



Amazon Chime SDK Voice Connector

SIPREC Configuration Guide

**FreePBX 16.0.40.4 Asterisk 20.4.0 and
AudioCodes Mediant Virtual Edition
(VE) SBC v7.40A.500.017**

September 2023

Document History

Rev. No.	Date	Description
1.0	September-26-2023	SIPREC Configuration Guide

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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 **FreePBX (Asterisk)** and **AudioCodes Mediant Virtual Edition Session Border Controller** to connect to **Amazon Chime SDK Voice Connector** for streaming audio to Kinesis Video Streams (KVS). The audio can then be processed by services such as Amazon Transcribe or Amazon Chime SDK Call Analytics to fulfill a number of business purposes.

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1.1 Amazon Chime SDK Voice Connector

Amazon Chime SDK 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 SDK Voice Connector uses the industry-standard Session Initiation Protocol (SIP). Amazon Chime SDK 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 SDK Voice Connector API. Amazon Chime SDK Voice Connector offers cost-effective rates for inbound and outbound calls. Calls into Amazon Chime SDK Voice Connector meetings, as well as calls to other Amazon Chime SDK Voice Connector customers are at no additional cost. With Amazon Chime SDK 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 topology for the SIPREC lab configuration is illustrated below. Customers should substitute their own configuration settings where appropriate.

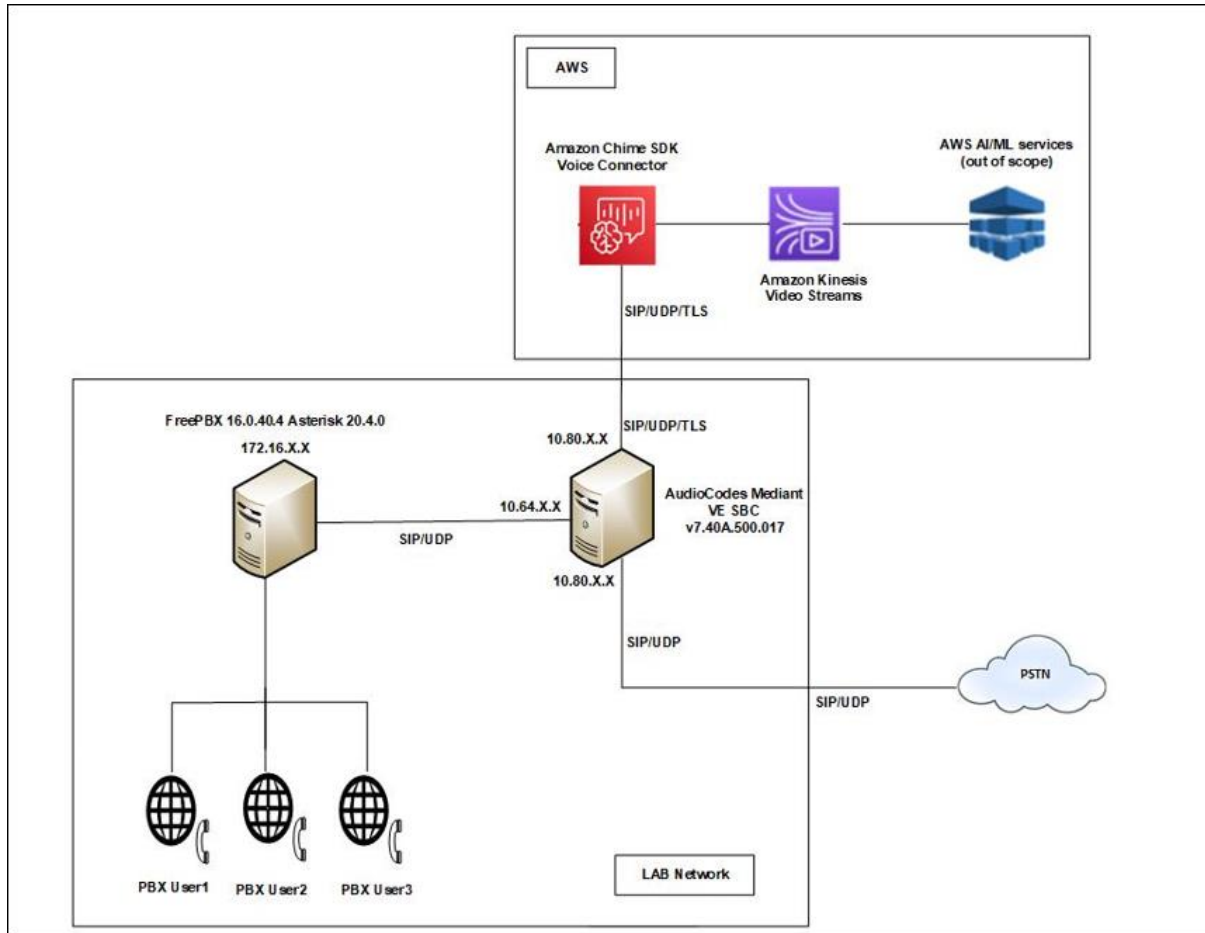


Figure 1: Network Topology

The signaling and media flow is illustrated below:

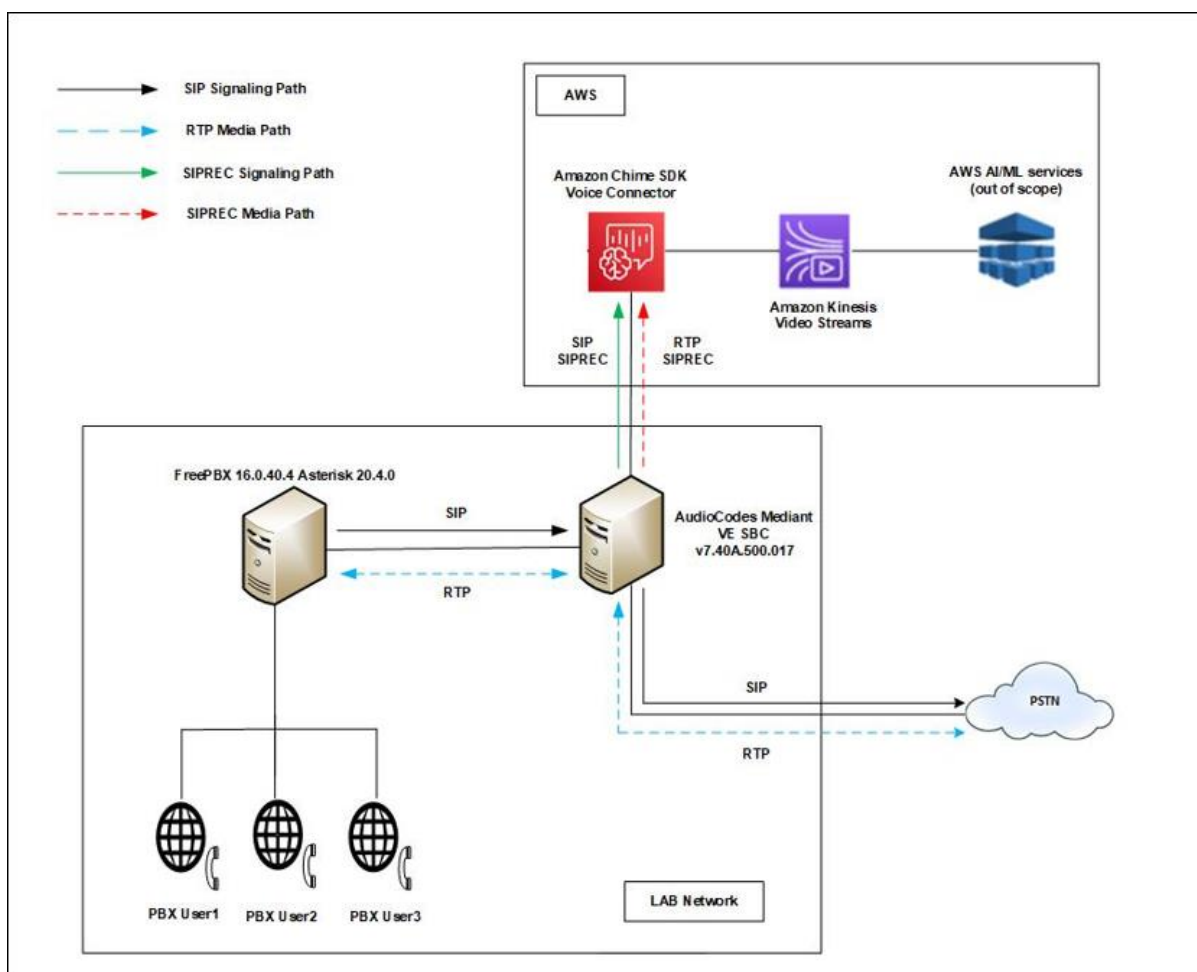


Figure 2: Signaling and Media Flow

2.1 Hardware Components

- VMWare server running ESXi 7.0 or later used for the following virtual machine
 - FreePBX Asterisk
- VMWare server running ESXi 6.7.0 or later used for the following virtual machine
 - AudioCodes Mediant VE SBC
- Polycom IP Phone(s)
 - VVX 150
 - VVX 250
 - SoundPoint IP 650

2.2 Software Requirements

- FreePBX 16.0.40.4 Asterisk 20.4.0
- AudioCodes Mediant VE SBC v7.40A.500.017

3 Features

3.1 Features Supported and Not Supported

Table 1 – Supported and Not Supported Features

SL. No.	Features/Services	Supported
1	Basic Calls	✓
2	Call Hold and Resume	✓
3	Attended Transfer	✓
4	Blind Transfer	✓
5	External Transfer	✓
6	Internal Conference	✓
7	External Conference	✓
8	Call Queueing	✓
9	Consultation	✓
10	Extended Consultation	✓
11	Multi-party Conference	✓
12	Emergency Calling	✓
13	International Calling	✓

3.2 Features Not Tested

- None

3.3 Caveats and Limitations

- There is no re-invite from PBX while the call is placed on HOLD. The recording is not paused and the music on hold is recorded. This observation is applicable to Transfer and Conference scenarios where the call hold feature is involved.
- SIPREC metadata is not updated for the internal transfer and internal conference scenarios.

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

This section presents an overview of the steps that are required to configure FreePBX Asterisk and AudioCodes SBC for SIPREC using SIP Trunking with Amazon Chime SDK Voice Connector.

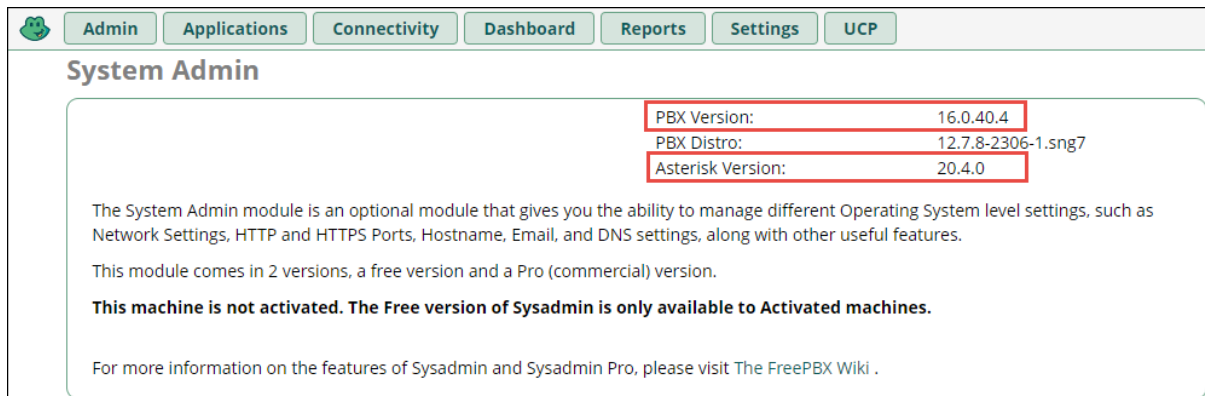
Table 2 – PBX and SBC Configuration Steps

Steps	Description	Reference
Step 1	FreePBX Asterisk Configuration	Section 4.2
Step 2	AudioCodes SBC Configuration	Section 4.3
Step 3	Amazon Chime SDK Voice Connector Configuration	Amazon Chime SDK Voice Connector
Step 4	Amazon Kinesis Configuration	Amazon Kinesis Configuration
Step 5	Amazon Chime SDK Call Analytics configuration	Amazon Chime SDK Call Analytics configuration

4.2 FreePBX Asterisk Configuration

This section, with screenshots taken from the FreePBX Asterisk system used for the interoperability testing, gives a general overview of the PBX configuration.

4.2.1 FreePBX Asterisk Version



The screenshot displays the 'System Admin' interface of FreePBX. At the top, there is a navigation bar with tabs for Admin, Applications, Connectivity, Dashboard, Reports, Settings, and UCP. The main content area is titled 'System Admin' and contains a table of system information. The table lists the PBX Version as 16.0.40.4, the PBX Distro as 12.7.8-2306-1.sng7, and the Asterisk Version as 20.4.0. Below the table, there is a paragraph explaining that the System Admin module is optional and manages OS-level settings. It also notes that the module comes in free and Pro versions. A bold warning states: 'This machine is not activated. The Free version of Sysadmin is only available to Activated machines.' At the bottom, it provides a link to the FreePBX Wiki for more information.

PBX Version:	16.0.40.4
PBX Distro:	12.7.8-2306-1.sng7
Asterisk Version:	20.4.0

The System Admin module is an optional module that gives you the ability to manage different Operating System level settings, such as Network Settings, HTTP and HTTPS Ports, Hostname, Email, and DNS settings, along with other useful features.

This module comes in 2 versions, a free version and a Pro (commercial) version.

This machine is not activated. The Free version of Sysadmin is only available to Activated machines.

For more information on the features of Sysadmin and Sysadmin Pro, please visit [The FreePBX Wiki](#) .

Figure 3: FreePBX Asterisk Version

4.2.2 Extensions

The Extension module is used to set up the extension number, the name of the extension, the password, voicemail settings for the extension, and other options.

Navigate to **Application → Extensions → Add New SIP[Chan_Pjsip] Extension**

- **User Extension:** Enter the Extension of the User
- **Outbound CID:** Enter the Outbound CID for the User

The screenshot shows the 'Add New SIP[Chan_Pjsip] Extension' configuration page for extension 0072. The 'Display Name' and 'Outbound CID' fields are highlighted with a red box. The 'Submit' button is also highlighted with a red box.

Figure 4: Asterisk Extension

- The below screenshot shows the extensions created in the FreePBX Asterisk

	Extension	Name	CW	DND	FM/FM	CF	CFB	CFU	Type	Actions
<input type="checkbox"/>	0072	0072	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	pjsip	
<input type="checkbox"/>	0073	0073	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	pjsip	
<input type="checkbox"/>	0083	0083	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	pjsip	

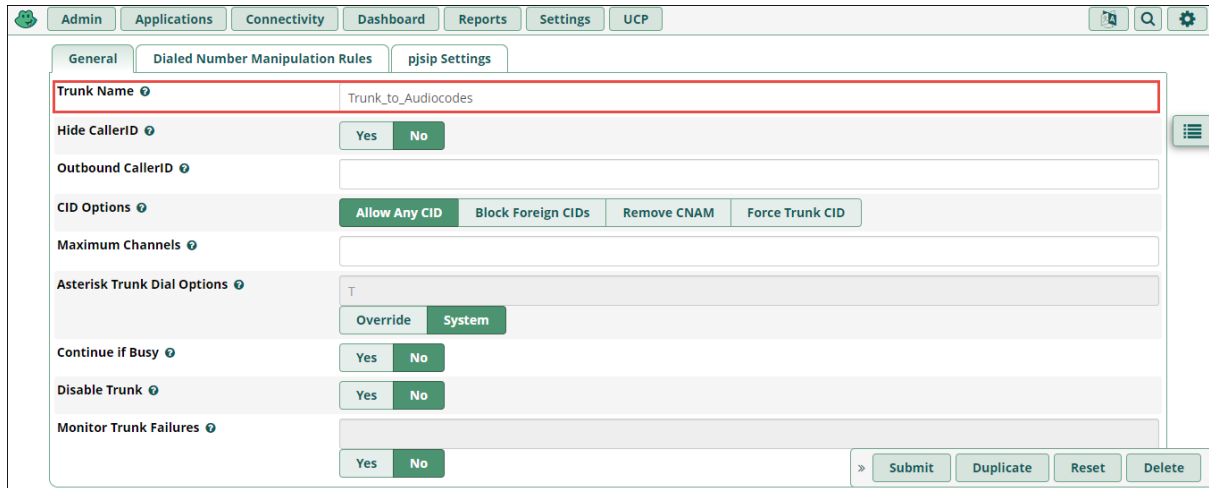
Figure 5: Asterisk Extension List

4.2.3 Trunk

The Trunks Module is used to connect the FreePBX Asterisk system to another VOIP system so that the calls can be sent out to and received in from that system.

Navigate to **Connectivity → Trunks → Add Trunk → Add SIP (Chan_Pjsip) Trunk**

Trunk Name: Enter a name for the Trunk



The screenshot shows the 'Add SIP (Chan_Pjsip) Trunk' configuration page. The 'Trunk Name' field is highlighted with a red box and contains the text 'Trunk_to_Audiocodes'. Other fields include 'Hide CallerID' (Yes/No), 'Outbound CallerID', 'CID Options' (Allow Any CID, Block Foreign CIDs, Remove CNAM, Force Trunk CID), 'Maximum Channels', 'Asterisk Trunk Dial Options' (Override/System), 'Continue if Busy' (Yes/No), 'Disable Trunk' (Yes/No), and 'Monitor Trunk Failures' (Yes/No). Buttons for 'Submit', 'Duplicate', 'Reset', and 'Delete' are at the bottom right.

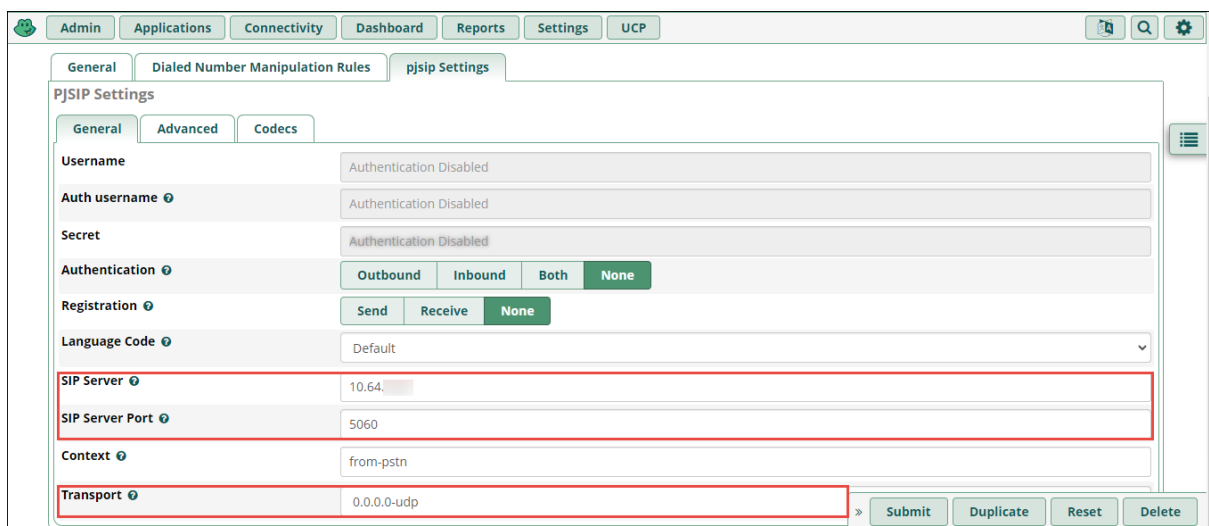
Figure 6: Asterisk Trunk

Navigate to **Pjsip settings → General**

SIP Server: 10.64.X.X (IP of AudioCodes SBC's Network Interface towards the FreePBX Asterisk)

SIP Server Port: 5060

Transport: 0.0.0.0-udp



The screenshot shows the 'Pjsip Settings' page. The 'SIP Server', 'SIP Server Port', and 'Transport' fields are highlighted with red boxes. The 'SIP Server' field contains '10.64', 'SIP Server Port' contains '5060', and 'Transport' contains '0.0.0.0-udp'. Other fields include 'Username', 'Auth username', 'Secret', 'Authentication', 'Registration', 'Language Code', and 'Context'. Buttons for 'Submit', 'Duplicate', 'Reset', and 'Delete' are at the bottom right.

Figure 7: Asterisk Trunk Continuation

Navigate to **Pjsip settings** → **Codecs**

Enable Ulaw

Click Submit

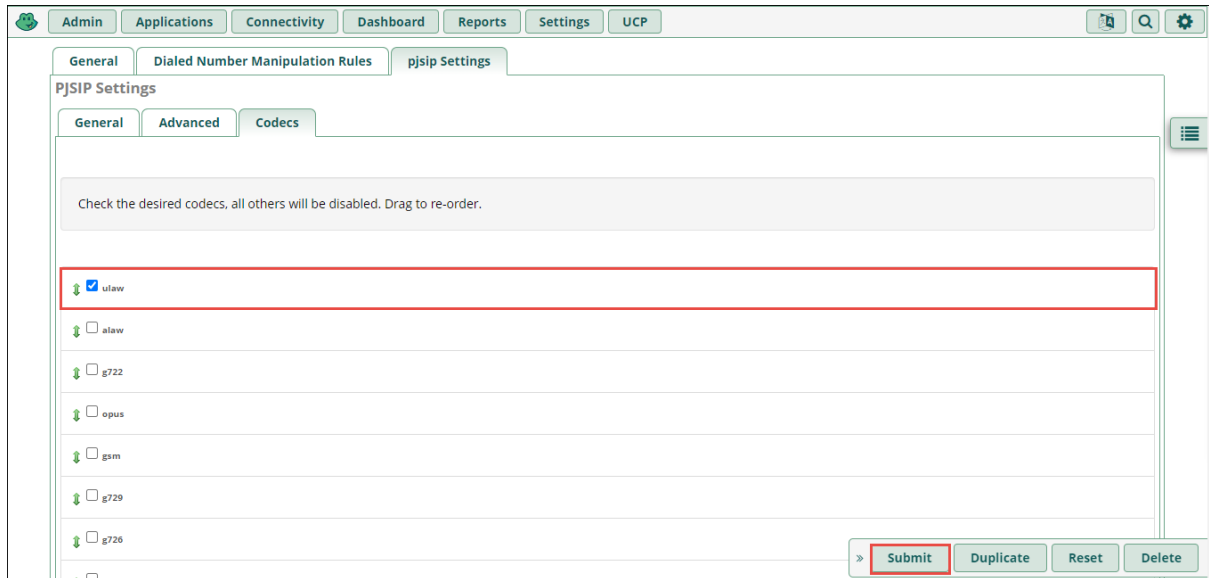


Figure 8: Asterisk Trunk Continuation

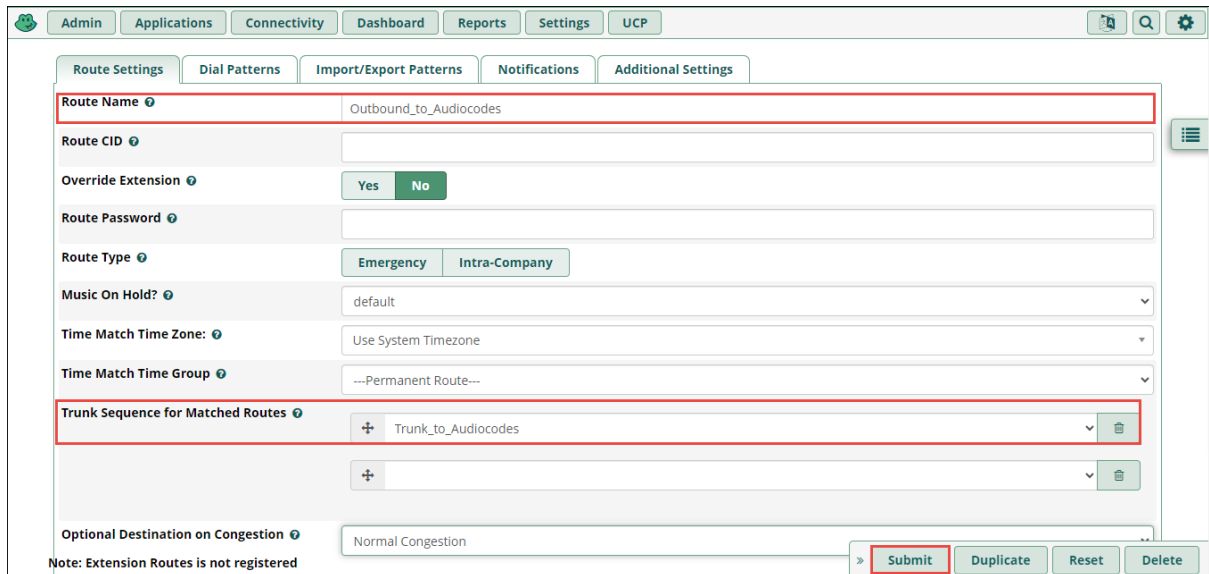
4.2.4 Outbound Route

The Outbound Route Module is configured to tell the endpoints registered in the PBX to which numbers they are permitted to call and which Trunk to send the calls to.

Navigate to **Connectivity** → **Outbound Routes** → **Add Outbound Route**

Route Name: Enter the Name for the Outbound Route

Trunk Sequence for Matched Route: Select the Trunk created



The screenshot shows the Asterisk Outbound Route configuration page. The 'Route Name' field is filled with 'Outbound_to_Audiocodes'. The 'Trunk Sequence for Matched Routes' field is filled with 'Trunk_to_Audiocodes'. The 'Submit' button is highlighted with a red box. Other fields include 'Route CID', 'Override Extension' (Yes/No), 'Route Password', 'Route Type' (Emergency/Intra-Company), 'Music On Hold?' (default), 'Time Match Time Zone' (Use System Timezone), and 'Time Match Time Group' (---Permanent Route---). The 'Optional Destination on Congestion' field is set to 'Normal Congestion'. A note at the bottom states 'Note: Extension Routes is not registered'.

Figure 9: Asterisk Outbound Route

Navigate to **Dial Patterns** and add the below patterns and click Submit

For PSTN dialing

Prefix: 7

Match Pattern: 214XXXXXXX

For International dialing

Prefix: 2

Match Pattern: 01191XXXXXXXXXX

For Short code dialing

Prefix: 1

Match Pattern: 511

Admin Applications Connectivity Dashboard Reports Settings UCP

Outbound Routes

Edit Route: Outbound_to_Audiocodes: Outbound_to_Audiocodes

Route Settings Dial Patterns Import/Export Patterns Notifications Additional Settings

Dial Patterns that will use this Route

Pattern Help +

Dial patterns wizards

{	prepend	}	1		[511	/	CallerID	}	+
{	prepend	}	2		[01191XXXXXXXXXX	/	CallerID	}	+
{	prepend	}	7		[214XXXXXXXX	/	CallerID	}	+

> Submit Duplicate Reset Delete

Figure 10: Asterisk Outbound Route Continuation

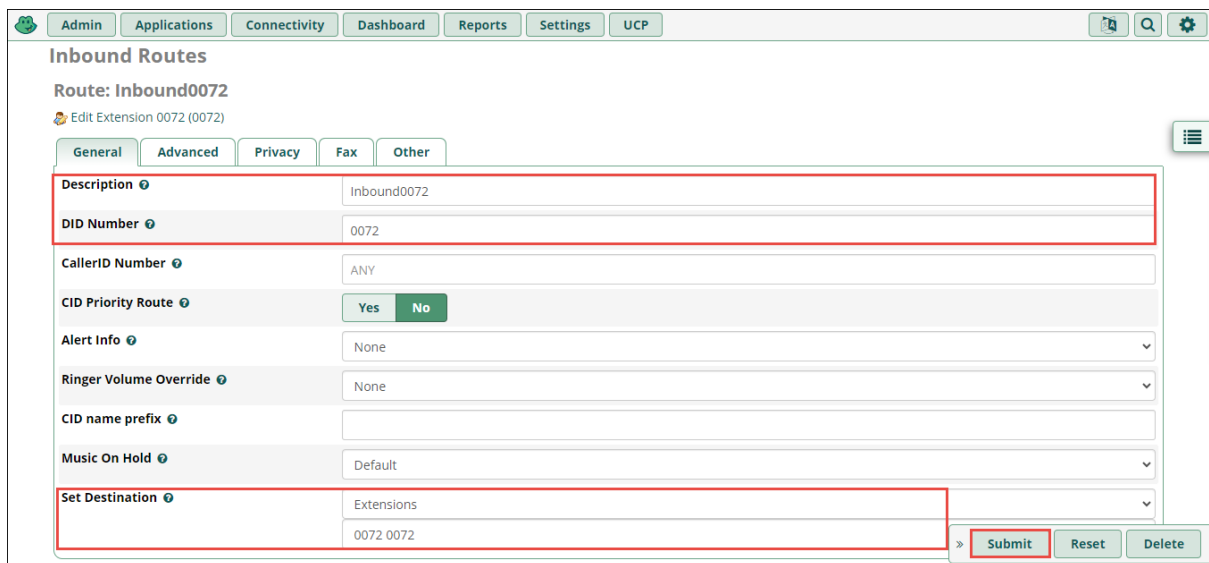
4.2.5 Inbound Route

The Inbound Route module is configured to route the incoming calls in the FreePBX Asterisk to the corresponding endpoints or the IVR or to the Call Queues or the other options available in the inbound routes settings.

Navigate to **Connectivity → Inbound Routes → Add Inbound Route**

DID Number: Enter the Extension of the user

Set Destination: Select Extensions



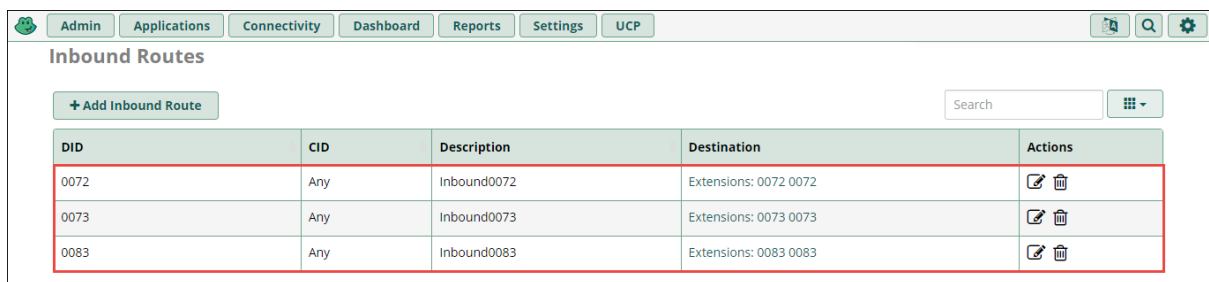
The screenshot shows the configuration page for an Inbound Route named 'Inbound0072'. The page has a navigation bar with tabs: Admin, Applications, Connectivity, Dashboard, Reports, Settings, and UCP. Below the navigation bar, there are tabs for 'General', 'Advanced', 'Privacy', 'Fax', and 'Other'. The 'General' tab is selected. The configuration fields are as follows:

- Description: Inbound0072
- DID Number: 0072
- CallerID Number: ANY
- CID Priority Route: Yes (selected), No
- Alert Info: None
- Ringer Volume Override: None
- CID name prefix: (empty)
- Music On Hold: Default
- Set Destination: Extensions (selected), 0072 0072

At the bottom right, there are buttons for 'Submit', 'Reset', and 'Delete'.

Figure 11: Asterisk Inbound Route

- The below screenshot shows the inbound routes created in the FreePBX Asterisk



The screenshot shows the 'Inbound Routes List' page. It features a navigation bar with tabs: Admin, Applications, Connectivity, Dashboard, Reports, Settings, and UCP. Below the navigation bar, there is a '+ Add Inbound Route' button and a search bar. The main content is a table with the following data:







DID	CID	Description	Destination	Actions
0072	Any	Inbound0072	Extensions: 0072 0072	 
0073	Any	Inbound0073	Extensions: 0073 0073	 
0083	Any	Inbound0083	Extensions: 0083 0083	 

Figure 12: Asterisk inbound Routes List

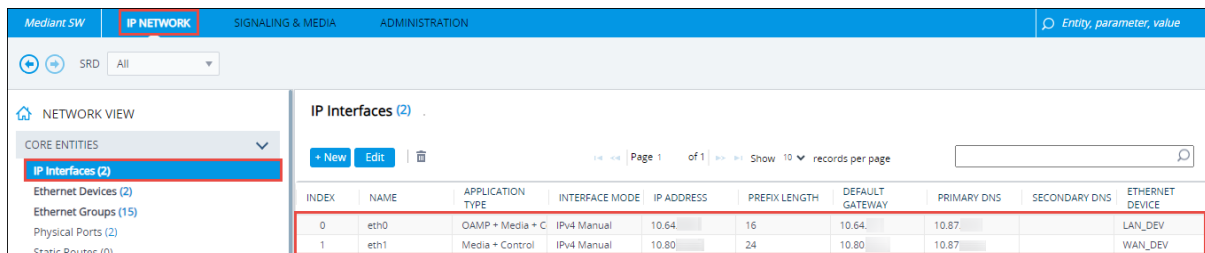
4.3 AudioCodes SBC Configuration

This section, with screenshots taken from the AudioCodes Mediant VE SBC system used for the interoperability testing, gives a general overview of the AudioCodes SBC configuration for enabling SIPREC streaming to the AWS Chime SDK Voice Connector system for interworking with FreePBX Asterisk and a SIP Trunk.

4.3.1 Network Interfaces

Two network interfaces are configured. One towards the FreePBX Asterisk and another towards Amazon Chime SDK Voice Connector.

Navigate to **Setup** → **IP Network** → **Core Entities** → **IP Interfaces**



The screenshot shows the 'IP NETWORK' configuration page in the Mediant SW interface. The 'CORE ENTITIES' sidebar on the left has 'IP Interfaces (2)' selected. 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	eth0	OAMP + Media + C	IPv4 Manual	10.64	16	10.64	10.87		LAN_DEV
1	eth1	Media + Control	IPv4 Manual	10.80	24	10.80	10.87		WAN_DEV

Figure 13: IP Interfaces List

- **eth0**: IP interface towards FreePBX Asterisk
- **eth1**: IP interface towards Amazon Chime SDK Voice Connector

4.3.1.1 LAN IP Interface

Name: eth0

Application Type: OAMP + Media + Control

Ethernet Device: LAN DEV

Primary DNS: 10.87.X.X

Interface Mode: IPv4 Manual

IP address: 10.64.X.X

Default Gateway: 10.64.X.X

IP Interfaces [eth0]

GENERAL

Index: 0

Name: eth0

Application Type: OAMP + Media + Control

Ethernet Device: #0 [LAN_DEV]

DNS

Primary DNS: 10.87

Secondary DNS:

Overwrite Dynamic DNS Servers: Disable

IP ADDRESS

Interface Mode: IPv4 Manual

IP Address: 10.64

Prefix Length: 16

Default Gateway: 10.64

Changes to the network interface will stop all services running on the interface, in particular, connectivity with the device's management interface and current calls

Cancel APPLY

Activate Windows
Go to Settings to activate Windows.

Figure 14: LAN IP Interface

4.3.1.2 WAN IP Interface

Name: eth1

Application Type: Media + Control

Ethernet Device: WAN DEV

Primary DNS: 10.87.X.X

Interface Mode: IPv4 Manual

IP address: 10.80.X.X

Default Gateway: 10.80.X.X

The screenshot shows the 'IP Interfaces [eth1]' configuration window. It is divided into three main sections: GENERAL, IP ADDRESS, and DNS. The GENERAL section includes fields for Index (1), Name (eth1), Application Type (Media + Control), and Ethernet Device (#1 [WAN_DEV]). The IP ADDRESS section includes Interface Mode (IPv4 Manual), IP Address (10.80.X.X), Prefix Length (24), and Default Gateway (10.80.X.X). The DNS section includes Primary DNS (10.87.X.X), Secondary DNS, and Overwrite Dynamic DNS Servers (Disable). A warning message at the bottom states: 'Changes to the network interface will stop all services running on the interface, in particular, current calls'. There are 'Cancel' and 'APPLY' buttons at the bottom center, and an 'Activate Windows' watermark at the bottom right.

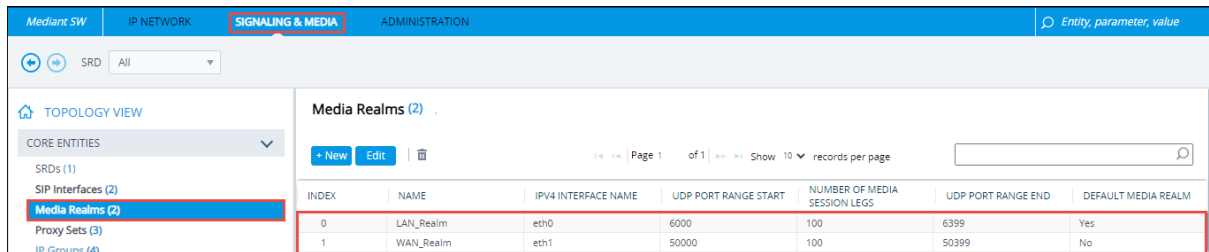
Section	Field	Value
GENERAL	Index	1
	Name	eth1
	Application Type	Media + Control
	Ethernet Device	#1 [WAN_DEV]
IP ADDRESS	Interface Mode	IPv4 Manual
	IP Address	10.80.X.X
	Prefix Length	24
	Default Gateway	10.80.X.X
DNS	Primary DNS	10.87.X.X
	Secondary DNS	
	Overwrite Dynamic DNS Servers	Disable

Figure 15: WAN IP Interface

4.3.2 Media Realms

Media Realm is specified by a UDP port range and a maximum number of permitted media sessions. Two Media Realms are configured- Internal (LAN) traffic and external (WAN) traffic.

Navigate to **Setup → Signaling & Media → Core Entities Media Realms**



INDEX	NAME	IPv4 INTERFACE NAME	UDP PORT RANGE START	NUMBER OF MEDIA SESSION LEGS	UDP PORT RANGE END	DEFAULT MEDIA REALM
0	LAN_Realm	eth0	6000	100	6399	Yes
1	WAN_Realm	eth1	50000	100	50399	No

Figure 16: Media Realms List

4.3.2.1 Media Realm LAN

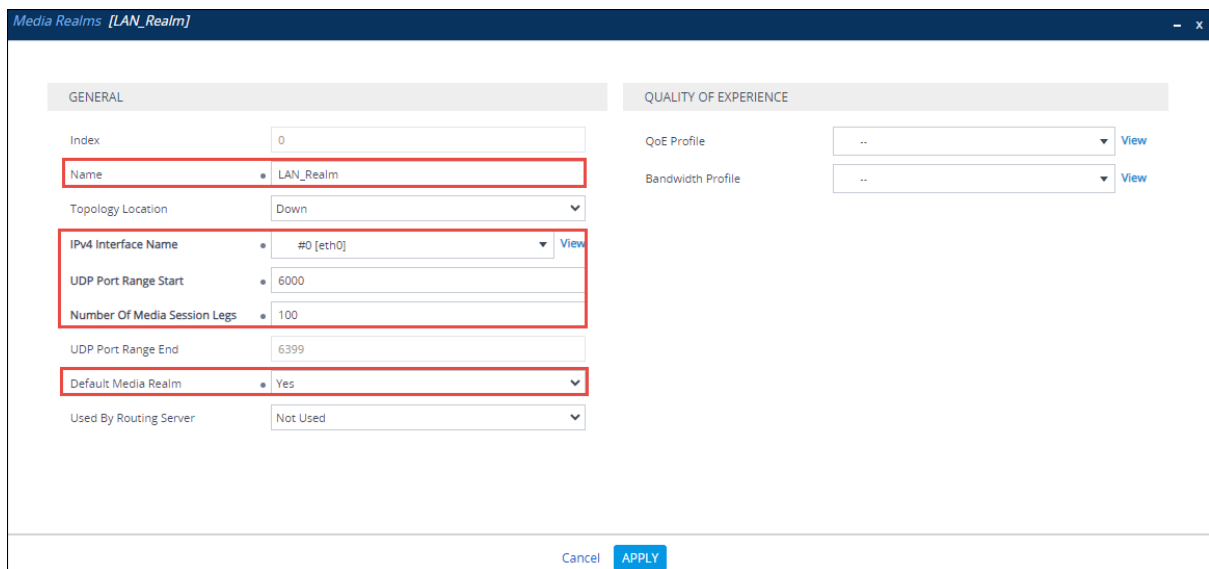
Name: LAN Realm

IPv4 Interface Name: eth0

UDP Port Start Range: 6000

Number of Media Session Legs: 100

Default Media realm: Yes



Media Realms [LAN_Realm]

GENERAL

Index: 0

Name: LAN_Realm

Topology Location: Down

IPv4 Interface Name: #0 [eth0]

UDP Port Range Start: 6000

Number Of Media Session Legs: 100

UDP Port Range End: 6399

Default Media Realm: Yes

Used By Routing Server: Not Used

QUALITY OF EXPERIENCE

QoE Profile: --

Bandwidth Profile: --

Cancel APPLY

Figure 17: Media Realm for LAN Interface

4.3.2.2 Media Realm WAN

Name: WAN Realm

IPv4 Interface Name: eth1

UDP Port Start Range: 50000

Number of Media Session Legs: 100

The screenshot shows a configuration window titled "Media Realms [WAN_Realm]". It is divided into two main sections: "GENERAL" and "QUALITY OF EXPERIENCE".

GENERAL Section:

- Index: 1
- Name: WAN_Realm (highlighted with a red box)
- Topology Location: Up
- IPv4 Interface Name: #1 [eth1] (highlighted with a red box)
- UDP Port Range Start: 50000 (highlighted with a red box)
- Number Of Media Session Legs: 100 (highlighted with a red box)
- UDP Port Range End: 50399
- Default Media Realm: No
- Used By Routing Server: Not Used

QUALITY OF EXPERIENCE Section:

- QoS Profile: -- (with a "View" link)
- Bandwidth Profile: -- (with a "View" link)

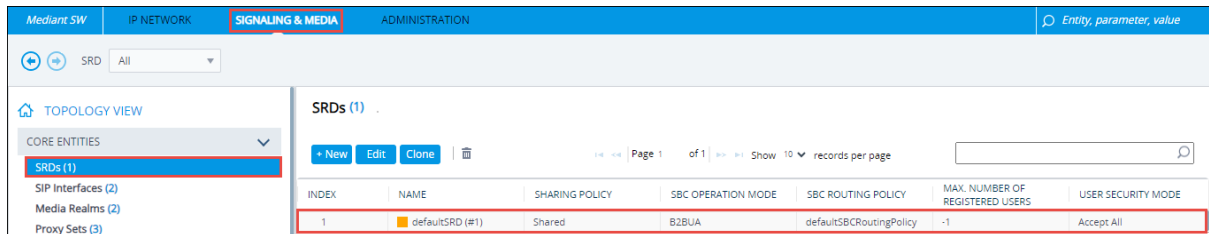
At the bottom of the window, there are "Cancel" and "APPLY" buttons. A watermark for "Activate Windows" is visible in the bottom right corner.

Figure 18: Media Realm for WAN Interface

4.3.3 SRD

Default Signaling Routing Domain (SRD) is used.

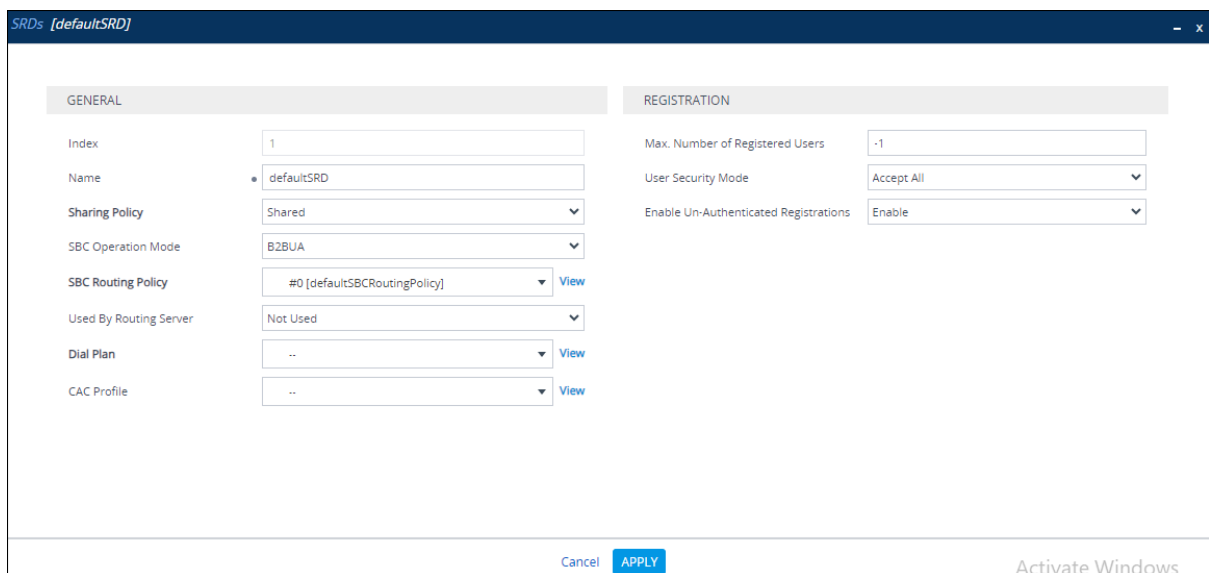
Navigate to **Setup → Signaling & Media → Core Entities →SRDs**



The screenshot shows the 'SRDs (1)' list in the management console. The table below represents the data shown in the interface:

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

Figure 19: SRD



The screenshot shows the configuration form for the default SRD. The form is divided into two sections: GENERAL and REGISTRATION.

GENERAL	REGISTRATION
Index: 1	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 [defaultSBCRoutingPolicy] View	
Used By Routing Server: Not Used	
Dial Plan: -- View	
CAC Profile: -- View	

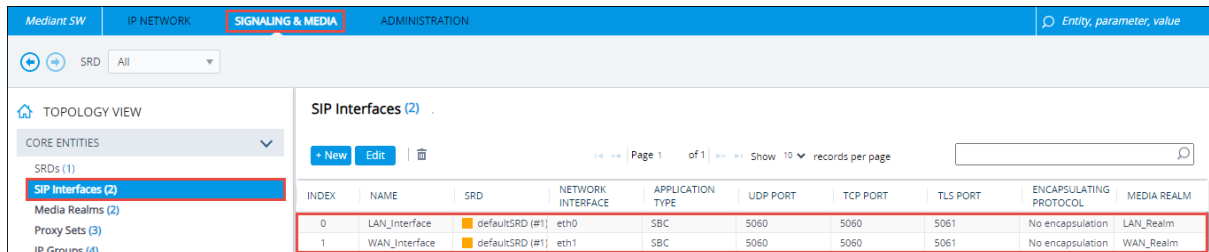
Buttons: Cancel, APPLY, Activate Windows

Figure 20: Default SRD

4.3.4 SIP Interfaces

A SIP Interface defines a local, listening port number and type (e.g., UDP), and is assigned with an IP network interface for SIP signaling traffic.

Navigate to **Setup → Signaling & Media → Core Entities → SIP Interfaces**



INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULATING PROTOCOL	MEDIA REALM
0	LAN_Interface	defaultSRD (#1)	eth0	SBC	5060	5060	5061	No encapsulation	LAN_Realm
1	WAN_Interface	defaultSRD (#1)	eth1	SBC	5060	5060	5061	No encapsulation	WAN_Realm

Figure 21: SIP Interfaces List

4.3.4.1 SIP Interface LAN

Name: LAN Interface

Network Interface: eth0

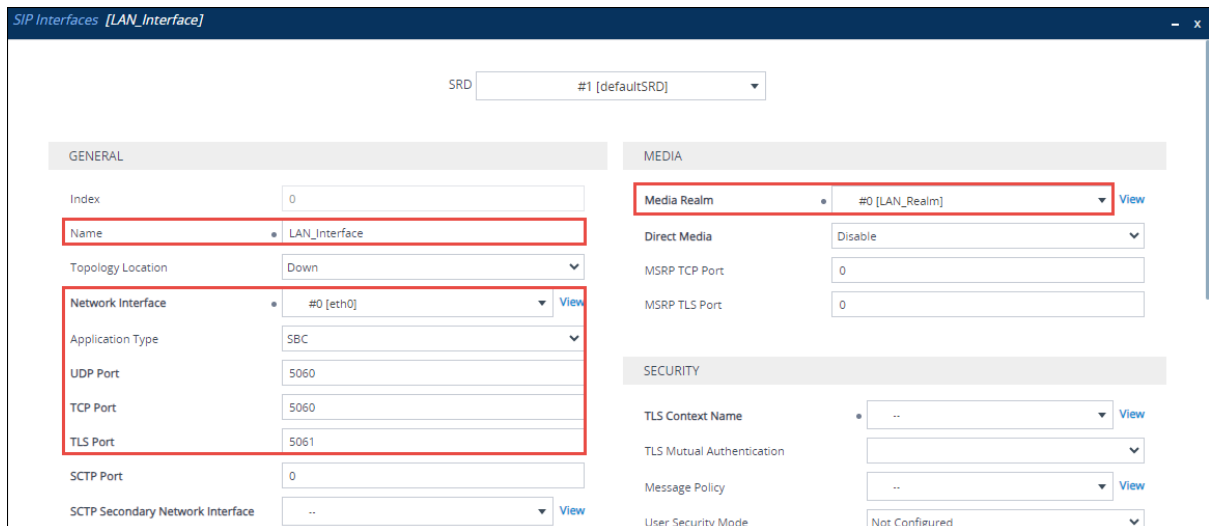
Application Type: SBC

UDP Port: 5060

TCP Port: 5060

TLS Port: 5061

Media Realm: LAN Realm



SRD: #1 [defaultSRD]

GENERAL

Index: 0

Name: LAN_Interface

Topology Location: Down

Network Interface: #0 [eth0]

Application Type: SBC

UDP Port: 5060

TCP Port: 5060

TLS Port: 5061

SCTP Port: 0

SCTP Secondary Network Interface: ..

MEDIA

Media Realm: #0 [LAN_Realm]

Direct Media: Disable

MSRP TCP Port: 0

MSRP TLS Port: 0

SECURITY

TLS Context Name: ..

TLS Mutual Authentication: ..

Message Policy: ..

User Security Mode: Not Configured

Figure 22: SIP Interface LAN

Additional UDP Ports	<input type="text"/>	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	Disable		
Used By Routing Server	Not Used		
Pre-Parsing Manipulation Set	..		View
CAC Profile	..		View
CLASSIFICATION			
Classification Failure Response Type	500		
Pre-classification Manipulation Set ID	-1		
Call Setup Rules Set ID	-1		
Cancel		APPLY	
Activate Windows			

Figure 23: SIP Interface LAN Continuation

4.3.4.2 SIP Interface WAN

Name: WAN Interface

Network Interface: eth1

Application Type: SBC

UDP Port: 5060

TCP Port: 5060

TLS Port: 5061

Media Realm: WAN Realm

SIP Interfaces [WAN_Interface]

SRD #1 [defaultSRD]

GENERAL

Index: 1

Name: WAN_Interface

Topology Location: Up

Network Interface: #1 [eth1]

Application Type: SBC

UDP Port: 5060

TCP Port: 5060

TLS Port: 5061

SCTP Port: 0

SCTP Secondary Network Interface: --

MEDIA

Media Realm: #1 [WAN_Realm]

Direct Media: Disable

MSRP TCP Port: 0

MSRP TLS Port: 0

SECURITY

TLS Context Name: --

TLS Mutual Authentication: --

Message Policy: --

User Security Mode: Not Configured

Enable Un-Authenticated Registrations: Not configured

Max. Number of Registered Users: -1

CLASSIFICATION

Classification Failure Response Type: 500

Pre-classification Manipulation Set ID: -1

Call Setup Rules Set ID: -1

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: --

CAC Profile: --

Cancel APPLY

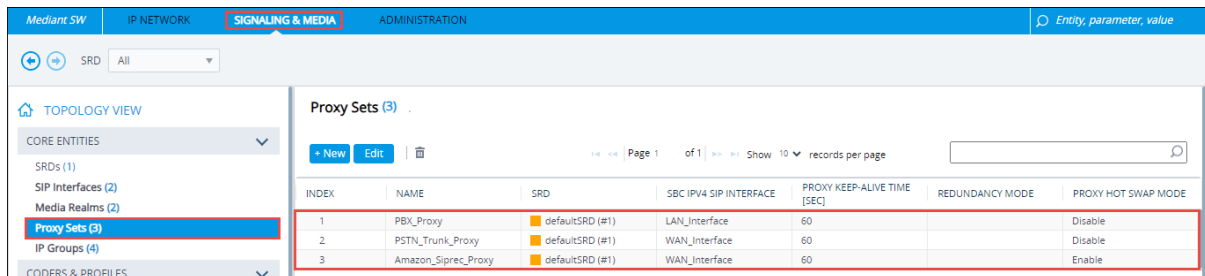
Activate Windows

Figure 24: SIP Interface WAN

4.3.5 Proxy Sets

A Proxy Set defines the address (IP address or FQDN) and transport type (e.g., UDP or TCP) of a SIP server. The Proxy Set represents the destination of the IP Group configuration entity.

Navigate to **Setup → Signaling & Media → Core Entities → Proxy Sets**



INDEX	NAME	SRD	SBC IPv4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP MODE
1	PBX_Proxy	defaultSRD (#1)	LAN_Interface	60		Disable
2	PSTN_Trunk_Proxy	defaultSRD (#1)	WAN_Interface	60		Disable
3	Amazon_Siprec_Proxy	defaultSRD (#1)	WAN_Interface	60		Enable

Figure 25: Proxy Sets List

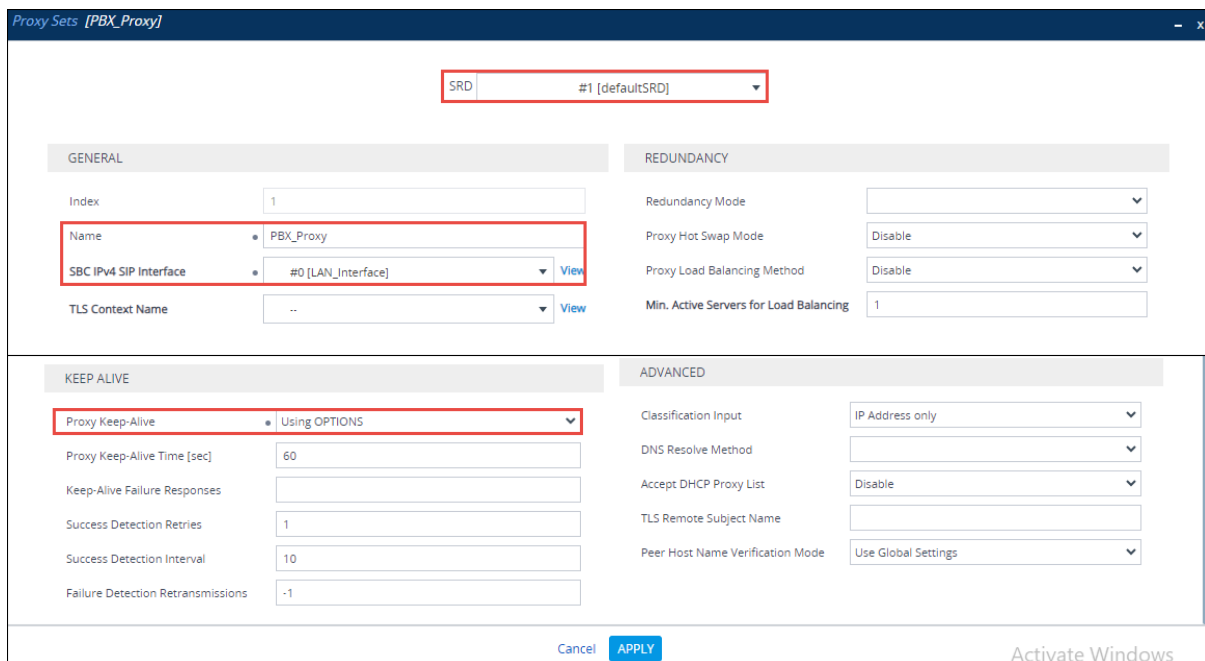
4.3.5.1 PBX Proxy Set

This proxy set points to the FreePBX Asterisk system

Name: PBX Proxy

SBC IPv4 SIP Interface: LAN Interface

Proxy Keep-Alive: Using Options



SRD: #1 [defaultSRD]

GENERAL

Index: 1

Name: PBX_Proxy

SBC IPv4 SIP Interface: #0 [LAN_Interface] View

TLS Context Name: ... View

REDUNDANCY

Redundancy Mode: [Dropdown]

Proxy Hot Swap Mode: Disable

Proxy Load Balancing Method: Disable

Min. Active Servers for Load Balancing: 1

KEEP ALIVE

Proxy Keep-Alive: Using OPTIONS

Proxy Keep-Alive Time [sec]: 60

Keep-Alive Failure Responses: [Text]

Success Detection Retries: 1

Success Detection Interval: 10

Failure Detection Retransmissions: -1

ADVANCED

Classification Input: IP Address only

DNS Resolve Method: [Dropdown]

Accept DHCP Proxy List: Disable

TLS Remote Subject Name: [Text]

Peer Host Name Verification Mode: Use Global Settings

Cancel APPLY Activate Windows

Figure 26: Proxy Set PBX

4.3.5.2 PSTN Proxy Set

This proxy set creates an outbound PSTN route for call testing and uses a third party PSTN provider.

Name: SIP Trunk Proxy

SBC IPv4 SIP Interface: WAN Interface

Proxy Keep-Alive: Using Options

The screenshot shows the configuration window for a Proxy Set named "PSTN_Trunk_Proxy". The window is divided into several sections:

- SRD:** #1 [defaultSRD]
- GENERAL:**
 - Index: 2
 - Name: SIP_Trunk_Proxy
 - SBC IPv4 SIP Interface: #1 [WAN_Interface]
 - TLS Context Name: --
- REDUNDANCY:**
 - Redundancy Mode: [Dropdown]
 - Proxy Hot Swap Mode: Disable
 - Proxy Load Balancing Method: Disable
 - Min. Active Servers for Load Balancing: 1
- KEEP ALIVE:**
 - Proxy Keep-Alive: Using OPTIONS
 - Proxy Keep-Alive Time [sec]: 60
 - Keep-Alive Failure Responses: [Empty]
 - Success Detection Retries: 1
 - Success Detection Interval: 10
 - Failure Detection Retransmissions: -1
- ADVANCED:**
 - Classification Input: IP Address only
 - DNS Resolve Method: [Dropdown]
 - Accept DHCP Proxy List: Disable
 - TLS Remote Subject Name: [Empty]
 - Peer Host Name Verification Mode: Use Global Settings

Buttons at the bottom: Cancel, APPLY, and Activate Windows.

Figure 27: Proxy Set PSTN

4.3.5.3 Amazon SIPREC Proxy Set

This proxy set points to the Amazon Chime SDK Voice Connector and is used for SIPREC transport.

Name: Amazon Siprec Proxy

SBC IPv4 SIP Interface: WAN Interface

Proxy Keep-Alive: Using Options

Proxy Hot Swap Mode: Enable

SRD #1 [defaultSRD]

GENERAL

Index: 3

Name: Amazon_Siprec_Proxy

SBC IPv4 SIP Interface: #1 [WAN_Interface] View

TLS Context Name: -- View

REDUNDANCY

Redundancy Mode: []

Proxy Hot Swap Mode: Enable

Proxy Load Balancing Method: Disable

Min. Active Servers for Load Balancing: 1

KEEP ALIVE

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

Classification Input: IP Address only

DNS Resolve Method: []

Accept DHCP Proxy List: Disable

TLS Remote Subject Name: []

Peer Host Name Verification Mode: Use Global Settings

Cancel APPLY

Activate Windows

Figure 28: Proxy Set Amazon SIPREC

4.3.6 Proxy Address

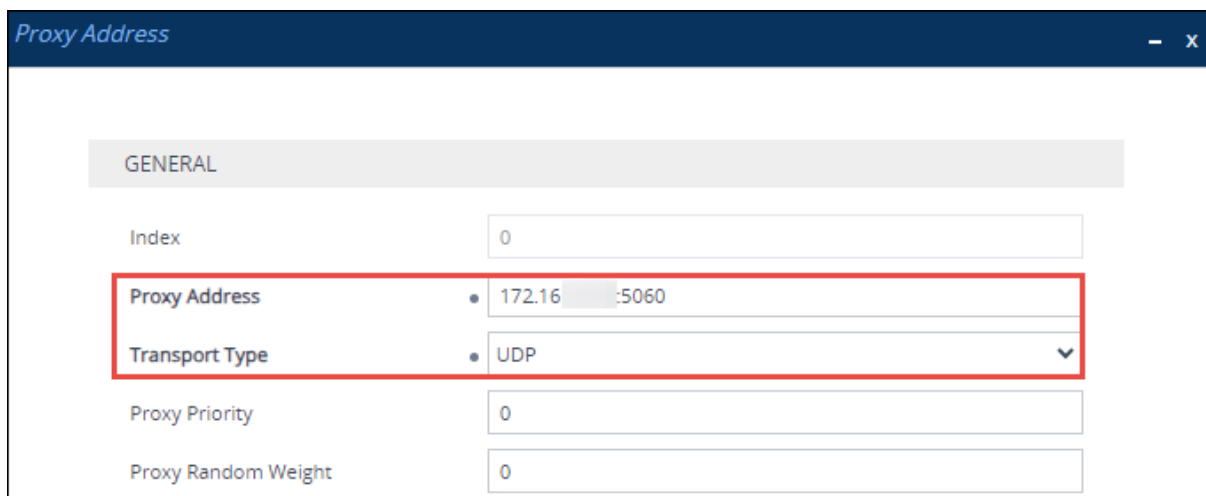
Navigate to **Setup → Signaling & Media → Core Entities → Proxy Sets**

Select the Proxy set and click proxy address at the bottom

4.3.6.1 PBX Proxy Address

Proxy Address: 172.16.X.X:5060

Transport Type: UDP



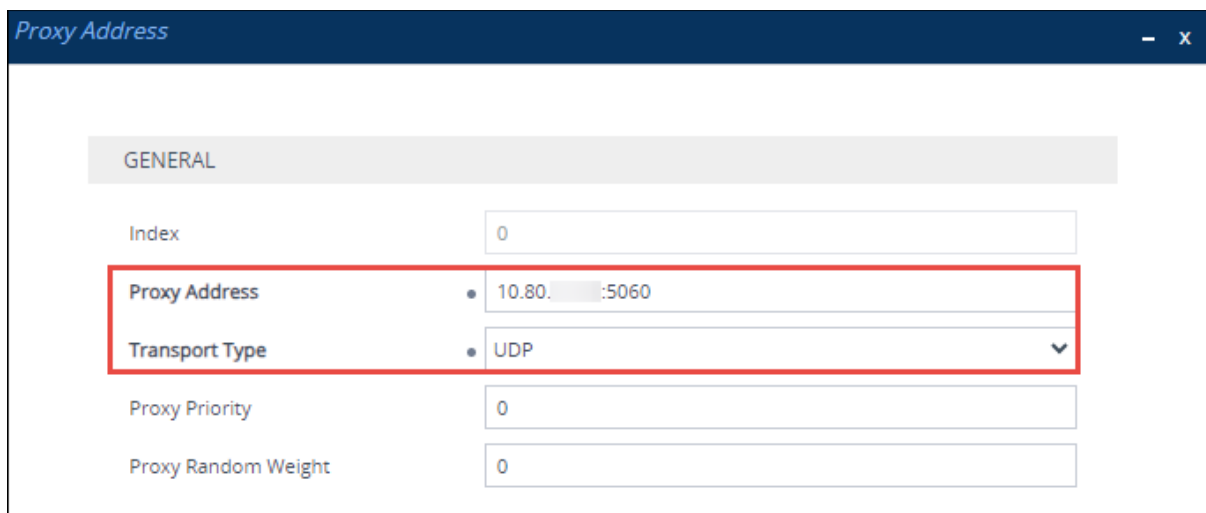
GENERAL	
Index	0
Proxy Address	172.16.X.X:5060
Transport Type	UDP
Proxy Priority	0
Proxy Random Weight	0

Figure 29: Proxy Address PBX

4.3.6.2 PSTN Proxy Address

Proxy Address: 10.80.X.X:5060

Transport Type: UDP



GENERAL	
Index	0
Proxy Address	10.80.X.X:5060
Transport Type	UDP
Proxy Priority	0
Proxy Random Weight	0

Figure 30: Proxy Address PSTN

4.3.6.3 Amazon Proxy Address

Proxy Address: gdnblgxxxxxxxxxxxxxxxxx.voiceconnector.chime.aws:5060 (FQDN of Amazon Chime SDK Voice Connector Trunk)

Transport Type: UDP

The screenshot shows a configuration window titled "Proxy Address". Under the "GENERAL" tab, the following fields are visible:

- Index: 0
- Proxy Address: gdnblgxxxxxxxxxxxxxxxxx.voiceconnector.chime.aws:5060
- Transport Type: UDP
- Proxy Priority: 0
- Proxy Random Weight: 0

The "Proxy Address" and "Transport Type" fields are highlighted with a red rectangular box.

Figure 31: Proxy Address Amazon SIPREC

4.3.7 Coder Groups

The Coders Group is configured to determine the audio (voice) coders used for calls.

Navigate to **Setup → Signaling & Media → Coders & Profiles → Coders Groups**

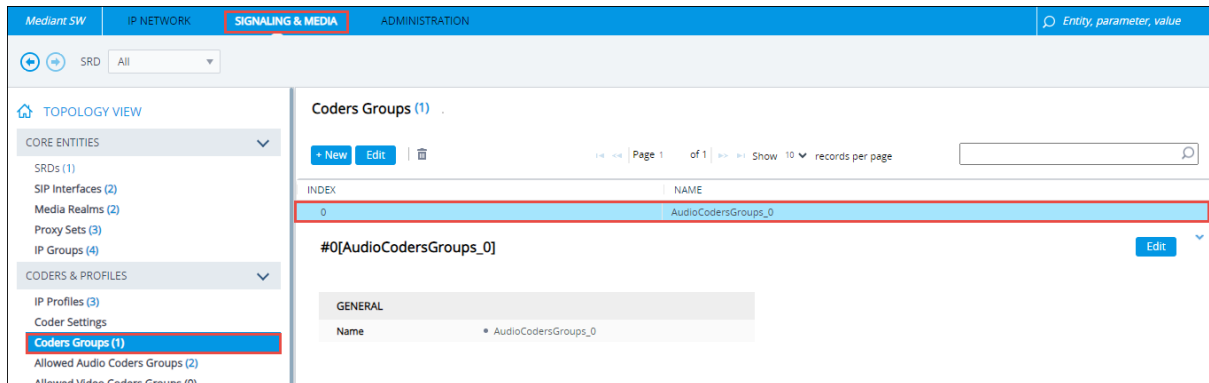


Figure 32: Coders Group

4.3.7.1 Coders Table

Enable **G711ulaw**

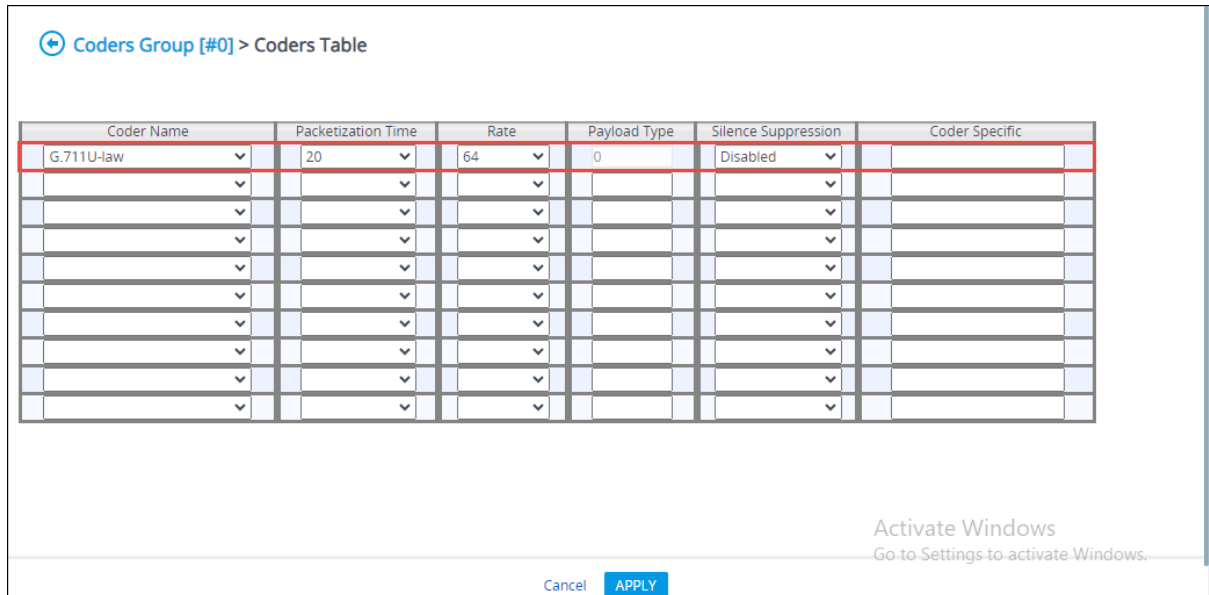
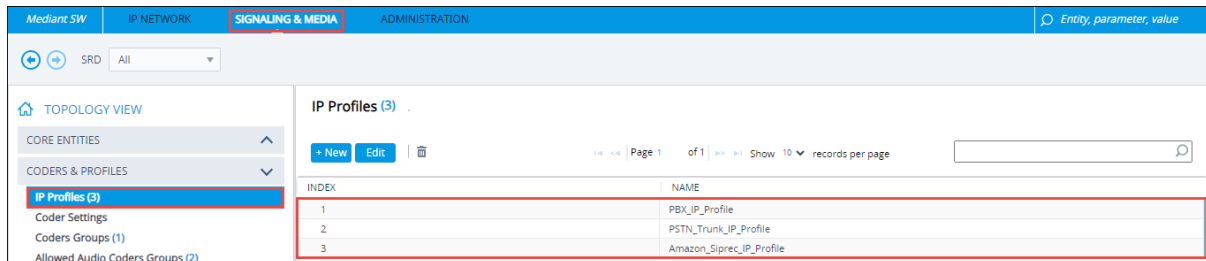


Figure 33: Coders Table

4.3.8 IP Profile

An IP Profile is a set of parameters with user-defined settings related to signaling and media (e.g., coder type).

Navigate to **Setup → Signaling & Media → Coders & Profiles → IP Profiles**



The screenshot shows the 'IP Profiles (3)' list in the Mediant SW interface. The interface includes a navigation menu on the left with 'IP Profiles (3)' selected. The main area displays a table with three rows of IP profiles. The table has columns for 'INDEX' and 'NAME'. The rows are: 1. PBX_IP_Profile, 2. PSTN_Trunk_IP_Profile, and 3. Amazon_Siprec_IP_Profile. The table also includes '+ New', 'Edit', and a trash icon. A search bar and pagination controls are visible at the top right of the table area.

INDEX	NAME
1	PBX_IP_Profile
2	PSTN_Trunk_IP_Profile
3	Amazon_Siprec_IP_Profile

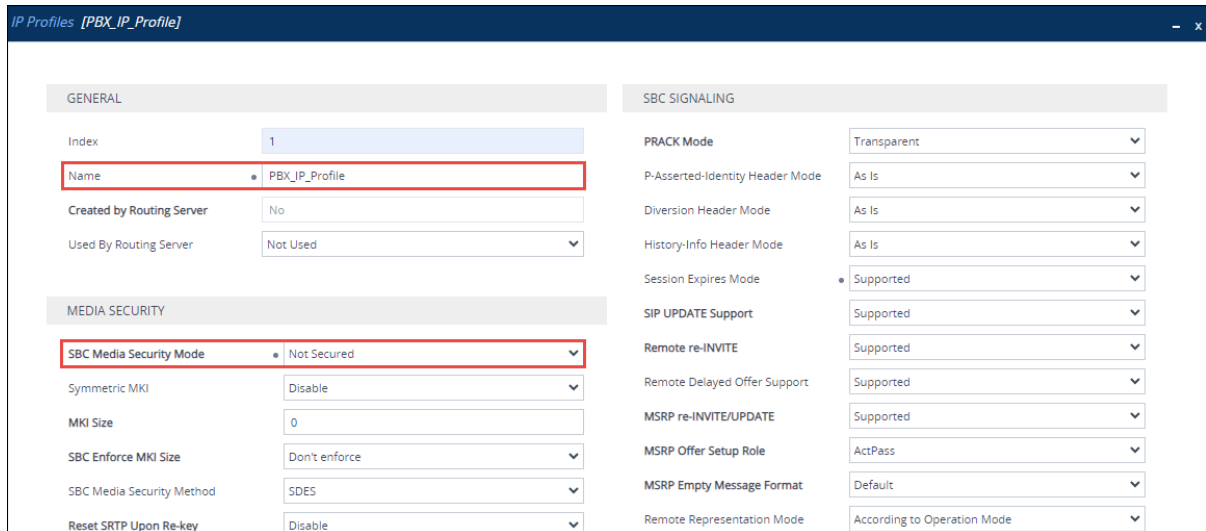
Figure 34: IP Profiles List

4.3.8.1 PBX IP Profile

Name: PBX IP Profile

SBC Media Security Mode: Not Secured

Extension Coders Group: AudioCodersGroups_0



The screenshot shows the configuration page for the 'PBX_IP_Profile' IP profile. The page is divided into two main sections: 'GENERAL' and 'SBC SIGNALING'. The 'GENERAL' section includes fields for 'Index' (1), 'Name' (PBX_IP_Profile), 'Created by Routing Server' (No), and 'Used By Routing Server' (Not Used). The 'MEDIA SECURITY' section includes 'SBC Media Security Mode' (Not Secured), 'Symmetric MKI' (Disable), 'MKI Size' (0), 'SBC Enforce MKI Size' (Don't enforce), 'SBC Media Security Method' (SDES), and 'Reset SRTP Upon Re-key' (Disable). The 'SBC SIGNALING' section includes various signaling modes such as '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), 'MSRP re-INVITE/UPDATE' (Supported), 'MSRP Offer Setup Role' (ActPass), 'MSRP Empty Message Format' (Default), and 'Remote Representation Mode' (According to Operation Mode).

Figure 35: IP Profile PBX

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 RTP Packets	As Is	ISUP Body Handling	Transparent
		ISUP Variant	Itu92
		Max Call Duration [min]	0
SBC EARLY MEDIA			
Remote Early Media	Supported	SBC REGISTRATION	
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	SBC FORWARD AND TRANSFER	
Remote Early Media RTP Detection Mode	By Signaling	Remote REFER Mode	Regular
Remote RFC 3960 Support	Not Supported	Remote Replaces Mode	Standard
Remote Can Play Ringback	Yes	Play RBT To Transferee	No
Generate RTP	None	Remote 3xx Mode	Transparent
		Send Header for Transfer	None
SBC MEDIA			
Mediation Mode	RTP Mediation	SBC HOLD	
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	..	SBC FAX	
Allowed Media Types		Fax Coders Group	..
Direct Media Tag		Fax Mode	As Is
RFC 2833 Mode	As Is	Fax Offer Mode	All coders
RFC 2833 DTMF Payload Type	0	Fax Answer Mode	Single coder
Alternative DTMF Method	As Is	Remote Renegotiate on Fax Detection	Transparent
Send Multiple DTMF Methods	Disable	Fax Rerouting Mode	Disable
Receive Multiple DTMF Methods	Disable	MEDIA	
Adapt RFC2833 BW to Voice coder BW	Disabled	Broken Connection Mode	Disconnect
SDP PTime Answer	Remote Answer	Media IP Version Preference	Only IPv4
Preferred PTime	0	RTP Redundancy Depth	Disable
Use Silence Suppression	Transparent	LOCAL TONES	
RTP Redundancy Mode	As Is	Local Ringback Tone Index	-1
RTCP Mode	Transparent	Local Held Tone Index	-1
Jitter Compensation	Disable		
ICE Mode	Disable		
SDP Handle RTCP	Don't Care		
RTCP Mux	Not Supported		
RTCP Feedback	Feedback Off		
Re-number MID	Disable		

Figure 36: IP Profile PBX Continuation

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
Remove CSRC	Disable
SBC Precondition	Not Supported

QUALITY OF SERVICE

RTP IP DiffServ	46
Signaling DiffServ	24
Data DiffServ	0

Cancel **APPLY**

Activate Windows

Figure 37: IP Profile PBX Continuation

4.3.8.2 PSTN IP Profile

Name: PSTN IP Profile

SBC Media Security Mode: Not Secured

P-Asserted-Identity Header Mode: Add

Extension Coders Group: AudioCodersGroups_0

The screenshot displays the configuration for the 'PSTN_Trunk_IP_Profile' in the AWS IAM console. The configuration is organized into several sections:

- GENERAL:** Index: 2; Name: PSTN_Trunk_IP_Profile; 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: Add; 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.
- SBC EARLY MEDIA:** Generate SRTP Keys Mode: Only If Required; SBC Remove Crypto Lifetime in SDP: No; SBC Remove Unknown Crypto: No; Crypto Suites Group: ..; Encryption on RTCP Packets: As Is.
- SBC REGISTRATION:** User Registration Time: 0; NAT UDP Registration Time: -1; NAT TCP Registration Time: -1.
- SBC FORWARD AND TRANSFER:** (Section header visible, no specific values shown).

Figure 38: IP Profile PSTN

Remote Can Play Ringback	Yes	Remote REFER Mode	Regular
Generate RTP	None	Remote Replaces Mode	Standard
SBC MEDIA		Play RBT To Transferee	No
Mediation Mode	RTP Mediation	Remote 3xx Mode	Transparent
Extension Coders Group	#0 [AudioCodersGroups_0]	Send Header for Transfer	None
Allowed Audio Coders	--	SBC HOLD	
Allowed Coders Mode	Restriction	Remote Hold Format	Transparent
Allowed Video Coders	--	Reliable Held Tone Source	Yes
Allowed Media Types		Play Held Tone	No
Direct Media Tag		SBC FAX	
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	MEDIA	
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	LOCAL TONES	
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		
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		
Remove CSRC	Disable		
SBC Precondition	Not Supported		
QUALITY OF SERVICE			
RTP IP DiffServ	46		
Signaling DiffServ	24		
Data DiffServ	0		

Cancel **APPLY**

Figure 39: IP Profile PSTN Continuation

4.3.8.3 Amazon SIPREC IP Profile

Name: Amazon Siprec IP Profile

SBC Media Security Mode: Not Secured

Extension Coders Group: AudioCodersGroups_0

The screenshot displays the configuration for the Amazon SIPREC IP Profile. The interface is organized into several sections:

- GENERAL:** Index is set to 3. Name is Amazon_Siprec_IP_Profile. Created by Routing Server is No. Used By Routing Server is Not Used.
- MEDIA SECURITY:** SBC Media Security Mode is Not Secured. Symmetric MKI is Disable. MKI Size is 0. SBC Enforce MKI Size is Don't enforce. SBC Media Security Method is SDES. Reset SRTP Upon Re-key is Disable.
- SBC EARLY MEDIA:** Remote Early Media is Supported. Remote Multiple 18x is Supported. Remote Early Media Response Type is Transparent. Remote Multiple Early Dialogs is According to Operation Mode. Remote Multiple Answers Mode is Disable. Remote Early Media RTP Detection Mode is By Signaling. Remote RFC 3960 Support is Not Supported.
- SBC MEDIA:** Mediation Mode is RTP Mediation. Extension Coders Group is #0 [AudioCodersGroups_0]. Allowed Audio Coders is .. Allowed Coders Mode is Restriction. Allowed Video Coders is .. Allowed Media Types is empty. Direct Media Tag is empty. RFC 2833 Mode is As Is. RFC 2833 DTMF Payload Type is 0. Alternative DTMF Method is As Is.
- SBC SIGNALING:** PRACK Mode is Transparent. P-Asserted-Identity Header Mode is As Is. Diversion Header Mode is As Is. History-Info Header Mode is As Is. Session Expires Mode is Transparent. SIP UPDATE Support is Supported. Remote re-INVITE is Supported. Remote Delayed Offer Support is Supported. MSRP re-INVITE/UPDATE is Supported. MSRP Offer Setup Role is ActPass. MSRP Empty Message Format is Default. Remote Representation Mode is According to Operation Mode.
- SBC REGISTRATION:** User Registration Time is 0. NAT UDP Registration Time is -1. NAT TCP Registration Time is -1.
- SBC FORWARD AND TRANSFER:** Remote REFER Mode is Regular. Remote Replaces Mode is Standard. Play RBT To Transferee is No. Remote 3xx Mode is Transparent. Send Header for Transfer is None.
- SBC HOLD:** Remote Hold Format is Transparent. Reliable Held Tone Source is Yes. Play Held Tone is No.
- SBC FAX:** Fax Coders Group is .. Fax Mode is As Is.

Figure 40: IP Profile Amazon SIPREC

Send Multiple DTMF Methods	Disable	Fax Offer Mode	All coders
Receive Multiple DTMF Methods	Disable	Fax Answer Mode	Single coder
Adapt RFC2833 BW to Voice coder BW	Disabled	Remote Renegotiate on Fax Detection	Transparent
SDP Ptime Answer	Remote Answer	Fax Rerouting Mode	Disable
Preferred PTime	0	MEDIA	
Use Silence Suppression	Transparent	Broken Connection Mode	Disconnect
RTP Redundancy Mode	As Is	Media IP Version Preference	Only IPv4
RTCP Mode	Transparent	RTP Redundancy Depth	Disable
Jitter Compensation	Disable	LOCAL TONES	
ICE Mode	Disable	Local Ringback Tone Index	-1
SDP Handle RTCP	Don't Care	Local Held Tone Index	-1
RTCP Mux	Not Supported		
RTCP Feedback	Feedback Off		
Re-number MID	Disable		
VOICE QUALITY ENHANCEMENT			
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		
Remove CSRC	Disable		
SBC Precondition	Not Supported		
QUALITY OF SERVICE			
RTP IP DiffServ	46		
Signaling DiffServ	24		
Data DiffServ	0		

Cancel **APPLY**

Figure 41: IP Profile Amazon SIPREC Continuation

4.3.9 IP Groups

An IP Group represents a SIP entity in the network with which the device communicates. This can be a server (e.g., IP PBX or ITSP) or a group of users (e.g., LAN IP phones). For servers, the address of the IP Group is typically defined by associating it with a Proxy Set.

Navigate to **Setup → Signaling & Media → Core Entities → IP Groups**

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	default	defaultSRD	Server	Not Configured	--	--	--		Enable	-1	-1
1	PBX_IP_Grp	defaultSRD	Server	Not Configured	PBX_Proxy	PBX_IP_Profile	LAN_Realm	172.16.X.X	Enable	-1	6
2	PSTN_Trunk_IP	defaultSRD	Server	Not Configured	PSTN_Trunk_Pr	PSTN_Trunk_IP	WAN_Realm	10.80.X.X	Enable	-1	7
3	Amazon_Siprec	defaultSRD	Server	Not Configured	Amazon_Siprec	Amazon_Siprec	WAN_Realm	gdnblj	Enable	-1	4

Figure 42: IP Groups List

4.3.9.1 PBX IP Group

Name: PBX IP Grp

Type: Server

Proxy Set: PBX Proxy

IP Profile: PBX IP Profile

Media Realm: LAN Realm

Outbound Message Manipulation Set: 6

SIP Group Name: 172.16.X.X (IP address of the FreePBX Asterisk)

SRD: #1 [defaultSRD]

GENERAL

Index: 1

Name: PBX_IP_Grp

Topology Location: Down

Type: Server

Proxy Set: #1 [PBX_Proxy]

IP Profile: #1 [PBX_IP_Profile]

Media Realm: #0 [LAN_Realm]

Internal Media Realm: ..

Contact User:

SIP Group Name: 172.16.X.X

QUALITY OF EXPERIENCE

QoE Profile: --

Bandwidth Profile: --

User Voice Quality Report: Disable

MESSAGE MANIPULATION

Inbound Message Manipulation Set: -1

Outbound Message Manipulation Set: 6

Message Manipulation User-Defined String 1:

Message Manipulation User-Defined String 2:

Proxy Keep-Alive using IP Group settings: Disable

Figure 43: IP Group PBX

Created By Routing Server	No	SBC REGISTRATION AND AUTHENTICATION	
Used By Routing Server	Not Used	Max. Number of Registered Users	-1
Proxy Set Connectivity	Connected	Registration Mode	User Initiates Registration
SBC GENERAL		Dedicated Connection Mode	Disable
Classify By Proxy Set	Enable	User Stickiness	Disable
Validate Source IP	Disable	User UDP Port Assignment	Disable
SBC Operation Mode	Not Configured	Authentication Mode	User Authenticates
SBC Client Forking Mode	Sequential	Authentication Method List	
CAC Profile	--	SBC Server Authentication Type	According to Global Parameter
SIP Source Host Name		OAuth HTTP Service	--
		Username As Client	
ADVANCED		Password As Client	
Local Host Name		Username As Server	
UII Format	Disable	Password As Server	
Always Use Src Address	No	GW GROUP STATUS	
SBC ADVANCED		GW Group Registered IP Address	
Source URI Input		GW Group Registered Status	NA
Destination URI Input			
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	--		
Teams Local Media Optimization Handling	None		
Teams Local Media Optimization Initial Behavior	DirectMedia		
Teams Local Media Optimization Site			
Teams Local Media Optimization Sync	Disable		
Teams Direct Routing Mode	Disable		
Metering Remote Type	Regular		
Report Metering	Enable		

Cancel **APPLY** Activate Windows

Figure 44: IP group PBX Continuation

4.3.9.2 PSTN IP Group

Name: PSTN IP Grp

Type: Server

Proxy Set: PSTN Trunk Proxy

IP Profile: PSTN Trunk IP Profile

Media Realm: WAN Realm

Outbound Message Manipulation Set: 7

SIP Group Name: 10.80.X.X (IP address of the PSTN Trunk)

The screenshot displays the configuration page for an IP Group PSTN in the AWS IAM console. The page is titled "IP Groups [PSTN_Trunk_IP_Grp]". At the top, there is a dropdown menu for "SRD" set to "#1 [defaultSRD]". The configuration is organized into several sections:

- GENERAL:** Includes fields for Index (2), Name (PSTN_Trunk_IP_Grp), Topology Location (Up), Type (Server), Proxy Set (#2 [PSTN_Trunk_Proxy]), IP Profile (#2 [PSTN_Trunk_IP_Profile]), Media Realm (#1 [WAN_Realm]), Internal Media Realm (--), Contact User, and SIP Group Name (10.80.X.X).
- QUALITY OF EXPERIENCE:** Includes QoS Profile (--), Bandwidth Profile (--), and User Voice Quality Report (Disable).
- MESSAGE MANIPULATION:** Includes Inbound Message Manipulation Set (-1) and Outbound Message Manipulation Set (7). It also has fields for Message Manipulation User-Defined String 1 and 2, and Proxy Keep-Alive using IP Group settings (Enable).
- SBC REGISTRATION AND AUTHENTICATION:** Includes 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), Authentication Method List, SBC Server Authentication Type (According to Global Parameter), OAuth HTTP Service (--), and Username As Client.

Several fields are highlighted with red boxes in the original image: SRD, Name, Proxy Set, IP Profile, Media Realm, SIP Group Name, and Outbound Message Manipulation Set.

Figure 45: IP Group PSTN

ADVANCED		Password As Client <input type="text"/> Username As Server <input type="text"/> Password As Server <input type="text"/>
Local Host Name	<input type="text"/>	
UII Format	Disable	
Always Use Src Address	No	
SBC ADVANCED		GW GROUP STATUS
Source URI Input	<input type="text"/>	GW Group Registered IP Address <input type="text"/>
Destination URI Input	<input type="text"/>	GW Group Registered Status <input type="text" value="NA"/>
SIP Connect	No	
SBC PSAP Mode	Disable	
Route Using Request URI Port	Disable	
Media TLS Context	#0 [Default] View	
Keep Original Call-ID	No	
Dial Plan	-- View	
Call Setup Rules Set ID	-1	
Tags	<input type="text"/>	
SBC Alternative Routing Reasons Set	-- View	
Teams Local Media Optimization Handling	None	
Teams Local Media Optimization Initial Behavior	DirectMedia	
Teams Local Media Optimization Site	<input type="text"/>	
Teams Local Media Optimization Sync	Disable	
Teams Direct Routing Mode	Disable	
Metering Remote Type	Regular	
Report Metering	Enable	
Cancel APPLY		Activate Windows

Figure 46: IP Group PSTN Continuation

4.3.9.3 Amazon SIPREC IP Group

Name: Amazon Siprec IP Grp

Type: Server

Proxy Set: Amazon Siprec Proxy

IP Profile: Amazon Siprec IP Profile

Media Realm: WAN Realm

Outbound Message Manipulation Set: 4

SIP Group Name: gdnblgxxxxxxxxxxxxxxxxx.voiceconnector.chime.aws (FQDN of Amazon Chime SDK Voice Connector Trunk)

The screenshot displays the configuration interface for an Amazon SIPREC IP Group. At the top, a dropdown menu is set to 'SRD #1 [defaultSRD]'. The configuration is organized into several sections:

- GENERAL:** Index (3), Name (Amazon_Siprec_IP_Grp), Topology Location (Up), Type (Server), Proxy Set (#3 [Amazon_Siprec_Proxy]), IP Profile (#3 [Amazon_Siprec_IP_Profile]), Media Realm (#1 [WAN_Realm]), Internal Media Realm (--), Contact User, and SIP Group Name (gdnblg:).
- QUALITY OF EXPERIENCE:** QoE Profile, Bandwidth Profile, and User Voice Quality Report (Disable).
- MESSAGE MANIPULATION:** Inbound Message Manipulation Set (-1) and Outbound Message Manipulation Set (4). It also includes Message Manipulation User-Defined String 1 and 2, and Proxy Keep-Alive using IP Group settings (Enable).
- SBC REGISTRATION AND AUTHENTICATION:** 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), Authentication Method List, SBC Server Authentication Type (According to Global Parameter), OAuth HTTP Service (--), and Username As Client.
- SBC GENERAL:** Classify By Proxy Set (Enable), Validate Source IP (Disable), SBC Operation Mode (Not Configured), SBC Client Forking Mode (Sequential), CAC Profile (--), and SIP Source Host Name.

Figure 47: IP Group Amazon Siprec

ADVANCED		Password As Client <input type="text"/> Username As Server <input type="text"/> Password As Server <input type="text"/>
Local Host Name	<input type="text"/>	
UII Format	Disable ▾	
Always Use Src Address	No ▾	
SBC ADVANCED		GW GROUP STATUS
Source URI Input	<input type="text"/> ▾	GW Group Registered IP Address <input type="text"/>
Destination URI Input	<input type="text"/> ▾	GW Group Registered Status <input type="text" value="NA"/>
SIP Connect	No ▾	
SBC PSAP Mode	Disable ▾	
Route Using Request URI Port	Disable ▾	
Media TLS Context	#0 [Default] ▾ View	
Keep Original Call-ID	No ▾	
Dial Plan	-- ▾ View	
Call Setup Rules Set ID	-1	
Tags	<input type="text"/>	
SBC Alternative Routing Reasons Set	-- ▾ View	
Teams Local Media Optimization Handling	None ▾	
Teams Local Media Optimization Initial Behavior	DirectMedia ▾	
Teams Local Media Optimization Site	<input type="text"/>	
Teams Local Media Optimization Sync	Disable ▾	
Teams Direct Routing Mode	Disable ▾	
Metering Remote Type	Regular ▾	
Report Metering	Enable ▾	
Cancel APPLY		Activate Windows

Figure 48: IP Group Amazon SIPREC Continuation

4.3.10 IP-to-IP Routing

The IP-to-IP routing rules are used to define the routes for forwarding SIP messages received from one IP entity to another.

Navigate to **Setup → Signaling & Media → SBC → Routing → IP-to-IP Routing**

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	Terminate OPT	defaultSBCRou	Route Row	Any	OPTIONS	*	*	Internal	--	--	
1	PBX_to_PSTN	defaultSBCRou	Route Row	PBX_IP_Grp	All	*	*	IP Group	PSTN_Trunk_IP	--	
2	PSTN_to_PBX	defaultSBCRou	Route Row	PSTN_Trunk_IP	All	*	*	IP Group	PBX_IP_Grp	--	

Figure 49: IP-to-IP Routing List

4.3.10.1 Terminate Options

Source IP Group: Any

Request Type: OPTIONS

Destination Type: Internal

Internal Action: Reply(response='200')

Routing Policy: #0 [defaultSBCRoutingPolicy]

GENERAL

Index: 0

Name: Terminate OPTIONS

Alternative Route Options: Route Row

MATCH

Source IP Group: Any

Request Type: OPTIONS

Source Username Pattern: *

Source Host: *

Source Tag:

Destination Username Pattern: *

Destination Host: *

Destination Tag:

Message Condition: --

Call Trigger: Any

ReRoute IP Group: Any

ACTION

Destination Type: Internal

Destination IP Group: --

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

Internal Action: Reply(Response='200')

Modified Destination User Name:

Buttons: Cancel, APPLY, Activate Windows

Figure 50: Terminate OPTIONS

4.3.10.2 PBX to PSTN

Name: PBX to PSTN

Source IP Group: PBX IP Grp

Destination Type: IP Group

Destination IP Group: PSTN Trunk IP Grp

IP-to-IP Routing [PBX_to_PSTN]

Routing Policy: #0 [defaultSBCRoutingPolicy]

GENERAL	ACTION
Index: 1	Destination Type: IP Group
Name: PBX_to_PSTN	Destination IP Group: #2 [PSTN_Trunk_IP_Grp]
Alternative Route Options: Route Row	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
	Internal Action:
	Modified Destination User Name:

Cancel APPLY Activate Windows

Figure 51: Routing PBX to PSTN

4.3.10.3 PSTN to PBX

Name: PSTN to PBX

Source IP Group: PSTN Trunk IP Grp

Destination Type: IP Group

Destination IP Group: PBX IP Grp

The screenshot shows the configuration page for an IP-to-IP Routing policy. The title bar reads "IP-to-IP Routing [PSTN_to_PBX]". At the top, the "Routing Policy" is set to "#0 [defaultSBCRoutingPolicy]".

The configuration is divided into several sections:

- GENERAL:** Index is 2. Name is "PSTN_to_PBX". Alternative Route Options is "Route Row".
- MATCH:** Source IP Group is "#2 [PSTN_Trunk_IP_Grp]". Request Type is "All". Source Username Pattern is "*". Source Host is "*".
- ACTION:** Destination Type is "IP Group". Destination IP Group is "#1 [PBX_IP_Grp]". Destination SIP Interface is "--". Destination Address is empty. Destination Port is "0". Destination Transport Type is "--". IP Group Set is "--". Call Setup Rules Set ID is "-1". Group Policy is "Sequential".
- Other Fields:** Source Tag, Destination Username Pattern (*), Destination Host (*), Destination Tag, Message Condition (--), Call Trigger (Any), ReRoute IP Group (Any), Cost Group (--), Routing Tag Name (default), Internal Action (Editor), and Modified Destination User Name.

At the bottom, there are "Cancel" and "APPLY" buttons. An "Activate Windows" watermark is visible in the bottom right corner.

Figure 52: Routing PSTN to PBX

4.3.11 SIP Recording

This section describes SBC's SIP Recording configuration.

Navigate to **Setup** → **Signaling & Media** → **SIP Recording** → **SIP Recording Settings**

4.3.11.1 SIP Recording Settings

Recording Server (SRS) Destination Username: gdnblgxxxxxxxxxxxxxxxxx.voice.connector.aws (FQDN of Amazon Chime SDK Voice Connector Trunk)

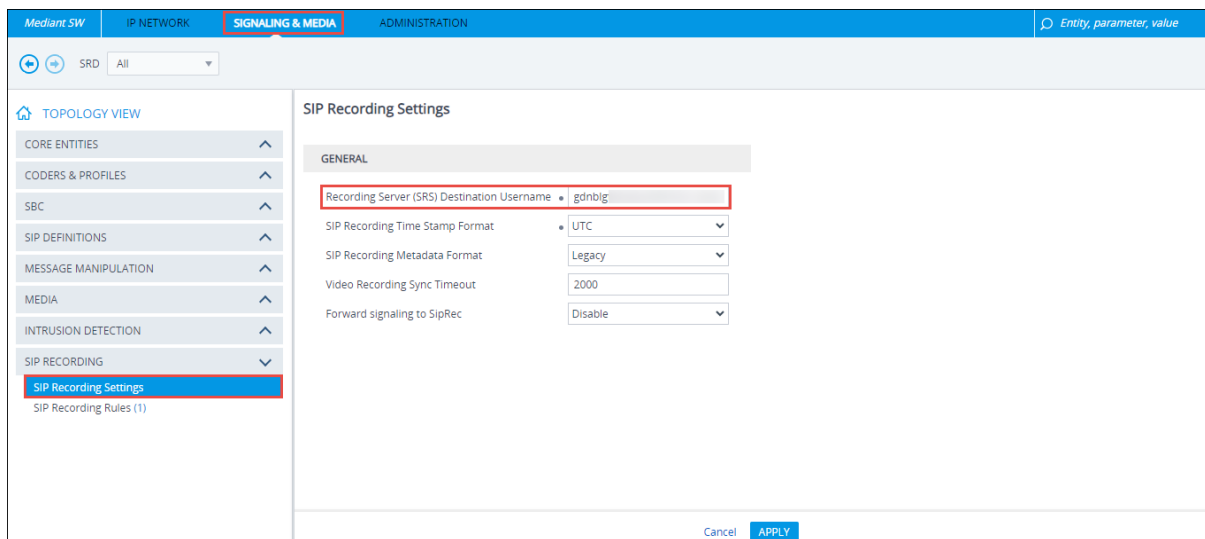


Figure 53: Amazon SIP Recording Settings

4.3.11.2 SIP Recording Rules

Recorded IP Group: PSTN Trunk IP Group

Peer IP Group: PBX IP Group

Recording Server (SRS) IP Group: Amazon SIPREC IP Group

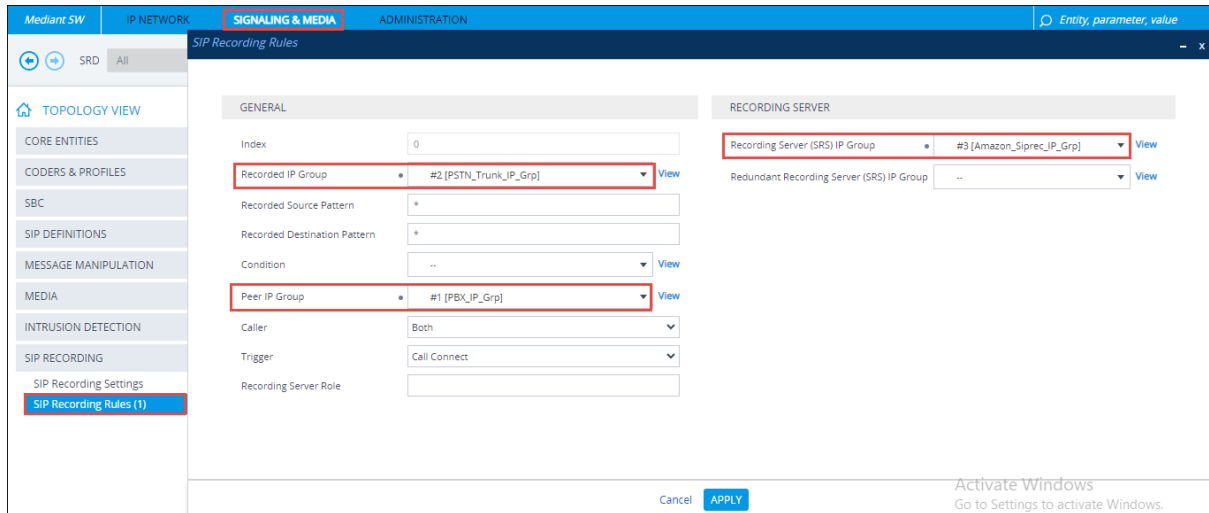


Figure 54: SIP Recording Rules

4.3.12 TLS Configuration

This section describes configuring the SBC to establish a secure SIP TLS connection with Amazon Chime SDK Voice Connector.

4.3.12.1 NTP Server Address

Navigate to **Setup** → **Administration** → **Time & Date**

Primary NTP Server Address: 10.10.X.X

The screenshot displays the 'Time & Date' configuration page in the Mediant SW Administration console. The page is divided into several sections:

- TIME:** Shows Local Time (22 Sep, 2023 00:57:43) and UTC Time (22 Sep, 2023 05:57:43). The Time Synchronization Source is set to NTP.
- SET TIME:** Includes a 'Set Local Time' button.
- NTP SERVER:** Contains the following settings:
 - Enable NTP: Enable
 - NTP Interface: eth0
 - Primary NTP Server Address (IP or FQDN): 10.10.X.X (highlighted with a red box)
- TIME ZONE:** Includes settings for UTC Offset (Hours: -5, Minutes: 0), Daylight Saving Time (Disable), DST Mode (Day of year), Start Time, End Time, Offset (min) (60), Day of Month Start, and Day of Month End.
- DATE HEADER TIME SYNC:** Includes a 'Synchronize Time from SIP Date Header' checkbox (checked) and a 'Disable' button.

The 'ADMINISTRATION' tab is selected in the top navigation bar. The 'TIME & DATE' section is highlighted in the left sidebar. The 'Primary NTP Server Address' field is highlighted with a red box. The 'APPLY' button is visible at the bottom right.

Figure 55: NTP Server

4.3.12.2 TLS Context

Navigate to **Setup → IP Network → Security → TLS Contexts**

Name: Enter a name for the TLS Context

TLS version: TLS v1.2

The screenshot shows the AWS IAM console configuration for a TLS Context. The 'Name' field is set to 'AWS_Siprec_TLS' and the 'TLS Version' is set to 'TLSv1.2'. Other settings include 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), Cipher Client TLS 1.3 (TLS_AES_256_GCM_SHA384:TLS_CHACHA20_POLY1305), Key Exchange Groups (X25519:P-256:P-384:X448), Strict Certificate Extension Validation (Disable), DH key Size (2048), TLS Renegotiation (Enable), and Use default CA Bundle (Disable). The OSCP settings are also visible on the right side of the form.

Figure 56: TLS Context

4.3.12.3 Trusted Root Certificates

- 1) Select the TLS Context
- 2) Click **Trusted Root Certificates** located at the bottom of the TLS Contexts page
- 3) Click Import
- 4) Upload the Amazon Chime root certificate
- 5) The Amazon Chime Root Certificate can be downloaded from the Amazon Chime SDK Voice Connector account.

The screenshot shows the 'Trusted Root Certificates' page for a TLS Context. The table lists various root certificates, including Amazon Root CA 1-4, Baltimore CyberTrust Root, Cybertrust Global Root, and DigiCert Assured ID Root CA G2-G3. The 'Import' button is highlighted with a red box.

INDEX	SUBJECT	ISSUER	EXPIRES
0	Amazon Root CA 1	Amazon Root CA 1	Sat, 16 Jan 2038 18:30:00 GMT
1	Amazon Root CA 2	Amazon Root CA 2	Fri, 25 May 2040 18:30:00 GMT
2	Amazon Root CA 3	Amazon Root CA 3	Fri, 25 May 2040 18:30:00 GMT
3	Amazon Root CA 4	Amazon Root CA 4	Fri, 25 May 2040 18:30:00 GMT
4	Baltimore CyberTrust Root	Baltimore CyberTrust Root	Mon, 12 May 2025 18:29:00 GMT
5	Cybertrust Global Root	Cybertrust Global Root	Wed, 15 Dec 2021 02:30:00 GMT
6	DigiCert Assured ID Root CA	DigiCert Assured ID Root CA	Sun, 09 Nov 2031 18:30:00 GMT
7	DigiCert Assured ID Root G2	DigiCert Assured ID Root G2	Fri, 15 Jan 2038 06:30:00 GMT
8	DigiCert Assured ID Root G3	DigiCert Assured ID Root G3	Fri, 15 Jan 2038 06:30:00 GMT
9	DigiCert Global Root CA	DigiCert Global Root CA	Sun, 09 Nov 2031 18:30:00 GMT

Figure 57: Amazon Trusted Root Certificates

4.3.12.4 SRTP

Navigate to **Setup → Signaling & Media → Media → Media Security**
Enable the **Media Security**

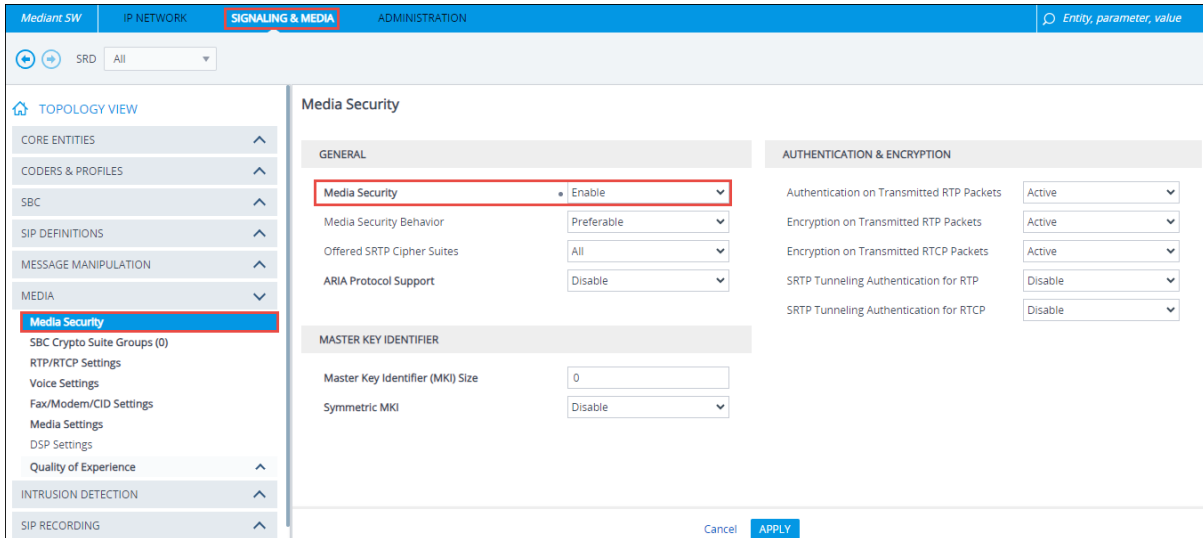


Figure 58: Media Security

4.3.12.5 Amazon SIPREC IP Profile-TLS

Navigate to **Setup → Signaling & Media → Coders & Profiles → IP Profiles**

1) Select **Amazon Siprec IP Profile**

SBC Media Security Mode: Secured

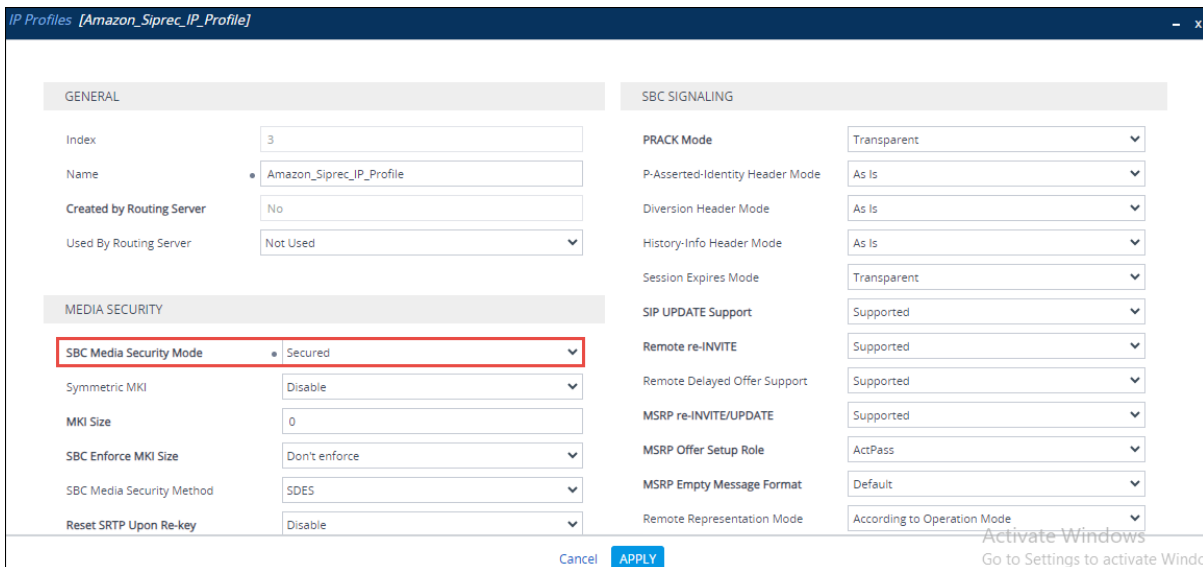


Figure 59: IP Profile TLS

4.3.12.6 Proxy Set -TLS

Navigate to **Setup → Signaling & Media → Core Entities → Proxy sets**

- 1) Select Amazon Siprec Proxy

TLS Context Name: Select the TLS Context created

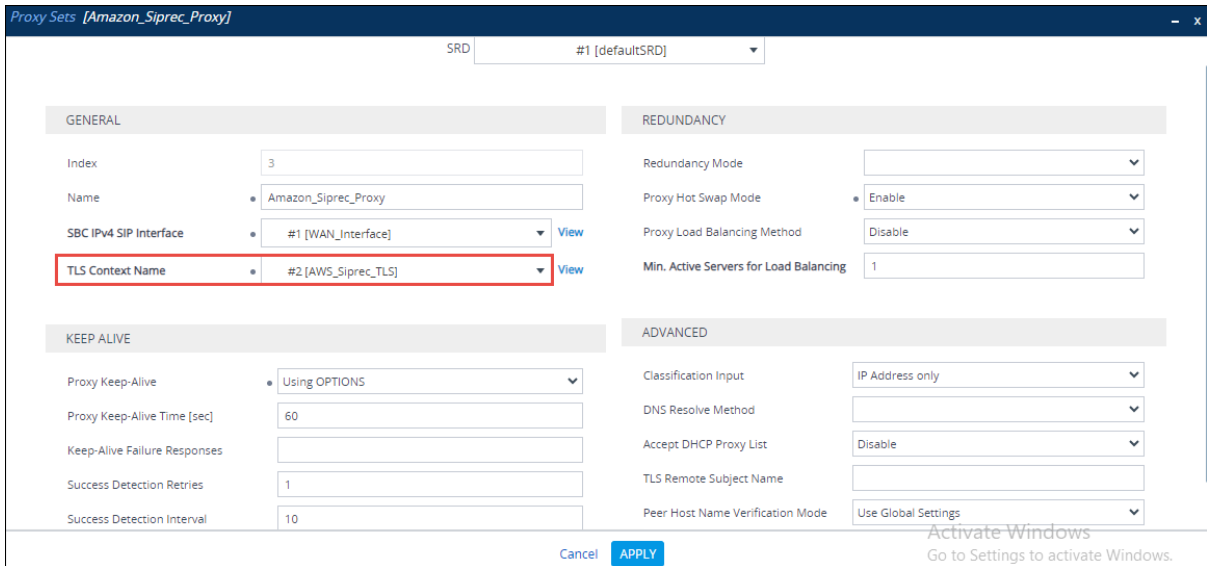


Figure 60: Proxy Set TLS

4.3.12.7 Proxy Address-TLS

- 1) Select the Amazon Siprec Proxy Set
- 2) Click the Proxy address located at the bottom of the proxy set page

Proxy Address: gdnblgxxxxxxxxxxxxxxxxx.voiceconnector.chime.aws:5061

Transport Type: TLS

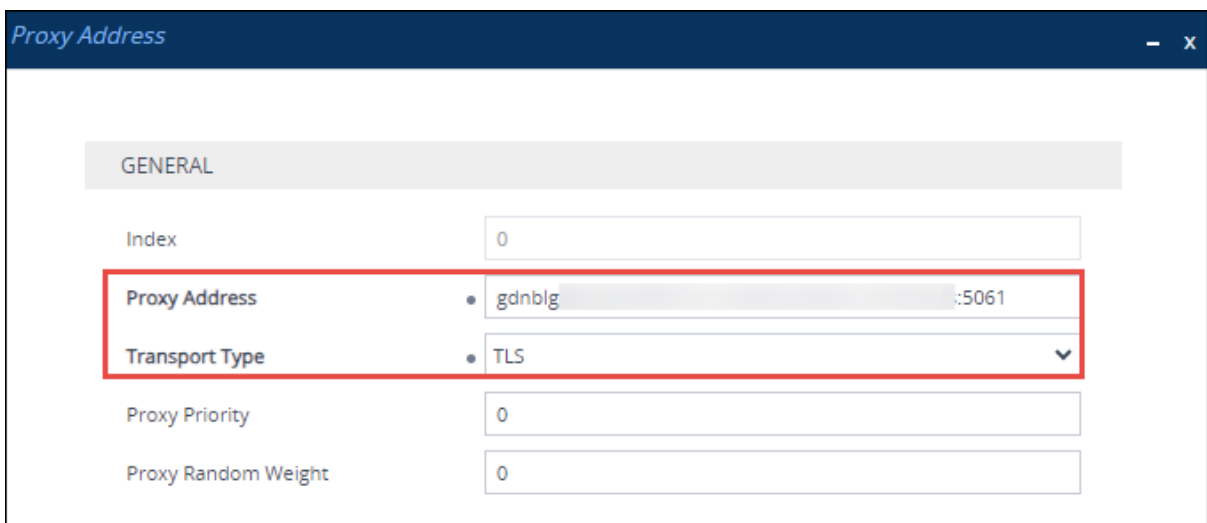


Figure 61: Proxy Address TLS

4.3.13 Number Manipulation

Number manipulation is configured to manipulate the SIP Request-URI user part (source or destination number). It uses the configured IP Groups to denote the source and destination of the call.

Navigate to **Setup → Signaling & Media → SBC → Manipulation**

INDEX	NAME	ROUTING POLICY	ADDITIONAL MANIPULATION	MANIPULATION PURPOSE	SOURCE IP GROUP	SOURCE USERNAME PATTERN	DESTINATION USERNAME PATTERN	MANIPULATED ITEM	REMOVE FROM LEFT	REMOVE FROM RIGHT	LEAVE FROM RIGHT	PREFIX TO ADD	SUFFIX TO ADD
0	Towards_PBX	defaultSBCR	No	Normal	PSTN_Trunk	*	*	Destination	6	0	255		

Figure 62: Number Manipulation

4.3.13.1 Inbound Manipulation

Name: Towards PBX

Source IP Group: PSTN Trunk IP Grp

Manipulated Item: Destination

Remove From Left: 6

Routing Policy: #0 [defaultSBCRoutingPolicy]

GENERAL

Index: 0

Name: Towards_PBX

Additional Manipulation: No

Manipulation Purpose: Normal

MATCH

Request Type: All

Source IP Group: #2 [PSTN_Trunk_IP_Grp] View

Source Username Pattern: *

Source Host: *

Destination Username Pattern: *

ACTION

Manipulated Item: Destination

Remove From Left: 6

Remove From Right: 0

Leave From Right: 255

Prefix to Add:

Suffix to Add:

Cancel APPLY

Figure 63: Inbound Manipulation PBX

4.3.13.2 Outbound Manipulation

Name: Towards PSTN

Source IP Group: PBX IP Grp

Destination IP Group: PSTN Trunk IP Grp

Manipulated Item: Source URI

Prefix to Add: 97XXX8

The screenshot shows the 'Outbound Manipulations' configuration window for a policy named 'Towards_PSTN'. The window is divided into several sections: GENERAL, MATCH, and ACTION. The 'Routing Policy' is set to '#0 [defaultSBCRoutingPolicy]'. In the GENERAL section, the Name is 'Towards_PSTN', Additional Manipulation is 'No', and Call Trigger is 'Any'. The MATCH section includes Request Type set to 'All', Source IP Group set to '#1 [PBX_IP_Grp]', and Destination IP Group set to '#2 [PSTN_Trunk_IP_Grp]'. The ACTION section shows Manipulated Item set to 'Source URI', Remove From Left set to '0', Remove From Right set to '0', Leave From Right set to '255', Prefix to Add set to '97XXX8', Suffix to Add is empty, and Privacy Restriction Mode is 'Transparent'. At the bottom, there are 'Cancel' and 'APPLY' buttons.

Figure 64: Outbound Manipulation PSTN

4.3.14 Message Manipulation Configuration

SIP message manipulation rules are created to modify, insert and/or remove the SIP headers. The manipulation Set ID has to be assigned to the relevant IP Groups.

Navigate to **Setup → Signaling & Media → Message Manipulation → Message Manipulations**

SIP Message Manipulation for PBX

(This manipulation is to replace the IP address in the From Header of the INVITE request with the AudioCodes SBC IP address)

```
message message-manipulations 9 (Optional)
  manipulation-name "Asterisk_From_hdr"
  manipulation-set-id 6
  message-type "Invite.Request"
  action-subject "Header.From.URL.Host"
  action-type modify
  action-value "'10.64.X.X'"
  activate
  exit
```

(This manipulation is to replace the IP address in the P-asserted Identity Header of the INVITE request with the AudioCodes SBC IP address)

```
message message-manipulations 10 (Optional)
  manipulation-name "Asterisk_PAI_hdr_host"
  manipulation-set-id 6
  message-type "Invite.Request"
  action-subject "Header.P-Asserted-Identity.URL.Host"
  action-type modify
  action-value "'10.64.X.X'"
  activate
  exit
```

(This manipulation is to replace the IP address in the From Header of the ACK with the AudioCodes SBC IP address)

```
message message-manipulations 13 (Optional)
  manipulation-name "Asterisk_Ack"
  manipulation-set-id 6
```

```
message-type "Ack.Request"
action-subject "Header.From.URL.Host"
action-type modify
action-value "'10.64.X.X'"
activate
exit
```

(This manipulation is to replace the IP address in the From Header of the BYE request with the AudioCodes SBC IP address)

```
message message-manipulations 14 (Optional)
manipulation-name "Asterisk_Bye"
manipulation-set-id 6
message-type "Bye.Request"
action-subject "Header.From.URL.Host"
action-type modify
action-value "'10.64.X.X'"
activate
exit
```

(This manipulation is to replace the IP address in the P-Asserted Identity Header of the BYE request with the AudioCodes SBC IP address)

```
message message-manipulations 16 (Optional)
manipulation-name "Asterisk_Bye_PAI_hdr"
manipulation-set-id 6
message-type "Bye.Request"
action-subject "Header.P-Asserted-Identity.URL.Host"
action-type modify
action-value "'10.64.X.X'"
activate
exit
```

(This manipulation is to replace the IP address in the To Header of the response for UPDATE with the AudioCodes SBC IP address)

```
message message-manipulations 17 (Optional)
manipulation-name "Asterisk_Update_Res"
manipulation-set-id 6
message-type "Update.Response"
action-subject "Header.To.URL.Host"
```

```
action-type modify
action-value ""10.64.X.X""
activate
exit
```

(This manipulation is to replace the IP address in the To Header of the response for BYE request with the AudioCodes SBC IP address)

```
message message-manipulations 22 (Optional)
```

```
manipulation-name "Asterisk_Bye_res"
manipulation-set-id 6
message-type "Bye.Response"
action-subject "Header.To.URL.Host"
action-type modify
action-value ""10.64.X.X""
activate
exit
```

SIP Message Manipulation for PSTN

(This manipulation is to replace the IP address in the From Header of the INVITE request with the AudioCodes SBC IP address)

```
message message-manipulations 11 (Optional)
  manipulation-name "Pstn_From_hdr"
  manipulation-set-id 7
  message-type "Invite.Request"
  action-subject "Header.From.URL.Host"
  action-type modify
  action-value "'10.80.X.X'"
  activate
  exit
```

(This manipulation is to replace the IP address in the P-asserted Identity Header of the INVITE request with the AudioCodes SBC IP address)

```
message message-manipulations 12 (Optional)
  manipulation-name "Pstn_PAI_Hdr"
  manipulation-set-id 7
  message-type "Invite.Request"
  action-subject "Header.P-Asserted-Identity.URL.Host"
  action-type modify
  action-value "'10.80.X.X'"
  activate
  exit
```

(This manipulation is to replace the IP address in the From Header of the ACK with the AudioCodes SBC IP address)

```
message message-manipulations 15 (Optional)
  manipulation-name "Pstn_ACK"
  manipulation-set-id 7
  message-type "Ack.Request"
  action-subject "Header.From.URL.Host"
  action-type modify
  action-value "'10.80.X.X'"
  activate
  exit
```

(This manipulation is to replace the IP address in the P-asserted Identity Header of the UPDATE request with the AudioCodes SBC IP address)

message message-manipulations 18 (Optional)

```
manipulation-name "Pstn_Update_PAI"  
manipulation-set-id 7  
message-type "Update.Request"  
action-subject "Header.P-Asserted-Identity.URL.Host"  
action-type modify  
action-value "'10.80.X.X'"  
activate  
exit
```

(This manipulation is to replace the IP address in the From Header of the BYE request with the AudioCodes SBC IP address)

message message-manipulations 23 (Optional)

```
manipulation-name "Pstn_bye_Req"  
manipulation-set-id 7  
message-type "Bye.Request"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value "'10.80.X.X'"  
activate  
exit
```

(This manipulation is to replace the IP address in the From Header of the UPDATE request with the AudioCodes SBC IP address)

message message-manipulations 24 (Optional)

```
manipulation-name "Pstn_Update_From"  
manipulation-set-id 7  
message-type "Update.Request"  
action-subject "Header.From.URL.Host"  
action-type modify  
action-value "'10.80.X.X'"  
activate  
exit
```

(This manipulation is to replace the IP address in the To Header of the response for BYE request with the AudioCodes SBC IP address)

message message-manipulations 25 (Optional)

manipulation-name "Pstn_bye_res"

manipulation-set-id 7

message-type "Bye.Response"

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

action-type modify

action-value "'10.80.X.X'"

activate

exit

SIP Message Manipulation for Amazon Chime SDK Voice Connector SIPREC

(This manipulation is to append '+sip.src' to the Contact Header of the INVITE request)

message message-manipulations 0 (Mandatory)

```
manipulation-name "AWS_siprec1"
manipulation-set-id 4
message-type "Invite.Request"
condition "Header.Contact regex (.*)(>);(.*)"
action-subject "Header.Contact"
action-type modify
action-value "$1+$2+'+sip.src'"
activate
exit
```

(This manipulation is to append the calling party's number to the From Header of the INVITE request)

message message-manipulations 1 (Optional)

```
manipulation-name "AWS_siprec_2"
manipulation-set-id 4
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
```

(This manipulation is to append the called party's number to the Req-Url of the INVITE request)

message message-manipulations 2 (Optional)

```
manipulation-name "AWS_siprec3"
manipulation-set-id 4
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
```

(This manipulation is to append the called party's number to the To Header of the INVITE request)

```
message message-manipulations 3 (Optional)
```

```
  manipulation-name "AWS_siprec4"  
  manipulation-set-id 4  
  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
```

5 Sample SIPREC Trace

This section contains the Sample Siprec Trace between SBC and Amazon Chime SDK Voice Connector with meta-data information

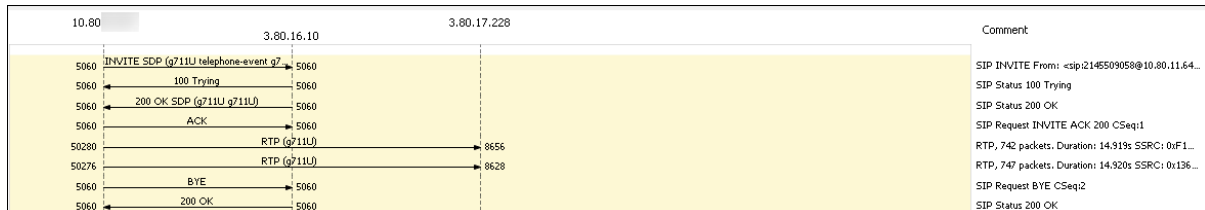


Figure 65: SIPREC sample trace

INVITE sip:0072@gdnblgttwnpzuvf4h1hj1.voiceconnector.chime.aws;user=phone SIP/2.0

Via: SIP/2.0/UDP 10.80.X.X:5060;branch=z9hG4bKac757879020

Max-Forwards: 70

From: <sip:214XXXXXXX@10.80.X.X;user=phone>;tag=1c455743820

To: <sip:0072@gdnblgttwnpzuvf4h1hj1.voiceconnector.chime.aws;user=phone>

Call-ID: 94987709413920238055@10.80.X.X

CSeq: 1 INVITE

Contact: <sip:10.80.X.X:5060>;+sip.src

Supported: replaces,resource-priority,sdp-anat

Allow:

REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE

Require: siprec

User-Agent: Mediant SW/v.7.40A.500.017

Content-Type: multipart/mixed;boundary=boundary_ac1dec

Content-Length: 2018

--boundary_ac1dec

Content-Type: application/sdp

v=0

o=AudiocodesGW 1984436309 1773405932 IN IP4 10.80.X.X

s=SBC-Call

c=IN IP4 10.80.X.X

t=0 0

m=audio 50276 RTP/AVP 0 101

c=IN IP4 10.80.X.X

a=ptime:20

```
a=sendonly
a=label:1
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,16
m=audio 50280 RTP/AVP 0 101
c=IN IP4 10.80.X.X
a=ptime:20
a=sendonly
a=label:2
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,16
```

```
--boundary_ac1dec
Content-Type: application/rs-metadata
Content-Disposition: recording-session
```

```
<?xml version="1.0" encoding="UTF-8"?>
<recording xmlns="urn:ietf:params:xml:ns:recording" xmlns:ac="http://AudioCodes">
  <datamode>complete</datamode>
  <group id="00000000-0000-00e2-f08e-1d00000738fa">
    <associate-time>2023-09-13T13:00:55Z</associate-time>
  </group>
  <session id="0000-0000-0000-0000-5d2ce63684edaeb9">
    <group-ref>00000000-0000-00e2-f08e-1d00000738fa</group-ref>
    <associate-time>2023-09-13T13:00:55Z</associate-time>
  </session>
  <participant id="214XXXXXXX" session="0000-0000-0000-0000-5d2ce63684edaeb9">
    <nameID aor="214XXXXXXX@10.80.X.X">
      <name xml:lang="en">Aravind Sankara</name>
    </nameID>
    <associate-time>2023-09-13T13:00:55Z</associate-time>
    <send>00000000-c460-00e2-f08e-1d00000738fa</send>
    <recv>00000001-19e6-00e2-f08e-1d00000738fa</recv>
  </participant>
```

```
<participant id="0072" session="0000-0000-0000-0000-5d2ce63684edaeb9">
  <nameID aor="0072@172.16.X.X"></nameID>
  <associate-time>2023-09-13T13:00:55Z</associate-time>
  <send>00000001-19e6-00e2-f08e-1d00000738fa</send>
  <recv>00000000-c460-00e2-f08e-1d00000738fa</recv>
</participant>
<stream id="00000000-c460-00e2-f08e-1d00000738fa" session="0000-0000-0000-0000-5d2ce63684edaeb9">
  <label>1</label>
</stream>
<stream id="00000001-19e6-00e2-f08e-1d00000738fa" session="0000-0000-0000-0000-5d2ce63684edaeb9">
  <label>2</label>
</stream>
</recording>
--boundary_ac1dec--
```

SIP/2.0 100 Trying

Via: SIP/2.0/UDP

10.80.11.64:5060;branch=z9hG4bKac757879020;rport=5060;received=199.182.124.60

From: <sip:214XXXXXXXX@10.80.X.X;user=phone>;tag=1c455743820

To: <sip:0072@gdnblgttwnpzuvf4h1hj1.voiceconnector.chime.aws;user=phone>

Call-ID: 94987709413920238055@10.80.X.X

CSeq: 1 INVITE

Content-Length: 0

SIP/2.0 200 OK

Via: SIP/2.0/UDP

10.80.11.64:5060;rport=5060;received=199.182.124.60;branch=z9hG4bKac757879020

Record-Route: <sip:3.80.16.10;lr;ftag=1c455743820;did=2b4.af8;nat=yes>

From: <sip:214XXXXXXXX@10.80.X.X;user=phone>;tag=1c455743820

To:

<sip:0072@gdnblgttwnpzuvf4h1hj1.voiceconnector.chime.aws;user=phone>;tag=Z2p021K4aZN5B

Call-ID: 94987709413920238055@10.80.X.X

CSeq: 1 INVITE

Contact: <sip:10.0.112.212:5060>

Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, INFO, REGISTER

Content-Type: application/sdp
Content-Length: 263
X-Vine-ID: bd89fc26-43eb-4b59-92c2-ad209b1964a5

v=0
o=- 1694610057396 1694610057396 IN IP4 3.80.17.228
s=session
c=IN IP4 3.80.17.228
t=0 0
m=audio 8628 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly
a=rtcp:8629
a=ptime:20
m=audio 8656 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly
a=rtcp:8657
a=ptime:20

ACK sip:10.0.112.212:5060 SIP/2.0
Via: SIP/2.0/UDP 10.80.X.X:5060;branch=z9hG4bKac999695496
Max-Forwards: 70
From: <sip:10.80.X.X;user=phone>;tag=1c455743820
To:
<sip:0072@gdnblgttnpzuvf4h1hj1.voiceconnector.chime.aws;user=phone>;tag=Z2p021K4aZN5B
Call-ID: 94987709413920238055@10.80.X.X
CSeq: 1 ACK
Contact: <sip:10.80.X.X:5060>
Route: <sip:3.80.16.10;lr;ftag=1c455743820;did=2b4.af8;nat=yes>
User-Agent: Mediant SW/v.7.40A.500.017
Content-Length: 0
BYE sip:10.0.112.212:5060 SIP/2.0
Via: SIP/2.0/UDP 10.80.X.X:5060;branch=z9hG4bKac539992007
Max-Forwards: 70

From: <sip:10.80.X.X;user=phone>;tag=1c455743820
To:
<sip:0072@gdnblgttnpzuvf4h1hj1.voiceconnector.chime.aws;user=phone>;tag=Z2p021K4aZN5B
Call-ID: 94987709413920238055@10.80.X.X
CSeq: 2 BYE
Route: <sip:3.80.16.10;lr;ftag=1c455743820;did=2b4.af8;nat=yes>
User-Agent: Mediant SW/v.7.40A.500.017
Content-Length: 0

SIP/2.0 200 OK
Via: SIP/2.0/UDP
10.80.11.64:5060;rport=5060;received=199.182.124.60;branch=z9hG4bKac539992007
From: <sip:10.80.X.X;user=phone>;tag=1c455743820
To:
<sip:0072@gdnblgttnpzuvf4h1hj1.voiceconnector.chime.aws;user=phone>;tag=Z2p021K4aZN5B
Call-ID: 94987709413920238055@10.80.X.X
CSeq: 2 BYE
Content-Length: 0

6 Test results

6.1 With UDP as Transport

Note: for the purposes of the test the SIPREC session was streamed to Kinesis Video Streams (KVS) and each call leg was recorded. A solution that results in only one recording that combines both call legs would be to use the Amazon Chime SDK Call Analytics service, which includes a call recording feature. For more information visit the [Call Analytics website](#).

Test Case ID	Title	Procedure	Expected Results	Status	Comments
1	Inbound call from PSTN	Inbound Call from PSTN to PBX User	<ol style="list-style-type: none">1) Call is connected2) RTP between PSTN and PBX User is captured3) Inbound caller number and PBX extension number are captured in the metadata (callerID capture to be tested)4) There is one call recording per call leg for the duration of the call, with accurate start and end timestamps5) Streaming and recording end when either PSTN or PBX user hangs up	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User Recording 2: PBX User to PSTN User
2	Outbound call to PSTN	Outbound call from PBX user to PSTN	<ol style="list-style-type: none">1) Call is connected2) RTP between PBX User and PSTN is captured	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User

			<p>3) PBX extension number and outbound caller number are captured in the metadata (callerID capture to be tested)</p> <p>4) There is one call recording per call leg for the duration of the call, with accurate start and end timestamps</p> <p>5) Streaming and recording end when either PBX or PSTN user hangs up</p>		Recording 2: PBX User to PSTN User
3	Inbound hold and resume	Inbound Call from PSTN to PBX User, PBX User places the call on hold and after some time period, resumes the call	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User is captured only when call is not on hold</p> <p>3) Inbound caller number and PBX extension number are captured in the metadata</p> <p>4) There is one call recording per call leg for the duration of the call</p> <p>5) The timestamps for the recording show accurate call duration for the entire call</p> <p>6) Streaming and recording end when either PSTN or PBX user hangs up</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User</p> <p>Recording 2: PBX User to PSTN User + Music on Hold</p> <p>There is no re-invite from PBX while the call is placed on HOLD. The recording is not paused, and the Music on Hold is captured.</p>
4	Outbound hold and resume	PBX User calls external PSTN number. After call	<p>1) Call is connected</p>	Passed	Two call recordings are available in AWS S3.

		is answered PBX User places the call on hold and after various time intervals resumes the call. Call ends when either PBX User or PSTN hangs up	<p>2) RTP between PBX User and PSTN is captured only when call is not on hold</p> <p>3) Outbound caller number and PBX extension number are captured in the metadata</p> <p>4) There is one call recording per call leg for the duration of the call</p> <p>5) The timestamps for the recording show accurate call duration for the entire call</p> <p>6) Streaming and recording end when either PSTN or PBX user hangs up</p>		<p>Recording 1: PSTN User to PBX User</p> <p>Recording 2: PBX User audio to PSTN User + Music on Hold</p> <p>There is no re-invite from PBX while the call is placed on HOLD. The recording is not paused and the music on hold is captured.</p>
5	Inbound call - attended call transfer	Inbound Call from PSTN to PBX User-1, PBX User-1 does an attended transfer to PBX User-2	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during transfer</p> <p>4) RTP between PSTN and PBX User-2 is captured after transfer</p> <p>5) Inbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>6) PBX User-2 extension number is added to the</p>	Passed	<p>Two call recordings are available in AWS S3</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX User-1's audio to PSTN + Music on Hold while PBX User-1 is on call with PBX User-2 + PBX User-2's audio to PSTN User</p> <p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with PBX User-2's extension.</p>

			<p>metadata after transfer completes</p> <p>7) There is one call recording per call leg for the duration of the call</p> <p>8) The timestamps for the recording show accurate call duration for the entire call</p> <p>9) Streaming and recording end when either PSTN or PBX User-2 hangs up</p>		
6	Outbound call - attended call transfer	Outbound call from PBX User-1 to PSTN. PBX User-1 does an attended transfer to PBX User-2	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during transfer</p> <p>4) RTP between PSTN and PBX User-2 is captured after transfer</p> <p>5) Outbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>6) PBX User-2 extension number is added to the metadata after transfer completes</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX user 1's audio to PSTN + Music on Hold while PBX User-1 is on call with PBX User-2 + PBX User-2's audio to PSTN User</p> <p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with PBX User-2's extension.</p>

			<p>7) There is one call recording per call leg for the duration of the call</p> <p>8) The timestamps for the recording show accurate call duration for the entire call</p> <p>9) Streaming and recording end when either PSTN or PBX User-2 hangs up</p>		
7	Inbound call - external transfer	Inbound call from PSTN User-1 to PBX User-1, PBX User-1 does an attended transfer to PSTN User-2	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN User-1 and PBX User-1 during transfer</p> <p>4) RTP between PSTN User-1 and PSTN User-2 is captured after transfer</p> <p>5) Inbound caller number and PBX User-1 extension number are captured in metadata</p> <p>6) PSTN User-2 caller number is added to the metadata after transfer completes</p> <p>7) There is one call recording per call leg for the duration of the call</p>	Passed	<p>Four call recordings are available in AWS S3.</p> <p>Recording 1: PBX User-1's audio to PSTN User-1 + Music on Hold while PBX User-1 is on call with PSTN User-2 + PSTN User-2's audio with PSTN User-1</p> <p>Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2</p> <p>Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1</p> <p>Recording 4: PBX User-1's audio to PSTN user 2</p> <p>There is no re-invite from PBX during transfer. Hence, the recording is not paused and the music on hold is captured.</p>

			<p>8) The timestamps for the recording show accurate call duration for the entire call</p> <p>9) Streaming and recording end when either PSTN User-1 or PSTN User-2 hangs up</p>		
8	Inbound call - blind call transfer	Inbound Call from PSTN to PBX User-1, PBX User-1 does a blind transfer to PBX User-2	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during transfer</p> <p>4) RTP between PSTN and PBX User-2 is captured after transfer</p> <p>5) Inbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>6) PBX User-2 extension number is added to the metadata after transfer completes</p> <p>7) There is one call recording per call leg for the duration of the call</p> <p>8) The timestamps for the recording show accurate call duration for the entire call</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX User-1's audio to PSTN + Music on Hold while PBX User-1 attempts to transfer + PBX User-2's audio to PSTN User</p> <p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with PBX User-2's extension.</p>

			9) Streaming and recording end when either PSTN or PBX User-2 hangs up		
9	Outbound call - blind call transfer	Outbound call from PBX User-1 to PSTN. PBX User-1 does a blind transfer to PBX User-2	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during transfer</p> <p>4) RTP between PSTN and PBX User-2 is captured after transfer</p> <p>5) Outbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>6) PBX User-2 extension number is added to the metadata after transfer completes</p> <p>7) There is one call recording per call leg for the duration of the call</p> <p>8) The timestamps for the recording show accurate call duration for the entire call</p> <p>9) Streaming and recording end when either PSTN or PBX User-2 hangs up</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX User-1's audio to PSTN + Music on Hold while PBX User-1 attempts to transfer + PBX User-2's audio to PSTN User</p> <p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with PBX User-2's extension.</p>

10	Inbound call - internal conference	Inbound call from PSTN to PBX User-1. PBX User-1 places PSTN on hold and consults with PBX User-2. PBX User-2 is conferenced into the call. The call terminates when one of the last two call participants hangs up	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PSTN and PBX User-1 is captured 3) RTP is not captured between PSTN and PBX User-1 during setup of call with PBX User-2 4) RTP between PSTN, PBX User-1, and PBX User-2 is captured after PBX User-2 is added to the call as an active participant 5) Inbound caller number and PBX User-1 extension number are captured in the metadata 6) PBX User-2 extension number is added to the metadata after conference starts 7) There is one call recording per call leg for the duration of the call 8) The timestamps for the recording show accurate call duration for the entire call 9) Streaming and recording end when either PSTN hangs up or last participant from PBX User-1 and User-2 hangs up 	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX User-1's audio to PSTN + Music on Hold from PBX User-1 while consulting PBX User-2 for conference + PBX User-1's audio to PSTN User and PBX User-2 + PBX User-2's audio to PSTN User and PBX User-1</p> <p>There is no mid-call signaling from PBX for call escalation to conference. Therefore, music on hold is recorded while escalating the call to conference and meta data is not updated with PBX User-2's extension.</p>
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11	Outbound call - internal conference	Outbound call from PBX User-1 to PSTN. PBX User-1 places PSTN on hold and consults with PBX User-2. PBX User-2 is conferenced into the call. The call terminates when one of the last two call participants hangs up	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PBX User-1 and PSTN is captured 3) RTP is not captured between PSTN and PBX User-1 during setup of call with PBX User-2 4) RTP between PSTN, PBX User-1, and PBX User-2 is captured after PBX User-2 is added to the call as an active participant 5) Outbound caller number and PBX User-1 extension number are captured in the metadata 6) PBX User-2 extension number is added to the metadata after conference starts 7) There is one call recording per call leg for the duration of the call 8) The timestamps for the recording show accurate call duration for the entire call 9) Streaming and recording end when either PSTN hangs up or last participant from PBX User-1 and User-2 hangs up 	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX User-1's audio to PSTN + Music on Hold from PBX User-1 while consulting PBX User-2 for conference + PBX User-1's audio to PSTN User and PBX User-2 + PBX User-2's audio to PSTN User and PBX User-1</p> <p>There is no mid-call signaling from PBX for call escalation to conference. Therefore, music on hold is recorded while escalating the call to conference and meta data is not updated with PBX User-2's extension.</p>
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12	Inbound call with external conference	Inbound call from PSTN User-1 to PBX User-1. PBX User-1 places PSTN User-1 on hold and calls with PSTN User-2. PSTN User-2 is conferenced into the call. The call ends when one of the last two call participants hangs up	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PSTN and PBX User-1 is captured 3) RTP is not captured between PSTN and PBX User-1 during setup of call with PSTN User-2 4) RTP between PBX User-1 and PSTN User-2 is captured 5) RTP between PSTN User-1, PBX User-1, and PSTN User-2 is captured after PSTN User-2 is added to the call as an active participant 6) Inbound caller number and PBX User-1 extension number are captured in the metadata 7) PSTN User-2 caller number is added to the metadata after the conference starts 8) There is one call recording per call leg for the duration of the call 9) The timestamps for the recording show accurate call duration for the entire call 10) Streaming and recording end when one of the last two call participants hangs up 	Passed	<p>Four call recordings are available in AWS S3.</p> <p>Recording 1: PBX User-1's audio to PSTN User-1 + Music on Hold from PBX User-1 while consulting PSTN User-2 for conference + PSTN User-2's audio and PBX User-1's audio to PSTN User-1</p> <p>Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2</p> <p>Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1</p> <p>Recording 4: PBX User-1's audio to PSTN User-2 + PSTN User-1's audio and PBX User-1's audio to PSTN User-2</p> <p>There is no mid-call signaling from PBX for call escalation to conference. Therefore, music on hold is recorded while escalating the call to conference.</p>
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13	Outbound call with external conference	Outbound call from PBX User-1 to PSTN User-1. PBX User-1 places PSTN User-1 on hold and calls PSTN User-2. PSTN User-2 is conferenced into the call. The call ends when one of the last two call participants hangs up	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PBX User-1 and PSTN User-1 is captured 3) RTP is not captured between PSTN User-1 and PBX User-1 during setup of call with PSTN User-2 4) RTP between PBX User-1 and PSTN User-2 is captured. 5) RTP between PSTN User-1, PBX User-1, and PSTN User-2 is captured after PSTN User-2 is added to the call as an active participant 6) Outbound caller number and PBX User-1 extension number are captured in the metadata 7) PSTN User-2 caller number is added to the metadata after conference starts 8) There is one call recording per call leg for the duration of the call 9) The timestamps for the recording show accurate call duration for the entire call 10) Streaming and recording end when one of the last two call participants hangs up 	Passed	<p>Four call recordings are available in AWS S3.</p> <p>Recording 1: PBX User-1's audio to PSTN User-1 + Music on Hold from PBX User-1 while consulting PSTN User-2 for conference + PSTN User-2's audio and PBX User-1's audio to PSTN User-1</p> <p>Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2</p> <p>Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1</p> <p>Recording 4: PBX User-1's audio to PSTN User-2 + PSTN User-1's audio and PBX User-1's audio to PSTN User-2</p> <p>There is no mid-call signaling from PBX for call escalation to conference. Therefore, music on hold is recorded while escalating the call to conference.</p>
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14	Inbound call - transfer to queue	Inbound call from PSTN to PBX User-1. PBX User-1 transfers the call to call queue. PSTN drops the call	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PSTN and PBX User-1 is captured 3) RTP is not captured between PSTN and PBX User-1 during transfer 4) RTP is captured when queue accepts call 5) Inbound caller number and PBX User-1 extension number are captured in the metadata 6) Queue number is captured in the metadata after call transfer 7) There is one call recording per call leg for the duration of the call 8) The timestamps for the recording show accurate start and end times 9) Streaming and recording end when PSTN hangs up 	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1</p> <p>Recording 2: PBX User-1's audio to PSTN User + Music on Hold while PBX User-1 attempts to transfer until extensions associated to call queue are ringing</p> <p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with call queue number.</p>
15	Inbound call - transfer to queue then to agent	Inbound call from PSTN to PBX User-1. PBX User-1 transfers the call to call queue. PBX User-2 picks up the call from the queue	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PSTN and PBX User-1 is captured 3) RTP is not captured between PSTN and PBX User-1 during transfer 	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p>

			<p>4) RTP is captured when queue accepts call</p> <p>5) RTP between PSTN and PBX User-2 is captured</p> <p>6) Inbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>7) Queue number is captured in the metadata after call transfer</p> <p>8) There is one call recording per call leg for the duration of the call</p> <p>9) The timestamps for the recording show accurate start and end times</p> <p>10) Streaming and recording end when PSTN hangs up</p>		<p>Recording 2: PBX User-1's audio to PSTN User + Music on Hold while PBX User-1 attempts to transfer + PBX User-2's audio to PSTN User</p> <p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with call queue number and PBX User-2's extension.</p>
16	Inbound call with consult	Inbound call from PSTN to PBX User-1. PBX User-1 places PSTN on hold and calls PBX User-2, who answers. PBX User-2 hangs up and PBX User-1 resumes call with PSTN	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured only when call is not on hold</p> <p>3) RTP between PBX User-1 and PBX User-2 is not captured</p> <p>4) Inbound caller number and PBX extension number are captured in the metadata</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1</p> <p>Recording 2: PBX User-1's audio to PSTN User + Music On Hold while PBX User-1 is on call with PBX User-2 + resumed PBX User-1's audio to PSTN User</p> <p>There is no re-invite from PBX while the call is placed on HOLD. The recording is not paused and the music on hold is captured.</p>

			<p>5) Metadata is captured when PBX User-2 is added and when they are dropped from the call</p> <p>6) There is one call recording per call leg for the duration of the call</p> <p>7) The timestamps for the recording show accurate call duration for the entire call</p> <p>8) Streaming and recording end when either PSTN or PBX user hangs up</p>		
17	Inbound call with extended consult	Inbound call from PSTN to PBX User-1. PBX User-1 places PSTN on hold and calls PBX User-2, who answers. PBX User-2 is put on hold and PBX User-1 resumes call with PSTN. This sequence may be repeated multiple times until either PSTN or PBX User-1 hangs up	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured only when call is not on hold</p> <p>3) RTP between PBX User-1 and PBX User-2 is not captured</p> <p>4) Inbound caller number and PBX extension number are captured in the metadata</p> <p>5) Metadata is captured when PBX User-2 is added and when they are dropped from the call</p> <p>6) There is one call recording per call leg for the duration of the call</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1</p> <p>Recording 2: PBX User-1's audio to PSTN User + Music On Hold while PBX User-1 is on call with PBX User-2 + resumed PBX User-1's audio to PSTN User</p> <p>There is no re-invite from PBX while the call is placed on HOLD. The recording is not paused and the music on hold is captured.</p>

			<p>7) The timestamps for the recording show accurate call duration for the entire call</p> <p>8) Streaming and recording end when either PSTN or PBX user hangs up</p>		
18	Inbound call with multi-party conference	Inbound call from PSTN to PBX User-1. PBX User-1 places PSTN on hold and consults with PBX User-2. PBX User-2 is conferenced into the call. PBX User-1 then adds PBX User-3 to the call. Call ends when either PSTN or last PBX User in the call hangs up	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during setup of call with PBX User-2</p> <p>4) RTP between PSTN, PBX User-1, and PBX User-2 is captured after PBX User-2 is added to the call as an active participant</p> <p>5) RTP between PSTN, PBX User-1, PBX User-2, and PBX User-3 is captured after PBX User-3 is added to the call</p> <p>6) Inbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>7) PBX User-2 extension number is added to the metadata after conference starts</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1, PBX User-2, and PBX User-3</p> <p>Recording 2: PBX User-1's audio with PSTN User + Music On Hold from PBX User-1 while consulting PBX User-2 for conference + PBX User-1's audio to PSTN User and PBX User-2 + PBX User-2's audio to PSTN User and PBX User-1 + Music On Hold from PBX User-1 while consulting PBX User-3 for conference + PBX User-1's audio to PBX User-2, PBX User-3 and PSTN User + PBX User-2's audio to PBX User-1, PBX User-3 and PSTN User + PBX User-3's audio to PBX User-1, PBX User-2 and PSTN User</p> <p>There is no mid-call signaling from PBX for call escalation to conference. Therefore, music on hold is recorded while escalating the call to conference and meta data is not updated with PBX User-2's and PBX User-3's extensions.</p>

			<p>8) PBX User-3 extension number is added to the metadata after addition to conference</p> <p>9) There is one call recording per call leg for the duration of the call</p> <p>10) The timestamps for the recording show accurate call duration for the entire call</p> <p>11) Streaming and recording end when either PSTN hangs up or last participant from PBX User-1 and User-2 hangs up</p>		
19	Outbound conference call	<p>PBX User-1 calls PBX User-2. PBX User-2 calls customer on PSTN number. Call ends when either of the last two call participants hangs up</p>	<p>1) Call is connected when customer answers call from PBX User-2</p> <p>2) RTP between PBX User-2 and customer on PSTN is captured.</p> <p>3) RTP between PBX User-1, PBX User-2 and customer is captured</p> <p>4) PBX User-1, PBX User-2, and customer called number are captured in the metadata</p> <p>5) There is one call recording per call leg for the duration of the call</p> <p>6) Call ends when customer or</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX User-2's audio to PBX User-1 and PSTN User + PBX User-1's audio to PBX User-2 and PSTN User</p> <p>Meta data information only has PBX User-2's extension and PSTN User number.</p>

			<p>last remaining PBX user hangs up</p> <p>7) The timestamps for the recording show accurate start and end times</p> <p>8) Streaming and recording end when condition 6 is met</p>		
20	Emergency calling	PBX User-1 calls the 511 service	<p>1) Call is connected</p> <p>2) RTP between PBX User and 511 is captured</p> <p>3) PBX extension number and outbound caller number (511) are captured in the metadata (caller ID capture to be tested)</p> <p>4) There is one call recording per call leg for the duration of the call, with accurate start and end timestamps</p> <p>5) Streaming and recording end when either PBX or 511 user hangs up</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: 511 User to PBX User</p> <p>Recording 2: PBX User to 511 User</p> <p>Note: - This scenario is locally simulated within Lab environment.</p>
21	Outbound international call	Outbound call from PBX User-1 to international PSTN number	<p>1) Call is connected</p> <p>2) RTP between PBX Users and PSTN is captured</p> <p>3) PBX extension number and outbound caller number are captured in the metadata (caller ID capture to be tested)</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: International PSTN User to PBX User</p> <p>Recording 2: PBX User to International PSTN User</p>

			4) There is one call recording per call leg for the duration of the call, with accurate start and end timestamps 5) Streaming and recording end when either PBX or PSTN user hangs up		
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6.2 With TLS as Transport

Test Case ID	Title	Procedure	Expected Results	Status	Comments
1	Inbound call from PSTN	Inbound Call from PSTN to PBX User	1) Call is connected 2) RTP between PSTN and PBX User is captured 3) Inbound caller number and PBX extension number are captured in the metadata (callerID capture to be tested) 4) There is one call recording per call leg for the duration of the call, with accurate start and end timestamps 5) Streaming and recording end when either PSTN or PBX user hangs up	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User Recording 2: PBX User to PSTN User
2	Outbound call to PSTN	Outbound call from PBX user to PSTN	1) Call is connected 2) RTP between PBX User and PSTN is captured 3) PBX extension number and outbound caller number are captured in the metadata (callerID capture to be tested) 4) There is one call recording per call leg for the duration of the call, with accurate start and end timestamps	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User Recording 2: PBX User to PSTN User

			5) Streaming and recording end when either PBX or PSTN user hangs up		
3	Inbound hold and resume	Inbound Call from PSTN to PBX User, PBX User places the call on hold and after some time period, resumes the call	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PSTN and PBX User is captured only when call is not on hold 3) Inbound caller number and PBX extension number are captured in the metadata 4) There is one call recording per call leg for the duration of the call 5) The timestamps for the recording show accurate call duration for the entire call 6) Streaming and recording end when either PSTN or PBX user hangs up 	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User</p> <p>Recording 2: PBX User to PSTN User + Music on Hold</p> <p>There is no re-invite from PBX while the call is placed on HOLD. The recording is not paused, and the Music on Hold is captured.</p>
4	Outbound hold and resume	PBX User calls external PSTN number. After call is answered PBX User places the call on hold and after various time intervals resumes the call. Call ends when either PBX	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PBX User and PSTN is captured only when call is not on hold 3) Outbound caller number and PBX extension number are captured in the metadata 4) There is one call recording per call leg for the duration of the call 	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1</p> <p>Recording 2: PBX User-1's audio to PSTN User + Music on Hold</p>

		User or PSTN hangs up	<p>5) The timestamps for the recording show accurate call duration for the entire call</p> <p>6) Streaming and recording end when either PSTN or PBX user hangs up</p>		There is no re-invite from PBX while the call is placed on HOLD. The recording is not paused and the music on hold is captured.
5	Inbound call - attended call transfer	Inbound Call from PSTN to PBX User-1, PBX User-1 does an attended transfer to PBX User-2	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during transfer</p> <p>4) RTP between PSTN and PBX User-2 is captured after transfer</p> <p>5) Inbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>6) PBX User-2 extension number is added to the metadata after transfer completes</p> <p>7) There is one call recording per call leg for the duration of the call</p> <p>8) The timestamps for the recording show accurate call duration for the entire call</p>	Passed	<p>Two call recordings are available in AWS S3</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX User-1's audio to PSTN User + Music on Hold while PBX User-1 is on call with PBX User-2 + PBX User-2's audio to PSTN User</p> <p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with PBX User-2's extension.</p>

			9) Streaming and recording end when either PSTN or PBX User-2 hangs up		
6	Outbound call - attended call transfer	Outbound call from PBX User-1 to PSTN. PBX User-1 does an attended transfer to PBX User-2	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during transfer</p> <p>4) RTP between PSTN and PBX User-2 is captured after transfer</p> <p>5) Outbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>6) PBX User-2 extension number is added to the metadata after transfer completes</p> <p>7) There is one call recording per call leg for the duration of the call</p> <p>8) The timestamps for the recording show accurate call duration for the entire call</p> <p>9) Streaming and recording end when either PSTN or PBX User-2 hangs up</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX user 1's audio to PSTN + Music on Hold while PBX User-1 is on call with PBX User-2 + PBX User-2's audio to PSTN User</p> <p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with PBX User-2's extension.</p>

7	Inbound call - external transfer	Inbound call from PSTN User-1 to PBX User-1, PBX User-1 does an attended transfer to PSTN User-2	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PSTN and PBX User-1 is captured 3) RTP is not captured between PSTN User-1 and PBX User-1 during transfer 4) RTP between PSTN User-1 and PSTN User-2 is captured after transfer 5) Inbound caller number and PBX User-1 extension number are captured in metadata 6) PSTN User-2 caller number is added to the metadata after transfer completes 7) There is one call recording per call leg for the duration of the call 8) The timestamps for the recording show accurate call duration for the entire call 9) Streaming and recording end when either PSTN User-1 or PSTN User-2 hangs up 	Passed	<p>Four call recordings are available in AWS S3.</p> <p>Recording 1: PBX User-1's audio to PSTN User-1 + Music on Hold while PBX User-1 is on call with PSTN User-2 + PSTN User-2's audio with PSTN User-1</p> <p>Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2</p> <p>Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1</p> <p>Recording 4: PBX User-1's audio to PSTN user 2</p> <p>There is no re-invite from PBX during transfer. Hence, the recording is not paused and the music on hold is captured.</p>
8	Inbound call - blind call transfer	Inbound Call from PSTN to PBX User-1, PBX User-1 does a blind transfer to PBX User-2	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PSTN and PBX User-1 is captured 	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p>

			<p>3) RTP is not captured between PSTN and PBX User-1 during transfer</p> <p>4) RTP between PSTN and PBX User-2 is captured after transfer</p> <p>5) Inbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>6) PBX User-2 extension number is added to the metadata after transfer completes</p> <p>7) There is one call recording per call leg for the duration of the call</p> <p>8) The timestamps for the recording show accurate call duration for the entire call</p> <p>9) Streaming and recording end when either PSTN or PBX User-2 hangs up</p>		<p>Recording 2: PBX User-1's audio to PSTN + Music on Hold while PBX User-1 attempts to transfer + PBX User-2's audio to PSTN User</p> <p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with PBX User-2's extension.</p>
9	Outbound call - blind call transfer	Outbound call from PBX User-1 to PSTN. PBX User-1 does a blind transfer to PBX User-2	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during transfer</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p>

			<p>4) RTP between PSTN and PBX User-2 is captured after transfer</p> <p>5) Outbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>6) PBX User-2 extension number is added to the metadata after transfer completes</p> <p>7) There is one call recording per call leg for the duration of the call</p> <p>8) The timestamps for the recording show accurate call duration for the entire call</p> <p>9) Streaming and recording end when either PSTN or PBX User-2 hangs up</p>		<p>Recording 2: PBX User-1's audio to PSTN + Music on Hold while PBX User-1 attempts to transfer + PBX User-2's audio to PSTN User</p> <p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with PBX User-2's extension.</p>
10	Inbound call - internal conference	Inbound call from PSTN to PBX User-1. PBX User-1 places PSTN on hold and consults with PBX User-2. PBX User-2 is conferenced into the call. The call terminates when one of the last two	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during setup of call with PBX User-2</p> <p>4) RTP between PSTN, PBX User-1, and PBX User-2 is captured after PBX User-2 is</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX User-1's audio to PSTN + Music on Hold from PBX User-1 while consulting PBX User-2 for conference + PBX User-1's audio to PSTN</p>

		call participants hangs up	<p>added to the call as an active participant</p> <p>5) Inbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>6) PBX User-2 extension number is added to the metadata after conference starts</p> <p>7) There is one call recording per call leg for the duration of the call</p> <p>8) The timestamps for the recording show accurate call duration for the entire call</p> <p>9) Streaming and recording end when either PSTN hangs up or last participant from PBX User-1 and User-2 hangs up</p>		<p>User and PBX User-2 + PBX User-2's audio to PSTN User and PBX User-1</p> <p>There is no mid-call signaling from PBX for call escalation to conference. Therefore, music on hold is recorded while escalating the call to conference and meta data is not updated with PBX User-2's extension.</p>
11	Outbound call - internal conference	Outbound call from PBX User-1 to PSTN. PBX User-1 places PSTN on hold and consults with PBX User-2. PBX User-2 is conferenced into the call. The call terminates when one of the last two	<p>1) Call is connected</p> <p>2) RTP between PBX User-1 and PSTN is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during setup of call with PBX User-2</p> <p>4) RTP between PSTN, PBX User-1, and PBX User-2 is captured after PBX User-2 is</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX User-1's audio to PSTN + Music on Hold from PBX User-1 while consulting PBX User-2 for conference + PBX User-1's audio to PSTN User and PBX User-2 + PBX User-2's audio to PSTN User and PBX User-1</p>

		call participants hangs up	<p>added to the call as an active participant</p> <p>5) Outbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>6) PBX User-2 extension number is added to the metadata after conference starts</p> <p>7) There is one call recording per call leg for the duration of the call</p> <p>8) The timestamps for the recording show accurate call duration for the entire call</p> <p>9) Streaming and recording end when either PSTN hangs up or last participant from PBX User-1 and User-2 hangs up</p>		<p>There is no mid-call signaling from PBX for call escalation to conference. Therefore, music on hold is recorded while escalating the call to conference and meta data is not updated with PBX User-2's extension.</p>
12	Inbound call with external conference	Inbound call from PSTN User-1 to PBX User-1. PBX User-1 places PSTN User-1 on hold and calls with PSTN User-2. PSTN User-2 is conferenced into the call. The call ends when one of the last two call	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during setup of call with PSTN User-2</p> <p>4) RTP between PBX User-1 and PSTN User-2 is captured</p>	Passed	<p>Four call recordings are available in AWS S3.</p> <p>Recording 1: PBX User-1's audio to PSTN User-1 + Music on Hold from PBX User-1 while consulting PSTN User-2 for conference + PSTN User-2's audio and PBX User-1's audio to PSTN User-1</p> <p>Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2</p>

		participants hangs up	<p>5) RTP between PSTN User-1, PBX User-1, and PSTN User-2 is captured after PSTN User-2 is added to the call as an active participant</p> <p>6) Inbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>7) PSTN User-2 caller number is added to the metadata after the conference starts</p> <p>8) There is one call recording per call leg for the duration of the call</p> <p>9) The timestamps for the recording show accurate call duration for the entire call</p> <p>10) Streaming and recording end when one of the last two call participants hangs up</p>		<p>Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1</p> <p>Recording 4: PBX User-1's audio to PSTN User-2 + PSTN User-1's audio and PBX User-1's audio to PSTN User-2</p> <p>There is no mid-call signaling from PBX for call escalation to conference. Therefore, music on hold is recorded while escalating the call to conference.</p>
13	Outbound call with external conference	Outbound call from PBX User-1 to PSTN User-1. PBX User-1 places PSTN User-1 on hold and calls PSTN User-2. PSTN User-2 is conferenced into the call. The call ends when one of the last two call	<p>1) Call is connected</p> <p>2) RTP between PBX User-1 and PSTN User-1 is captured</p> <p>3) RTP is not captured between PSTN User-1 and PBX User-1 during setup of call with PSTN User-2</p> <p>4) RTP between PBX User-1 and PSTN User-2 is captured.</p>	Passed	<p>Four call recordings are available in AWS S3.</p> <p>Recording 1: PBX User-1's audio to PSTN User-1 + Music on Hold from PBX User-1 while consulting PSTN User-2 for conference + PSTN User-2's audio and PBX User-1's audio to PSTN User-1</p> <p>Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2</p>

		participants hangs up	<p>5) RTP between PSTN User-1, PBX User-1, and PSTN User-2 is captured after PSTN User-2 is added to the call as an active participant</p> <p>6) Outbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>7) PSTN User-2 caller number is added to the metadata after conference starts</p> <p>8) There is one call recording per call leg for the duration of the call</p> <p>9) The timestamps for the recording show accurate call duration for the entire call</p> <p>10) Streaming and recording end when one of the last two call participants hangs up</p>		<p>Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1</p> <p>Recording 4: PBX User-1's audio to PSTN User-2 + PSTN User-1's audio and PBX User-1's audio to PSTN User-2</p> <p>There is no mid-call signaling from PBX for call escalation to conference. Therefore, music on hold is recorded while escalating the call to conference.</p>
14	Inbound call - transfer to queue	Inbound call from PSTN to PBX User-1. PBX User-1 transfers the call to call queue. PSTN drops the call	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during transfer</p> <p>4) RTP is captured when queue accepts call</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1</p> <p>Recording 2: PBX User-1's audio to PSTN User + Music on Hold while PBX User-1 attempts to transfer until extensions associated to call queue are ringing</p>

			<p>5) Inbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>6) Queue number is captured in the metadata after call transfer</p> <p>7) One recording for the entire call duration is stored</p> <p>8) The timestamps for the recording show accurate start and end times</p> <p>9) Streaming and recording end when PSTN hangs up</p>		<p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with call queue number.</p>
15	Inbound call - transfer to queue then to agent	Inbound call from PSTN to PBX User-1. PBX User-1 transfers the call to call queue. PBX User-2 picks up the call from the queue	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured</p> <p>3) RTP is not captured between PSTN and PBX User-1 during transfer</p> <p>4) RTP is captured when queue accepts call</p> <p>5) RTP between PSTN and PBX User-2 is captured</p> <p>6) Inbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>7) Queue number is captured in the metadata after call transfer</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX User-1's audio to PSTN User + Music on Hold while PBX User-1 attempts to transfer + PBX User-2's audio to PSTN User</p> <p>There is no re-invite from PBX during transfer. Hence, Music on Hold is captured, and meta data is not updated with call queue number and PBX User-2's extension.</p>

			<p>8) There is one call recording per call leg for the duration of the call</p> <p>9) The timestamps for the recording show accurate start and end times</p> <p>10) Streaming and recording end when PSTN hangs up</p>		
16	Inbound call with consult	Inbound call from PSTN to PBX User-1. PBX User-1 places PSTN on hold and calls PBX User-2, who answers. PBX User-2 hangs up and PBX User-1 resumes call with PSTN	<p>1) Call is connected</p> <p>2) RTP between PSTN and PBX User-1 is captured only when call is not on hold</p> <p>3) RTP between PBX User-1 and PBX User-2 is not captured</p> <p>4) Inbound caller number and PBX extension number are captured in the metadata</p> <p>5) Metadata is captured when PBX User-2 is added and when they are dropped from the call</p> <p>6) One recording for the entire call duration is stored</p> <p>7) The timestamps for the recording show accurate call duration for the entire call</p> <p>8) Streaming and recording end when either PSTN or PBX user hangs up</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1</p> <p>Recording 2: PBX User-1's audio to PSTN User + Music On Hold while PBX User-1 is on call with PBX User-2 + resumed PBX User-1's audio to PSTN User</p> <p>There is no re-invite from PBX while the call is placed on HOLD. The recording is not paused and the music on hold is captured.</p>

17	Inbound call with extended consult	Inbound call from PSTN to PBX User-1. PBX User-1 places PSTN on hold and calls PBX User-2, who answers. PBX User-2 is put on hold and PBX User-1 resumes call with PSTN. This sequence may be repeated multiple times until either PSTN or PBX User-1 hangs up	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PSTN and PBX User-1 is captured only when call is not on hold 3) RTP between PBX User-1 and PBX User-2 is not captured 4) Inbound caller number and PBX extension number are captured in the metadata 5) Metadata is captured when PBX User-2 is added and when they are dropped from the call 6) There is one call recording per call leg for the duration of the call 7) The timestamps for the recording show accurate call duration for the entire call 8) Streaming and recording end when either PSTN or PBX user hangs up 	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1</p> <p>Recording 2: PBX User-1's audio to PSTN User + Music On Hold while PBX User-1 is on call with PBX User-2 + resumed PBX User-1's audio to PSTN User</p> <p>There is no re-invite from PBX while the call is placed on HOLD. The recording is not paused and the music on hold is captured.</p>
18	Inbound call with multi-party conference	Inbound call from PSTN to PBX User-1. PBX User-1 places PSTN on hold and consults with PBX User-2. PBX User-2 is conferenced into	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PSTN and PBX User-1 is captured 3) RTP is not captured between PSTN and PBX User-1 during setup of call with PBX User-2 	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1, PBX User-2, and PBX User-3</p> <p>Recording 2: PBX User-1's audio with PSTN User + Music On Hold from PBX User-1 while consulting</p>

		<p>the call. PBX User-1 then adds PBX User-3 to the call. Call ends when either PSTN or last PBX User in the call hangs up</p>	<p>4) RTP between PSTN, PBX User-1, and PBX User-2 is captured after PBX User-2 is added to the call as an active participant</p> <p>5) RTP between PSTN, PBX User-1, PBX User-2, and PBX User-3 is captured after PBX User-3 is added to the call</p> <p>6) Inbound caller number and PBX User-1 extension number are captured in the metadata</p> <p>7) PBX User-2 extension number is added to the metadata after conference starts</p> <p>8) PBX User-3 extension number is added to the metadata after addition to conference</p> <p>9) There is one call recording per call leg for the duration of the call</p> <p>10) The timestamps for the recording show accurate call duration for the entire call</p> <p>11) Streaming and recording end when either PSTN hangs up or last participant from PBX User-1 and User-2 hangs up</p>		<p>PBX User-2 for conference + PBX User-1's audio to PSTN User and PBX User-2 + PBX User-2's audio to PSTN User and PBX User-1 + Music On Hold from PBX User-1 while consulting PBX User-3 for conference + PBX User-1's audio to PBX User-2, PBX User-3 and PSTN User + PBX User-2's audio to PBX User-1, PBX User-3 and PSTN User + PBX User-3's audio to PBX User-1, PBX User-2 and PSTN User</p> <p>There is no mid-call signaling from PBX for call escalation to conference. Therefore, music on hold is recorded while escalating the call to conference and meta data is not updated with PBX User-2's and PBX User-3's extensions.</p>
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19	Outbound conference call	PBX User-1 calls PBX User-2. PBX User-2 calls customer on PSTN number. Call ends when either of the last two call participants hangs up	<ol style="list-style-type: none"> 1) Call is connected when customer answers call from PBX User-2 2) RTP between PBX User-2 and customer on PSTN is captured. 3) RTP between PBX User-1, PBX User-2 and customer is captured 4) PBX User-1, PBX User-2, and customer called number are captured in the metadata 5) There is one call recording per call leg for the duration of the call 6) Call ends when customer or last remaining PBX user hangs up 7) The timestamps for the recording show accurate start and end times 8) Streaming and recording end when condition 6 is met 	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: PSTN User to PBX User-1 and PBX User-2</p> <p>Recording 2: PBX User-2's audio to PBX User-1 and PSTN User + PBX User-1's audio to PBX User-2 and PSTN User</p> <p>Meta data information only has PBX User-2's extension and PSTN User number.</p>
20	Emergency calling	PBX User-1 calls the 511 service	<ol style="list-style-type: none"> 1) Call is connected 2) RTP between PBX User and 511 is captured 3) PBX extension number and outbound caller number (511) are captured in the metadata (caller ID capture to be tested) 	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: 511 User to PBX User</p> <p>Recording 2: PBX User to 511 User</p>

			<p>4) There is one call recording per call leg for the duration of the call, with accurate start and end timestamps</p> <p>5) Streaming and recording end when either PBX or 511 user hangs up</p>		Note: - This scenario is locally simulated within Lab environment.
21	Outbound international call	Outbound call from PBX User-1 to international PSTN number	<p>1) Call is connected</p> <p>2) RTP between PBX Users and PSTN is captured</p> <p>3) PBX extension number and outbound caller number are captured in the metadata (caller ID capture to be tested)</p> <p>4) There is one call recording for the duration of the call, with accurate start and end timestamps</p> <p>5) Streaming and recording end when either PBX or PSTN user hangs up</p>	Passed	<p>Two call recordings are available in AWS S3.</p> <p>Recording 1: International PSTN User to PBX User</p> <p>Recording 2: PBX User to International PSTN User</p>