLS Context	<input type="text"/>		

Figure 52: IP Group Table for Amazon Chime Voice Connector (SIP Trunk) Contd.,

SIP Connect	No	▼
SBC PSAP Mode	Disable	▼
Route Using Request URI Port	Disable	▼
Media TLS Context	#0 [default]	▼ <a href="#">View</a>
Keep Original Call-ID	No	▼
Dial Plan	--	▼ <a href="#">View</a>
Call Setup Rules Set ID	-1	
Tags		
SBC Alternative Routing Reasons Set	--	▼ <a href="#">View</a>
Teams Local Media Optimization Handling	None	▼
Teams Local Media Optimization Initial Behavior	DirectMedia	▼
Teams Local Media Optimization Site		
Teams Direct Routing Mode	Disable	▼

Cancel [APPLY](#)

Figure 53: IP Group Table for Amazon Chime Voice Connector (SIP Trunk) Contd.,

Enter the name of the IP Group for Amazon Chime Voice Connector (SIPREC) and associate **SRD**, **Proxy Set**, **IP Profile**, **Media Realm** and the remaining parameters are set to default.

IP Groups [IPG\_AWS\_SIPREC] - x

SRD #0 [DefaultSRD] ▼

GENERAL	QUALITY OF EXPERIENCE
Index: 2	QoE Profile: -- ▼ <a href="#">View</a>
Name: IPG_AWS_SIPREC	Bandwidth Profile: -- ▼ <a href="#">View</a>
Topology Location: Down ▼	User Voice Quality Report: Disable ▼
Type: Server ▼	
Proxy Set: #2 [PS_AWS_SIPREC] ▼ <a href="#">View</a>	
IP Profile: #2 [IPProf_AWS_SIPREC] ▼ <a href="#">View</a>	
Media Realm: #0 [MR_Genesys] ▼ <a href="#">View</a>	
Internal Media Realm: -- ▼ <a href="#">View</a>	
Contact User: <input type="text"/>	
SIP Group Name: <input type="text"/>	
Created By Routing Server: No	
	MESSAGE MANIPULATION
	Inbound Message Manipulation Set: -1
	Outbound Message Manipulation Set: 12
	Message Manipulation User-Defined String 1: <input type="text"/>
	Message Manipulation User-Defined String 2: <input type="text"/>
	Proxy Keep-Alive using IP Group settings: Disable ▼

Figure 54: IP Group Table for Amazon Chime Voice Connector (SIPREC)

SIP Group Name	<input type="text"/>	Proxy Keep-Alive using IP Group settings	Disable
Created By Routing Server	No	<b>SBC REGISTRATION AND AUTHENTICATION</b>	
Used By Routing Server	Not Used	Max. Number of Registered Users	-1
Proxy Set Connectivity	NA	Registration Mode	User Initiates Registration
<b>SBC GENERAL</b>		User Stickiness	Disable
Classify By Proxy Set	Enable	User UDP Port Assignment	Disable
Validate Source IP	Disable	Authentication Mode	User Authenticates
SBC Operation Mode	Not Configured	Authentication Method List	<input type="text"/>
SBC Client Forking Mode	Sequential	SBC Server Authentication Type	According to Global Parameter
CAC Profile	-- <a href="#">View</a>	OAuth HTTP Service	-- <a href="#">View</a>
SIP Source Host Name	<input type="text"/>	Username As Client	Admin
<b>ADVANCED</b>		Password As Client	*
		Username As Server	<input type="text"/>

Figure 55: IP Group Table for Amazon Chime Voice Connector (SIPREC) Contd.,

<b>ADVANCED</b>		Username As Server	<input type="text"/>
Local Host Name	<input type="text"/>	Password As Server	<input type="text"/>
UUI Format	Disable	<b>GW GROUP STATUS</b>	
Always Use Src Address	No	GW Group Registered IP Address	<input type="text"/>
<b>SBC ADVANCED</b>		GW Group Registered Status	Not Registered
Source URI Input	<input type="text"/>		
Destination URI Input	<input type="text"/>		
SIP Connect	No		
SBC PSAP Mode	Disable		
Route Using Request URI Port	Disable		
Media TLS Context	#0 [default] <a href="#">View</a>		
Keep Original Call-ID	No		

Figure 56: IP Group Table for Amazon Chime Voice Connector (SIPREC) Contd.,

SIP Connect	No	▼
SBC PSAP Mode	Disable	▼
Route Using Request URI Port	Disable	▼
Media TLS Context	#0 [default]	▼ <a href="#">View</a>
Keep Original Call-ID	No	▼
Dial Plan	--	▼ <a href="#">View</a>
Call Setup Rules Set ID	-1	
Tags		
SBC Alternative Routing Reasons Set	--	▼ <a href="#">View</a>
Teams Local Media Optimization Handling	None	▼
Teams Local Media Optimization Initial Behavior	DirectMedia	▼
Teams Local Media Optimization Site		
Teams Direct Routing Mode	Disable	▼

Cancel [APPLY](#)

Figure 57: IP Group Table for Amazon Chime Voice Connector (SIPREC) Contd.,

### 4.3.7 Coder Groups configuration

Navigate to **SETUP** and select **SIGNALING & MEDIA**. Expand **CODERS & PROFILES** and select **Coder Groups**. G711 U-law is configured in Coder Groups.

Mediant VE SBC
Entity, parameter, value

IP NETWORK
ADMINISTRATION

SRD
All

TOPOLOGY VIEW

CORE ENTITIES

CODERS & PROFILES

IP Profiles (3)

Coder Settings

Coder Groups

Allowed Audio Coders Groups (0)

Allowed Video Coders Groups (0)

SBC

Classification (0)

Routing

Routing Policies (1)

IP-to-IP Routing (3)

Alternative Reasons Set (0)

IP Group Set (0)

Manipulation

#### Coder Groups

Coder Group Name: 0: AudioCodersGroups\_0 [Delete Group](#)

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression	Coder Specific
G.711U-law	20	64	0	Disabled	
▼	▼	▼	▼	▼	
▼	▼	▼	▼	▼	
▼	▼	▼	▼	▼	
▼	▼	▼	▼	▼	
▼	▼	▼	▼	▼	
▼	▼	▼	▼	▼	
▼	▼	▼	▼	▼	
▼	▼	▼	▼	▼	
▼	▼	▼	▼	▼	

Cancel [APPLY](#)

Figure 58: Coder Groups

### 4.3.8 IP Profile Configuration

Navigate to **SETUP** and select **SIGNALING & MEDIA**. Expand **CODERS & PROFILES** and select **IP Profiles**. Three IP Profiles are created- Genesys Cloud, Amazon Chime Voice Connector (SIPREC) & Amazon Chime Voice Connector(SIP Trunk)

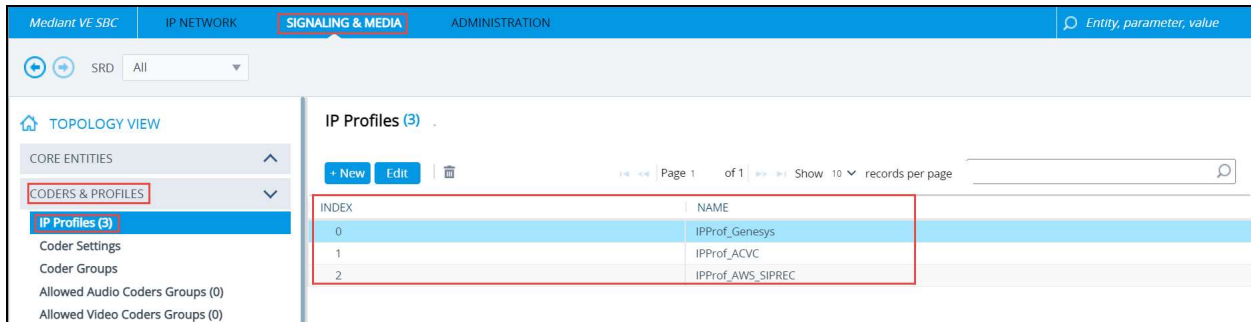


Figure 59: IP Profiles

In the IP Profile for Genesys Cloud enter the **Name** select **Session Expire Mode** as Supported, **SBC Media Security Mode** as Not Secured and **Extension Coder Group** is associated appropriately.

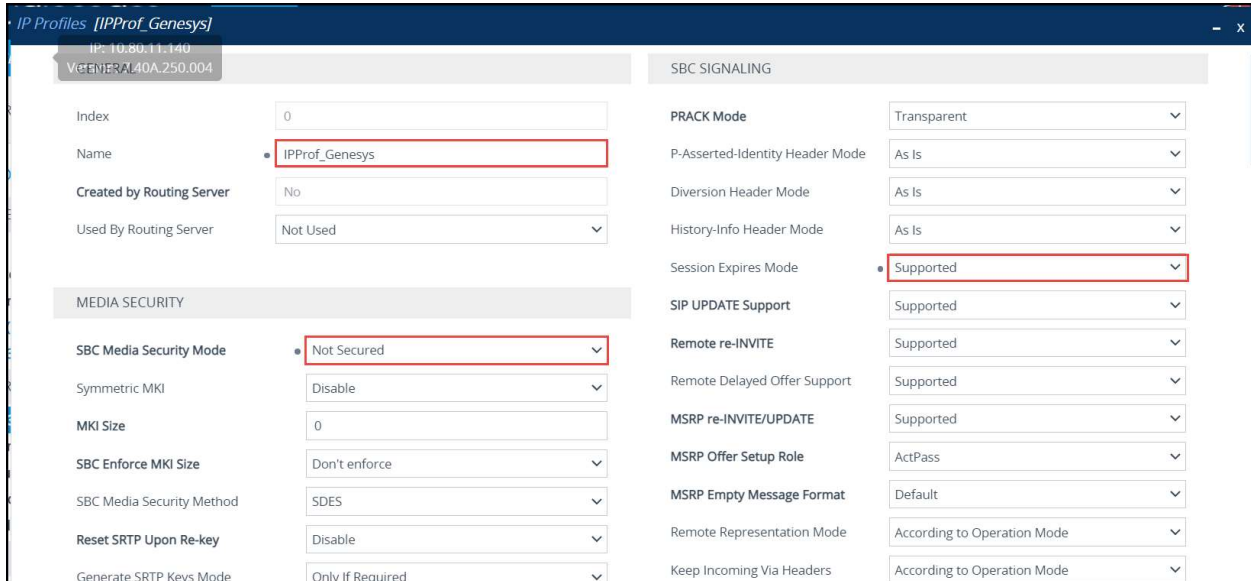


Figure 60: IP Profile for Genesys Cloud

Reset SRTP Upon Re-key	Disable	Remote Representation Mode	According to Operation Mode
Generate SRTP Keys Mode	Only If Required	Keep Incoming Via Headers	According to Operation Mode
SBC Remove Crypto Lifetime in SDP	No	Keep Incoming Routing Headers	According to Operation Mode
SBC Remove Unknown Crypto	No	Keep User-Agent Header	According to Operation Mode
Crypto Suites Group	--	Handle X-Detect	No
Encryption on RTCP Packets	As Is	ISUP Body Handling	Transparent
		ISUP Variant	Itu92
		Max Call Duration [min]	0
<b>SBC EARLY MEDIA</b>			
Remote Early Media	Supported	<b>SBC REGISTRATION</b>	
Remote Multiple 18x	Supported	User Registration Time	0
Remote Early Media Response Type	Transparent	NAT UDP Registration Time	-1
Remote Multiple Early Dialogs	According to Operation Mode	NAT TCP Registration Time	-1
Remote Multiple Answers Mode	Disable		
Remote Early Media RTP Detection Mode	By Signaling		

Figure 61: IP Profile for Genesys Cloud Contd.,

Remote Early Media RTP Detection Mode	By Signaling	<b>SBC FORWARD AND TRANSFER</b>	
Remote RFC 3960 Support	Not Supported	Remote REFER Mode	Regular
Remote Can Play Ringback	Yes	Remote Replaces Mode	Standard
Generate RTP	None	Play RBT To Transferee	No
		Remote 3xx Mode	Transparent
<b>SBC MEDIA</b>			
Mediation Mode	RTP Mediation	<b>SBC HOLD</b>	
Extension Coders Group	#0 [AudioCodersGroups_0]	Remote Hold Format	Transparent
Allowed Audio Coders	--	Reliable Held Tone Source	Yes
Allowed Coders Mode	Restriction	Play Held Tone	No
Allowed Video Coders	--	<b>SBC FAX</b>	
Allowed Media Types		Fax Coders Group	--
Direct Media Tag			
RFC 2833 Mode	As Is		

Figure 62: IP Profile for Genesys Cloud Contd.,

RFC 2833 Mode	As Is	Fax Coders Group	--
RFC 2833 DTMF Payload Type	0	Fax Mode	As Is
Alternative DTMF Method	As Is	Fax Offer Mode	All coders
Send Multiple DTMF Methods	Disable	Fax Answer Mode	Single coder
Receive Multiple DTMF Methods	Disable	Remote Renegotiate on Fax Detection	Transparent
Adapt RFC2833 BW to Voice coder BW	Disabled	Fax Rerouting Mode	Disable
SDP Ptime Answer	Remote Answer	<b>MEDIA</b>	
Preferred PTime	0	Broken Connection Mode	Disconnect
Use Silence Suppression	Transparent	Media IP Version Preference	Only IPv4
RTP Redundancy Mode	As Is	RTP Redundancy Depth	Disable
RTCP Mode	Transparent	<b>LOCAL TONES</b>	
Jitter Compensation	Disable	Local Ringback Tone Index	-1
ICE Mode	Disable		
SDP Handle RTCP	Don't Care		

Figure 63: IP Profile for Genesys Cloud Contd.,

RTCP Mux	Not Supported	Local Ringback Tone Index	-1
RTCP Feedback	Feedback Off	Local Held Tone Index	-1
Re-number MID	Disable		
Voice Quality Enhancement	Disable		
Max Opus Bandwidth	0		
Generate No-Op Packets	Disable		
Enhanced PLC	Disable		
SBC Multiple Coders	Not Supported		
SBC Allow Only Negotiated PT	Disable		
<b>QUALITY OF SERVICE</b>			
RTP IP DiffServ	46		
Signaling DiffServ	24		
Data DiffServ	0		
Cancel		<b>APPLY</b>	

Figure 64: IP Profile for Genesys Cloud Contd.,

In the IP Profile for Amazon Chime Voice Connector(SIP Trunk), enter the **Name**, select **Session Expire Mode** as Supported, **SBC Media Security Mode** as Not Secured and **Extension Coder Group** is associated appropriately.

Figure 65: IP Profiles for Amazon Chime Voice Connector (SIP Trunk)

Figure 66: IP Profiles for Amazon Chime Voice Connector (SIP Trunk) Contd.,

Remote RFC 3960 Support	Not Supported	<b>SBC FORWARD AND TRANSFER</b>	
Remote Can Play Ringback	Yes	<b>Remote REFER Mode</b>	Regular
Generate RTP	None	Remote Replaces Mode	Standard
<b>SBC MEDIA</b>		Play RBT To Transferee	No
Mediation Mode	RTP Mediation	<b>Remote 3xx Mode</b>	Transparent
<b>Extension Coders Group</b>	#0 [AudioCodersGroups_0]	<b>SBC HOLD</b>	
<b>Allowed Audio Coders</b>	-- <a href="#">View</a>	<b>Remote Hold Format</b>	Transparent
<b>Allowed Coders Mode</b>	Restriction	Reliable Held Tone Source	Yes
Allowed Video Coders	-- <a href="#">View</a>	Play Held Tone	No
Allowed Media Types		<b>SBC FAX</b>	
Direct Media Tag		Fax Coders Group	--
<b>RFC 2833 Mode</b>	As Is	Fax Mode	As Is
<b>RFC 2833 DTMF Payload Type</b>	0		

Figure 67: IP Profile for Amazon Chime Voice Connector (SIP Trunk) Contd.,

<b>RFC 2833 Mode</b>	As Is	Fax Coders Group	--
<b>RFC 2833 DTMF Payload Type</b>	0	Fax Mode	As Is
Alternative DTMF Method	As Is	Fax Offer Mode	All coders
Send Multiple DTMF Methods	Disable	Fax Answer Mode	Single coder
Receive Multiple DTMF Methods	Disable	Remote Renegotiate on Fax Detection	Transparent
Adapt RFC2833 BW to Voice coder BW	Disabled	Fax Rerouting Mode	Disable
SDP Ptime Answer	Remote Answer	<b>MEDIA</b>	
Preferred PTime	0	<b>Broken Connection Mode</b>	Disconnect
Use Silence Suppression	Transparent	Media IP Version Preference	Only IPv4
RTP Redundancy Mode	As Is	RTP Redundancy Depth	Disable
RTCP Mode	Transparent	<b>LOCAL TONES</b>	
Jitter Compensation	Disable	Local Ringback Tone Index	-1
ICE Mode	Disable		
SDP Handle RTCP	Don't Care		

Figure 68: IP Profile for Amazon Chime Voice Connector (SIP Trunk) Contd.,

RTCP Mux	Not Supported	Local Held Tone Index	-1
RTCP Feedback	Feedback Off		
Re-number MID	Disable		
Voice Quality Enhancement	Disable		
Max Opus Bandwidth	0		
Generate No-Op Packets	Disable		
Enhanced PLC	Disable		
SBC Multiple Coders	Not Supported		
SBC Allow Only Negotiated PT	Disable		

QUALITY OF SERVICE	
RTP IP DiffServ	46
Signaling DiffServ	24
Data DiffServ	0

Cancel **APPLY**

Figure 69: IP Profiles for Amazon Chime Voice Connector (SIP Trunk) Contd.,

In the IP profile for Amazon Chime Voice Connector (SIPREC) , enter the **Name**, select **Session Expire Mode** as Supported, **SBC Media Security Mode** as Not Secured and **Extension Coder Group** is associated appropriately.

GENERAL		SBC SIGNALING	
Index	2	PRACK Mode	Transparent
Name	IPProf_AWS_SIPREC	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
		Session Expires Mode	Supported
		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
		Keep Incoming Via Headers	According to Operation Mode

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
Generate SRTP Keys Mode	Only If Required

Figure 70: IP Profiles for Amazon Chime Voice Connector (SIPREC)

Generate SRTP Keys Mode	Only If Required	Keep Incoming Via Headers	According to Operation Mode
SBC Remove Crypto Lifetime in SDP	No	Keep Incoming Routing Headers	According to Operation Mode
SBC Remove Unknown Crypto	No	Keep User-Agent Header	According to Operation Mode
Crypto Suites Group	-- <a href="#">View</a>	Handle X-Detect	No
Encryption on RTCP Packets	As Is	ISUP Body Handling	Transparent
<b>SBC EARLY MEDIA</b>		ISUP Variant	Itu92
Remote Early Media	Supported	Max Call Duration [min]	0
Remote Multiple 18x	Supported	<b>SBC REGISTRATION</b>	
Remote Early Media Response Type	Transparent	User Registration Time	0
Remote Multiple Early Dialogs	According to Operation Mode	NAT UDP Registration Time	-1
Remote Multiple Answers Mode	Disable	NAT TCP Registration Time	-1
Remote Early Media RTP Detection Mode	By Signaling	<b>SBC FORWARD AND TRANSFER</b>	
Remote RFC 3960 Support	Not Supported		

Figure 71: IP Profiles for Amazon Chime Voice Connector (SIPREC) Contd.,

Remote RFC 3960 Support	Not Supported	<b>SBC FORWARD AND TRANSFER</b>	
Remote Can Play Ringback	Yes	Remote REFER Mode	Regular
Generate RTP	None	Remote Replaces Mode	Standard
<b>SBC MEDIA</b>		Play RBT To Transferee	No
Mediation Mode	RTP Mediation	Remote 3xx Mode	Transparent
Extension Coders Group	#0 [AudioCodersGroups_0]	<b>SBC HOLD</b>	
Allowed Audio Coders	-- <a href="#">View</a>	Remote Hold Format	Transparent
Allowed Coders Mode	Restriction	Reliable Held Tone Source	Yes
Allowed Video Coders	-- <a href="#">View</a>	Play Held Tone	No
Allowed Media Types		<b>SBC FAX</b>	
Direct Media Tag		Fax Coders Group	--
RFC 2833 Mode	As Is	Fax Mode	As Is
RFC 2833 DTMF Payload Type	0		

Figure 72: IP Profiles for Amazon Chime Voice Connector (SIPREC) Contd.,

RFC 2833 DTMF Payload Type	0	Fax Mode	As Is
Alternative DTMF Method	As Is	Fax Offer Mode	All coders
Send Multiple DTMF Methods	Disable	Fax Answer Mode	Single coder
Receive Multiple DTMF Methods	Disable	Remote Renegotiate on Fax Detection	Transparent
Adapt RFC2833 BW to Voice coder BW	Disabled	Fax Rerouting Mode	Disable
SDP Ptime Answer	Remote Answer	<b>MEDIA</b>	
Preferred PTime	0	Broken Connection Mode	Disconnect
Use Silence Suppression	Transparent	Media IP Version Preference	Only IPv4
RTP Redundancy Mode	As Is	RTP Redundancy Depth	Disable
RTCP Mode	Transparent	<b>LOCAL TONES</b>	
Jitter Compensation	Disable	Local Ringback Tone Index	-1
ICE Mode	Disable	Local Held Tone Index	-1
SDP Handle RTCP	Don't Care		
RTCP Mux	Not Supported		

Figure 73: IP Profiles for Amazon Chime Voice Connector (SIPREC) Contd.,

RTCP Mux	Not Supported	Local Held Tone Index	-1
RTCP Feedback	Feedback Off		
Re-number MID	Disable		
Voice Quality Enhancement	Disable		
Max Opus Bandwidth	0		
Generate No-Op Packets	Disable		
Enhanced PLC	Disable		
SBC Multiple Coders	Not Supported		
SBC Allow Only Negotiated PT	Disable		
<b>QUALITY OF SERVICE</b>			
RTP IP DiffServ	46		
Signaling DiffServ	24		
Data DiffServ	0		
Cancel		<b>APPLY</b>	

Figure 74: IP Profiles for Amazon Chime Voice Connector (SIPREC) Contd.,

### 4.3.9 IP-to-IP Routing

Navigate to **SETUP** and select **SIGNALING & MEDIA**. Expand **SBC** and select **IP-to-IP Routing** under **Routing**. Routing rules are defined for forwarding SIP messages based on IP Groups from source to destination.

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	GenToACVC	Default_SBCR	Route Row	IPG_Genesis	All	*	*	IP Group	IPG_ACVC	--	
1	ACVCToGen	Default_SBCR	Route Row	IPG_ACVC	All	*	*	IP Group	IPG_Genesis	--	
2											

Figure 75: IP-to-IP Routing

In the IP to IP routing from Genesys Cloud to Amazon Chime Voice Connector (SIP Trunk), enter the **Name**, select **Source IP Group**, **Destination Type**, **Destination IP Group** and **Source IP Group** is associated appropriately.

Routing Policy: #0 [Default\_SBCRoutingPolicy]

**GENERAL**

Index: 0

Name: GenToACVC

Alternative Route Options: Route Row

**MATCH**

Source IP Group: #0 [IPG\_Genesis]

Request Type: All

Source Username Pattern: \*

Source Host: \*

Source Tag:

Destination Username Pattern: \*

**ACTION**

Destination Type: IP Group

Destination IP Group: #1 [IPG\_ACVC]

Destination SIP Interface: --

Destination Address:

Destination Port: 0

Destination Transport Type:

IP Group Set: --

Call Setup Rules Set ID: -1

Group Policy: Sequential

Cost Group: --

Routing Tag Name: default

Buttons: Cancel, APPLY

Figure 76: IP-to-IP Routing from Genesys Cloud to Amazon Chime Voice Connector (SIP Trunk)

In the IP to IP routing from Amazon Chime Voice Connector (SIP Trunk) to Genesys Cloud, enter the **Name**, select **Source IP Group**, **Destination Type**, **Destination IP Group** and **Source IP Group** is associated appropriately.

The screenshot shows the configuration for IP-to-IP Routing. The window title is "IP-to-IP Routing [ACVCToGen]". At the top, there is a "Routing Policy" dropdown set to "#0 [Default\_SBCRoutingPolicy]".

**GENERAL**

- Index: 1
- Name: ACVCToGen
- Alternative Route Options: Route Row

**MATCH**

- Source IP Group: #1 [IPG\_ACVC] (highlighted with a red box)
- Request Type: All
- Source Username Pattern: \*
- Source Host: \*
- Source Tag:
- Destination Username Pattern: \*

**ACTION**

- Destination Type: IP Group
- Destination IP Group: #0 [IPG\_Genesys] (highlighted with a red box)
- Destination SIP Interface: --
- Destination Address:
- Destination Port: 0
- Destination Transport Type:
- IP Group Set: --
- Call Setup Rules Set ID: -1
- Group Policy: Sequential
- Cost Group: --
- Routing Tag Name: default

Buttons: Cancel, APPLY

Figure 77: IP-to-IP Routing from Amazon Chime Voice Connector (SIP Trunk) to Genesys Cloud

### 4.3.10 SIPREC Configuration

To configure SIP Recording, navigate to **SETUP** and select **SIGNALING & MEDIA**. Expand **SIP RECORDING** and select **SIP Recording Settings**. Enter the FQDN of Amazon Chime Voice Connector in Recording Server (SRS) Destination Username

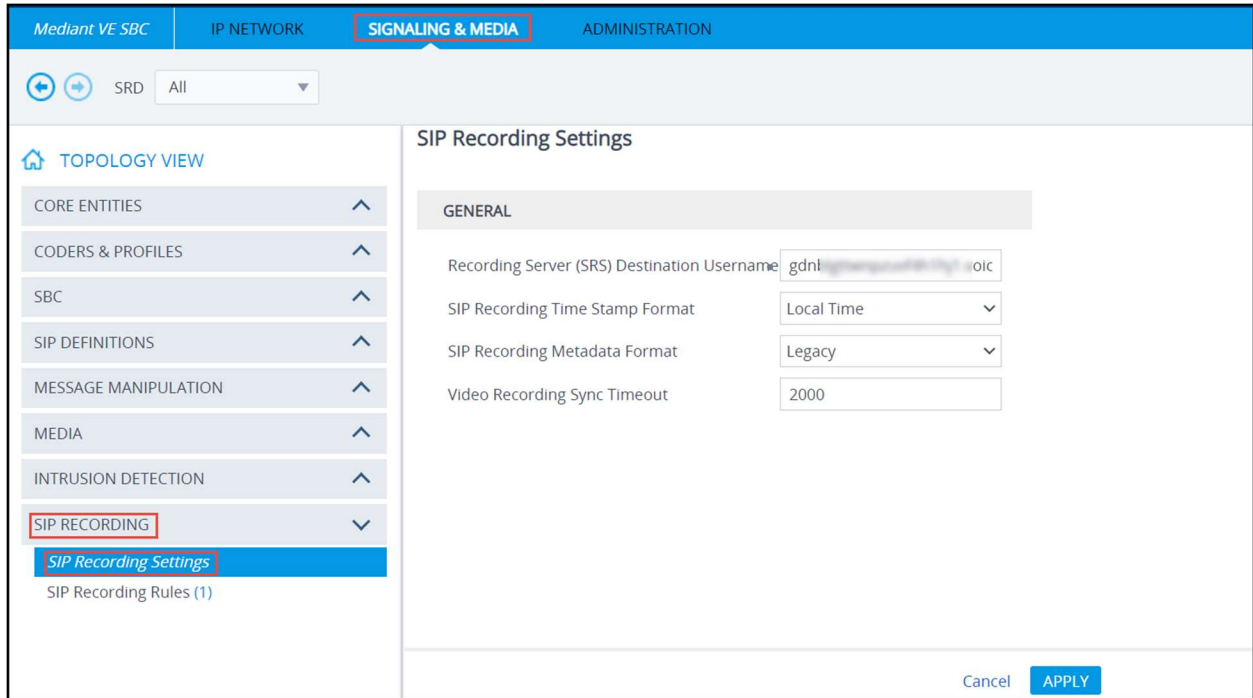


Figure 78: SIPREC Configuration

Navigate to **SIP Recording Rules** in the **SIP Recording** Section and create a new rule. Select **Recorded IP Group**- IP Group created for Genesys Cloud, **Peer IP Group**-IP Group created for Amazon Chime Voice Connector (SIP Trunk) and **Recording Server (SRS) IP Group**- IP Group created for Amazon Chime Voice Connector (SIPREC). Select **Caller** to Both and **Trigger** to Call Connect.

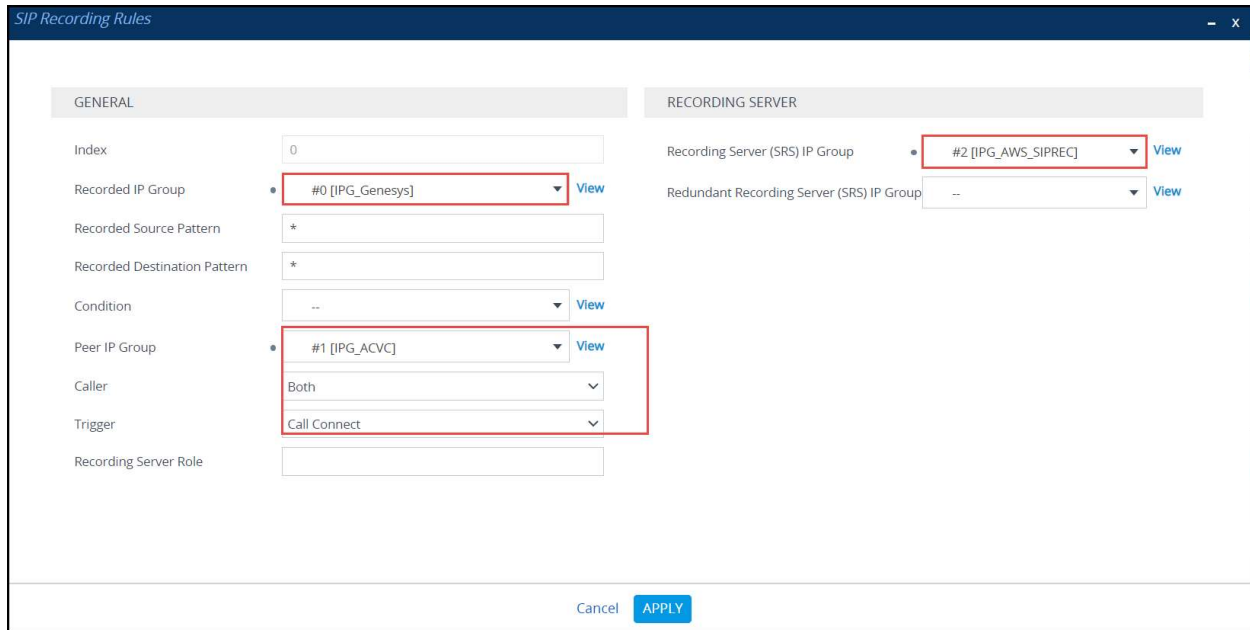


Figure 79: SIPREC Configuration Contd.,

### 4.3.11 TLS Configuration

TLS(SRTP) is configured between,

- Audiocodes SBC and Amazon Chime Voice Connector(SIP Trunk)
- Audiocodes SBC and Amazon Chime Voice Connector(SIPREC)

Navigate to **SETUP** and select **IP NETWORK**. Expand **SECURITY** and click on **TLS Contexts**.

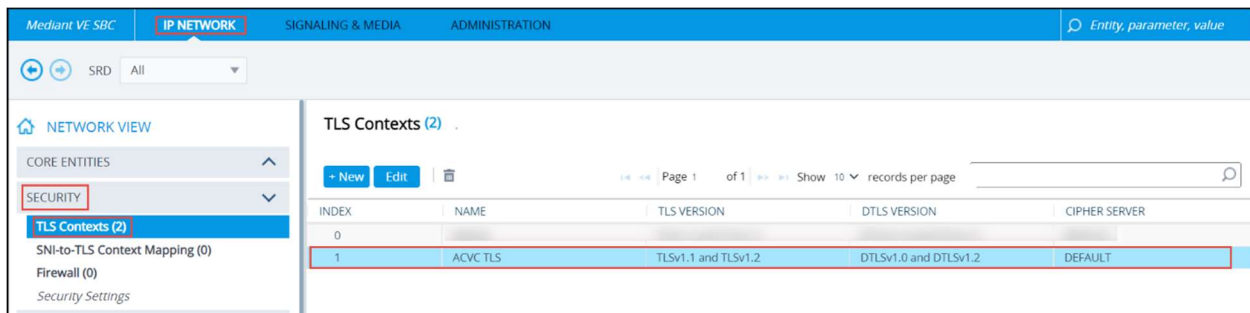


Figure 80: TLS Contexts

In the **TLS Context**, Enter the **Name**, select the **TLS Version** and leave the rest of the values to default

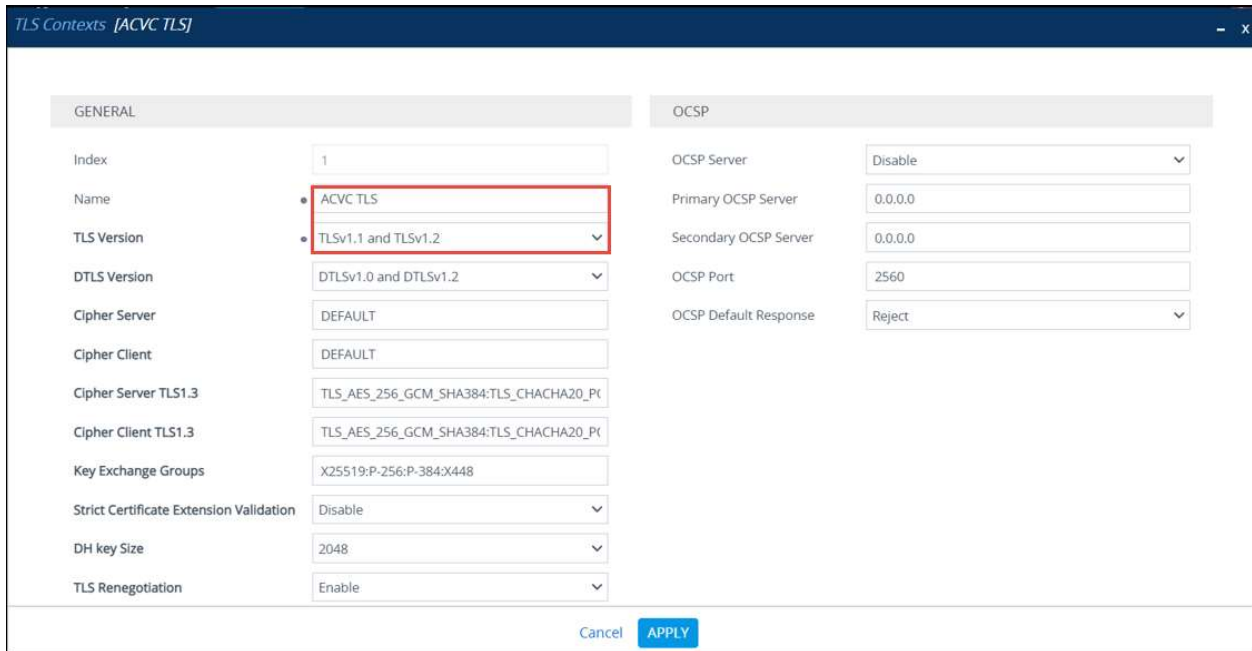


Figure 81: TLS Context for Amazon Chime Voice Connector (SIP Trunk & SIPREC)

Amazon Trust Root Certificate is to be installed in the Trusted Root Certificates list under TLS Context. Amazon Chime Voice Connector Root Certificate can be downloaded from Amazon Chime Voice Connector account. In the TLS Context page, select the **TLS Context** for Amazon Chime Voice Connector and click **Trusted Root Certificates** link located in the bottom. Click on **Import** button and select the certificate file.

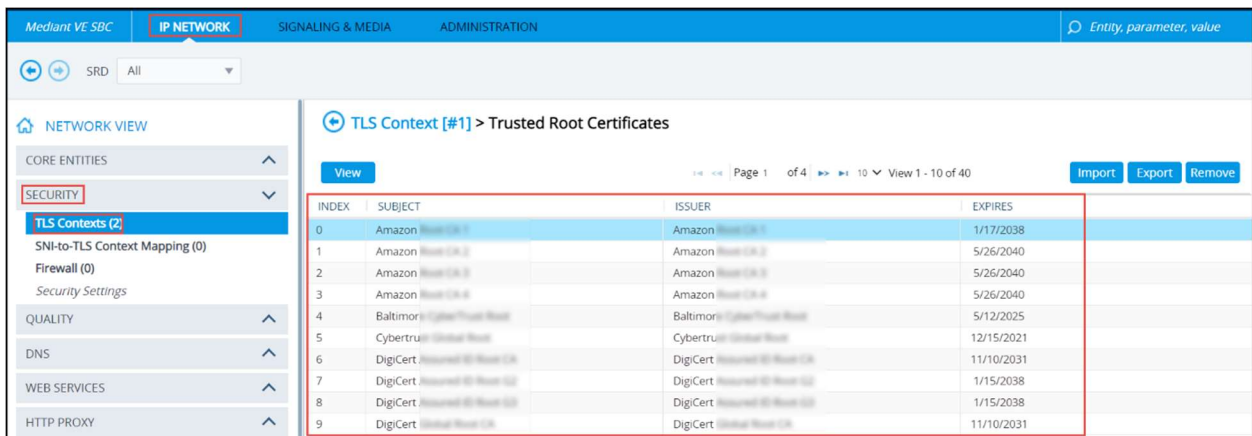


Figure 82: Trusted Root Certificate Import option

Amazon Chime Voice Connector Root Certificate can be downloaded from Amazon Chime Voice Connector account. Generate Self-signed certificate for the created TLS context. To configure

media security, navigate to **SETUP** and select **SIGNALING & MEDIA**. Expand **MEDIA** and click on **Media Security**. Under General section, set **Media Security** as Enable.

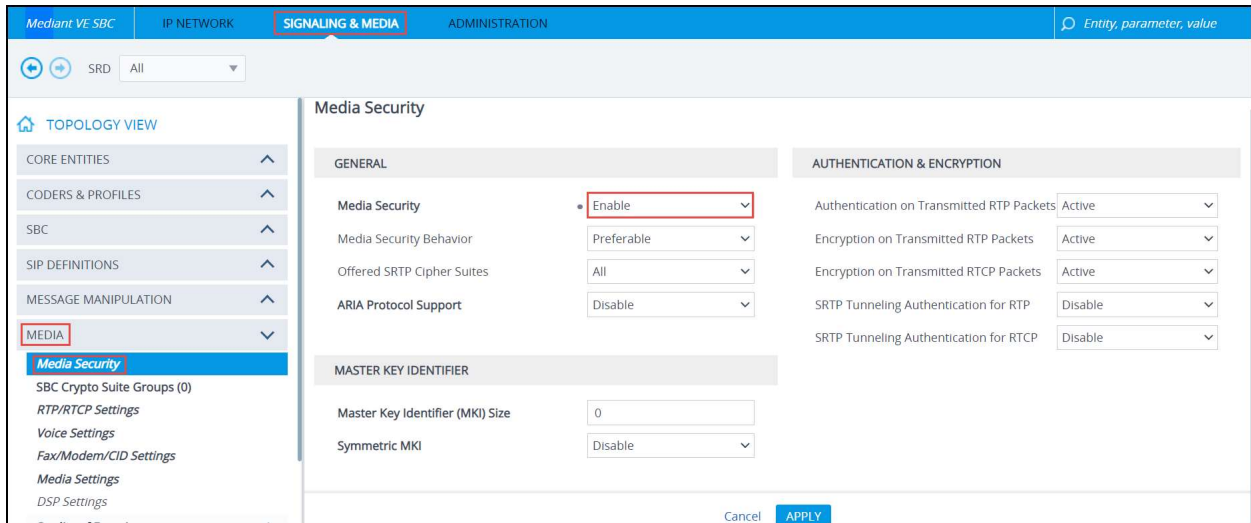


Figure 83: Media Security

In the IP Profile for Amazon Chime Voice Connector (SIP Trunk & SIPREC) **SBC Media Security Mode** must be set to Secured.

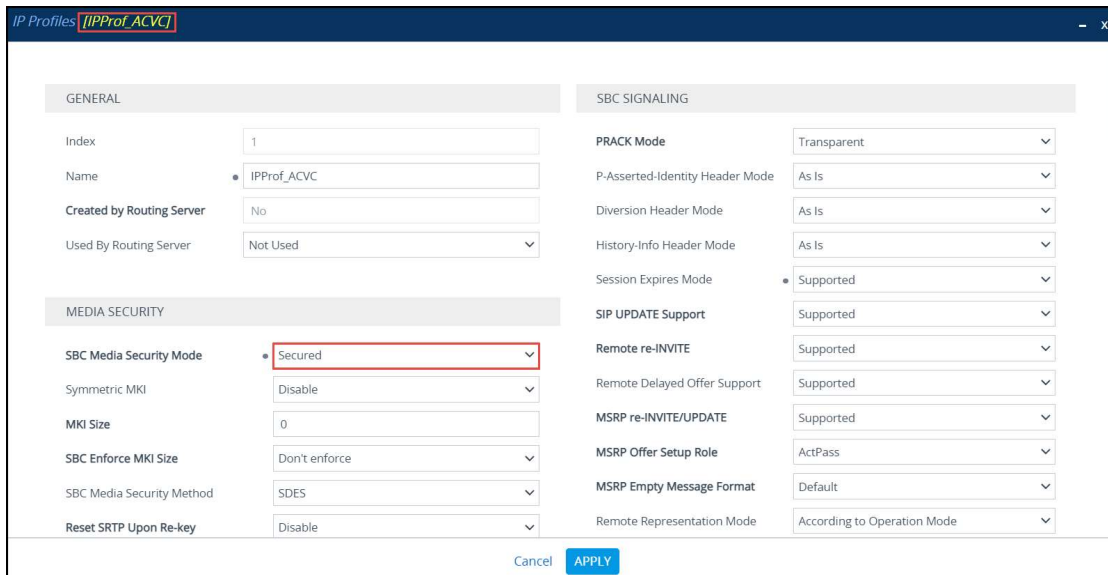


Figure 84: IP Profile - Amazon Chime Voice Connector (SIP Trunk)

Figure 85: IP Profile- Amazon Chime Voice Connector (SIPREC)

In the **Proxy Sets**, **Proxy Address** for Amazon Chime Voice Connector (SIP Trunk & SIPREC) **Transport Type** must be set to TLS and **Proxy Address** with Port Number 5061

Figure 86: Proxy Sets- Amazon Chime Voice Connector (SIP Trunk & SIPREC)

### 4.3.12 Message Manipulation configuration

SIP message manipulation rules are created to modify SIP headers for each IP entity based on manipulation sets enabled in IP Groups.

The following are the message manipulation created for interoperability between **Genesys Cloud** and **AudioCodes VE**.

Note, the configurations are taken from the CLI of AudioCodes SBC.

```
message message-manipulations 3
  manipulation-name "GenesysREQURI"
  manipulation-set-id 11
  message-type "Invite.Request"
  action-subject "Header.Request-URI.URL.Host"
  action-type modify
  action-value "'10.64.4.14'"
  activate
exit
message message-manipulations 4
  manipulation-name "GenesysToHdr"
  manipulation-set-id 11
  message-type "Invite.Request"
  action-subject "Header.To.URL.Host"
  action-type modify
  action-value "'10.64.4.14'"
  activate
exit
message message-manipulations 5
  manipulation-name "GenesysFROMHdr"
  manipulation-set-id 11
```

message-type "Invite.Request"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value "'10.64.5.147'"  
activate  
exit  
message message-manipulations 19  
manipulation-name "GenesysACKHdr"  
manipulation-set-id 11  
message-type "Ack.Request"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value "'10.64.5.147'"  
activate  
exit  
message message-manipulations 20  
manipulation-name "GenesysUPDATEFromHdr"  
manipulation-set-id 11  
message-type "Update"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value "'10.64.5.147'"  
activate  
exit  
message message-manipulations 23  
manipulation-name "GenesysBYEFROMHdr"  
manipulation-set-id 11

```
message-type "BYE.Request"
action-subject "Header.From.URL.Host"
action-type modify
action-value "'10.64.5.147'"
activate
exit
message message-manipulations 28
manipulation-name "GenesysToHdrRemvTLS"
manipulation-set-id 11
message-type "Invite.Request"
action-subject "Header.To.Param"
action-type remove
action-value "'transport=TLS'"
activate
exit
```

Following are the message manipulation created for interoperability between **Audiocodes VE** and **Amazon Chime Voice Connector (SIP Trunk)**

```
message message-manipulations 0
manipulation-name "REQURIACVCHost"
manipulation-set-id 10
message-type "Invite.Request"
action-subject "Header.Request-URI.URL.Host"
action-type modify
action-value "'gdxxxxxx.voiceconnector.chime.aws'"
activate
```

exit  
message message-manipulations 1  
manipulation-name "ACVCToHdrHost"  
manipulation-set-id 10  
message-type "Invite.Request"  
action-subject "Header.To.URL.Host"  
action-type modify  
action-value ""gdxxxxxx.voiceconnector.chime.aws""  
activate

exit  
message message-manipulations 2  
manipulation-name "FromHdrACVC"  
manipulation-set-id 10  
message-type "Invite.Request"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value ""gdxxxxxx.voiceconnector.chime.aws""  
activate

exit  
message message-manipulations 24  
manipulation-name "ACVCACKFromhdr"  
manipulation-set-id 10  
message-type "Ack.Request"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value ""gdxxxxxx.voiceconnector.chime.aws""  
activate

exit  
message message-manipulations 25  
manipulation-name "ACVCBYEFromHdr"  
manipulation-set-id 10  
message-type "Bye.Request"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value "'gdxxxxxx.voiceconnector.chime.aws'"  
activate

exit  
message message-manipulations 26  
manipulation-name "ACVCFromHdrReINVITE"  
manipulation-set-id 10  
message-type "Reinvite.Request"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value "'gdxxxxxx.voiceconnector.chime.aws'"  
activate

exit  
message message-manipulations 27  
manipulation-name "ACVCFromHdrUpdate"  
manipulation-set-id 10  
message-type "Update"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value "'gdxxxxxx.voiceconnector.chime.aws'"  
activate

exit

Following are the message manipulation created for interoperability between **Audiocodes VE** and **Amazon Chime Voice Connector (SIPREC)**

message message-manipulations 6

manipulation-name "ACVCHdrFromSIPREC"

manipulation-set-id 12

message-type "Invite.Request"

condition 'Body.application/rs-metadata regex (.\*)(<nameID aor=\*)(.\*)(@)(.)(.\*)(<nameID aor=\*)(.\*)'

action-subject "Header.From.URL.User"

action-type modify

action-value "\$3"

activate

exit

message message-manipulations 7

manipulation-name "ACVCCContactSIPREC"

manipulation-set-id 12

message-type "Invite.Request"

condition "Header.Contact regex (.\*)(>);(.\*)"

action-subject "Header.Contact"

action-type modify

action-value "\$1+\$2+'+sip.src'"

activate

exit

message message-manipulations 8

manipulation-name "ACVCToHdrHostSIPREC"

manipulation-set-id 12  
message-type "Invite.Request"  
action-subject "Header.To.URL.Host"  
action-type modify  
action-value ""gdxxxxxx.voiceconnector.chime.aws""  
activate  
exit  
message message-manipulations 9  
manipulation-name "REQURIACVCHostSIPREC"  
manipulation-set-id 12  
message-type "Invite.Request"  
action-subject "Header.Request-URI.URL.Host"  
action-type modify  
action-value ""gdxxxxxx.voiceconnector.chime.aws""  
activate  
exit  
message message-manipulations 10  
manipulation-name "ACVCFFromHdrHostSIPREC"  
manipulation-set-id 12  
message-type "Invite.Request"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value ""gdxxxxxx.voiceconnector.chime.aws""  
activate  
exit  
message message-manipulations 11  
manipulation-name "ACVCToHdrUserSIPREC"

```
manipulation-set-id 12
message-type "Invite.Request"
condition 'Body.application/rs-metadata regex (.*)(<nameID aor=")(.)(@)(.)(<nameID
aor=")(.)(@)(.)*'
action-subject "Header.To.URL.User"
action-type modify
action-value "$7"
activate
exit
message message-manipulations 12
manipulation-name "ACVCREQUIUserSIPREC"
manipulation-set-id 12
message-type "Invite.Request"
condition 'Body.application/rs-metadata regex (.*)(<nameID aor=")(.)(@)(.)(<nameID
aor=")(.)(@)(.)*'
action-subject "Header.Request-URI.URL.User"
action-type modify
action-value "$7"
activate
exit
message message-manipulations 13
manipulation-name "ACVCFFromRemvUserPhoneSIPREC"
manipulation-set-id 12
message-type "Invite.Request"
action-subject "Header.From.URL.UserPhone"
action-type remove
activate
exit
```

message message-manipulations 14

manipulation-name "ACVCToRemvUserPhoneSIPREC"

manipulation-set-id 12

message-type "Invite.Request"

action-subject "Header.To.URL.UserPhone"

action-type remove

activate

exit

message message-manipulations 15

manipulation-name "ACVCREQURIRemvUserPhoneSIPREC"

manipulation-set-id 12

message-type "Invite.Request"

action-subject "Header.Request-URI.URL.UserPhone"

action-type remove

activate

exit

message message-manipulations 16

manipulation-name "ACVCHdrFromReINVITESIPREC"

manipulation-set-id 12

message-type "Reinvite.Request"

condition 'Body.application/rs-metadata regex (.\*)(<nameID aor=\*)(.\*)(@)(.\*)(<nameID aor=\*)(.\*)'

action-subject "Header.From.URL.User"

action-type modify

action-value "\$3"

activate

exit

```
message message-manipulations 17
  manipulation-name "ACVCHdrToReINVITESIPREC"
  manipulation-set-id 12
  message-type "Reinvite.Request"
  condition 'Body.application/rs-metadata regex (.*)(<nameID aor=")(.*)(@)(.*)(<nameID
aor=")(.*)(@)(.*)'
  action-subject "Header.To.URL.User"
  action-type modify
  action-value "$7"
  activate
  exit
message message-manipulations 18
  manipulation-name "ACVCREQUIHdrReINVITESIPREC"
  manipulation-set-id 12
  message-type "Reinvite.Request"
  condition 'Body.application/rs-metadata regex (.*)(<nameID aor=")(.*)(@)(.*)(<nameID
aor=")(.*)(@)(.*)'
  action-subject "Header.Request-URI.URL.User"
  action-type modify
  action-value "$7"
  activate
  exit
message message-manipulations 21
  manipulation-name "ACVCACKFROMSIPREC"
  manipulation-set-id 12
  message-type "Ack.Request"
  action-subject "Header.From.URL.Host"
  action-type modify
```

action-value ""10.64.5.147""  
activate  
exit  
message message-manipulations 22  
manipulation-name "ACVCBYEFROMSIPREC"  
manipulation-set-id 12  
message-type "Bye.Request"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value ""10.64.5.147""  
activate  
exit