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.

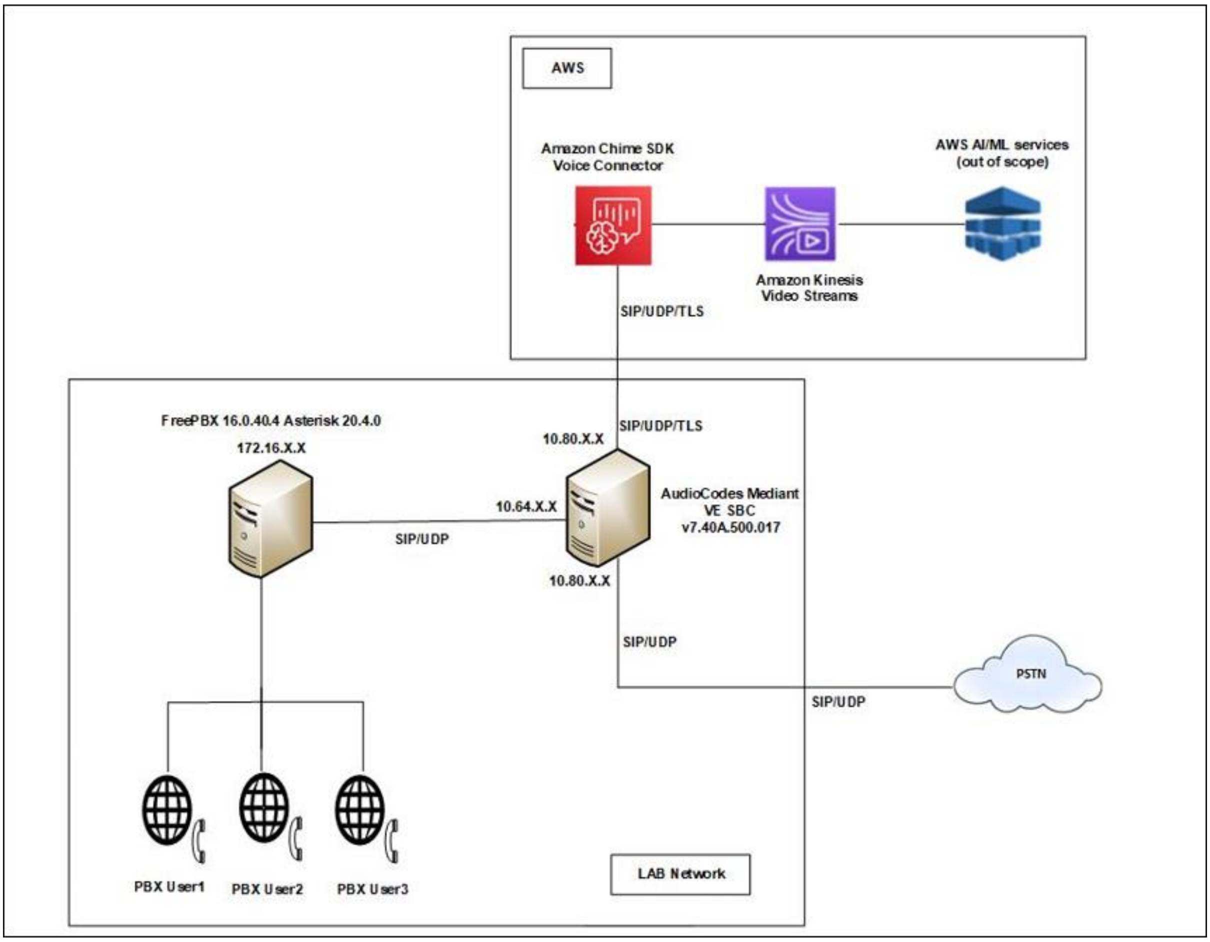

Figure 1: Network Topology

The signaling and media flow is illustrated below:

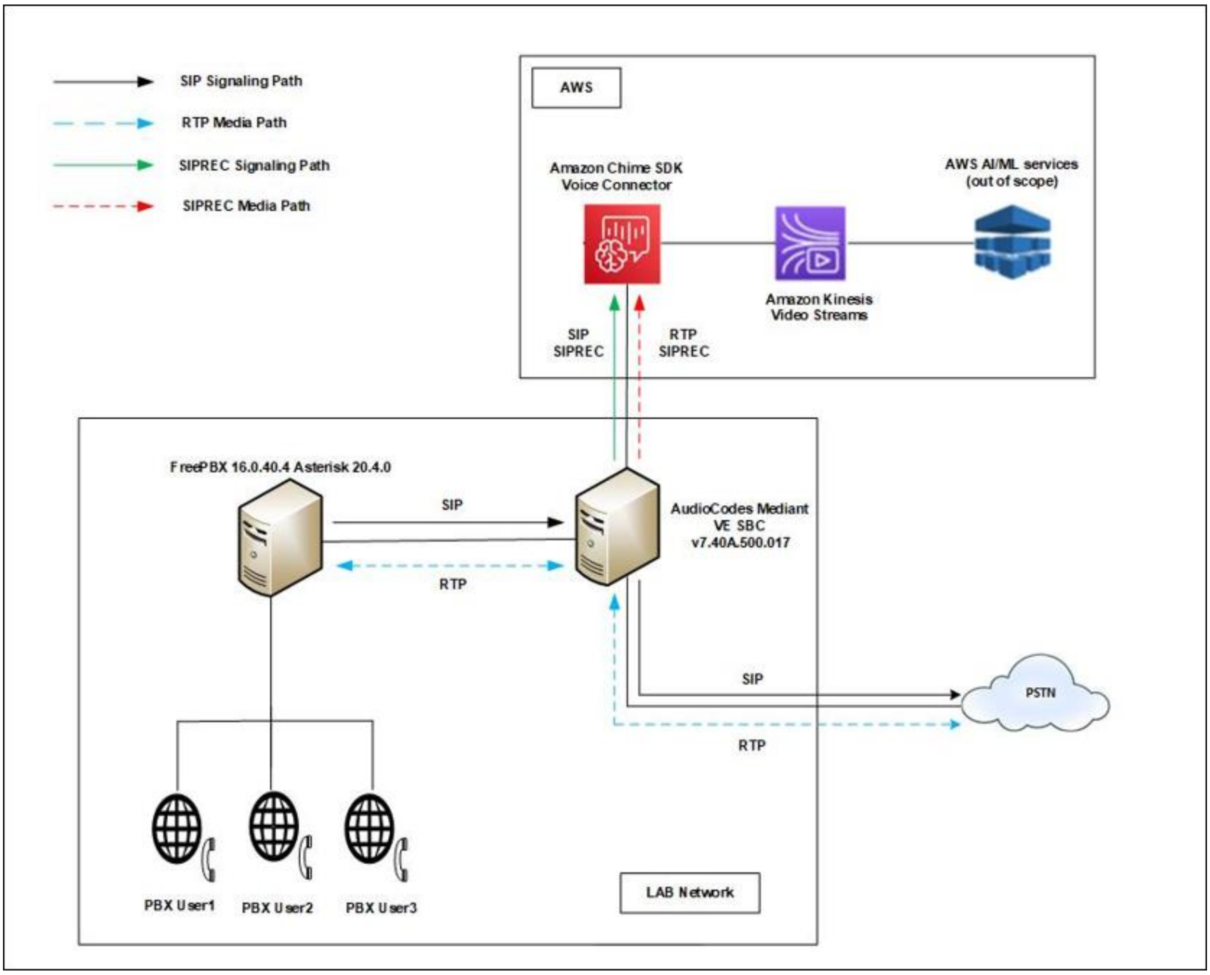

Figure 2: Signaling and Media Flow

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

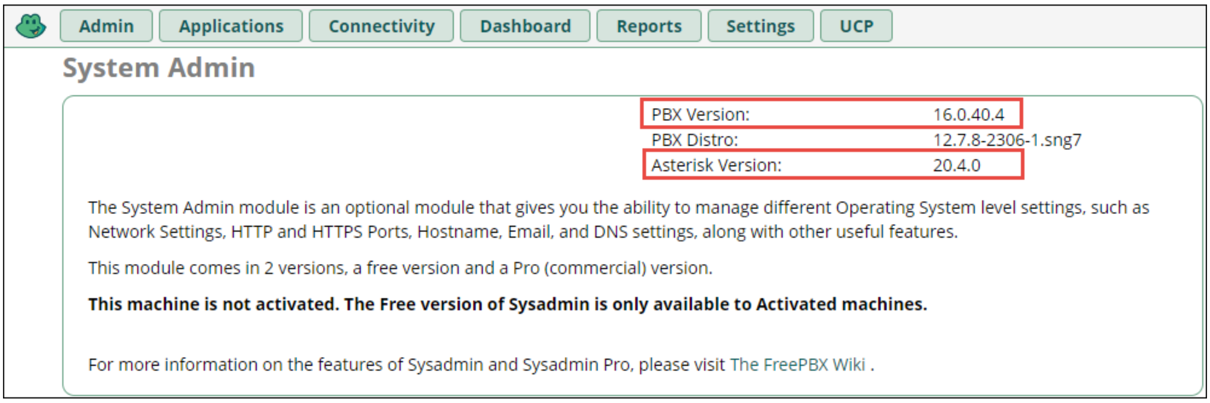

Figure 3: FreePBX Asterisk Version

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

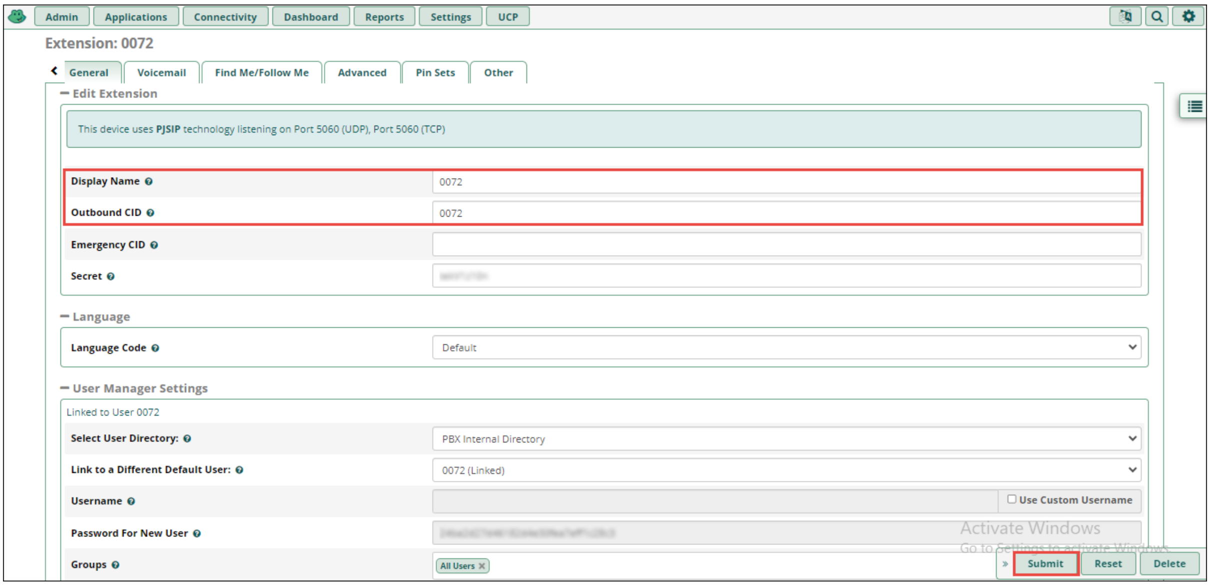

Figure 4: Asterisk Extension

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

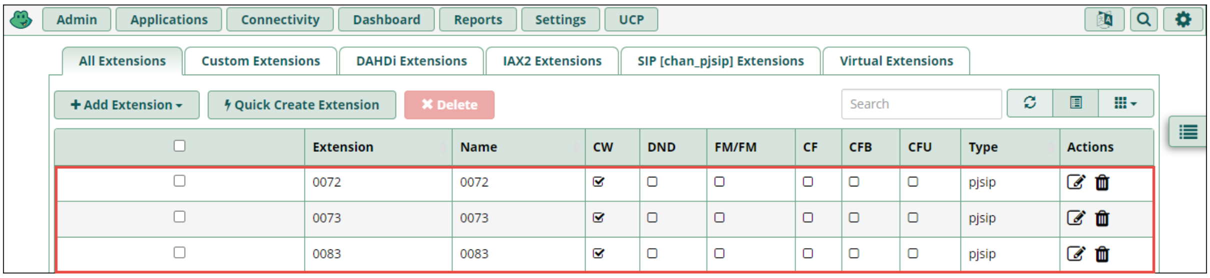

Figure 5: Asterisk Extension List

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

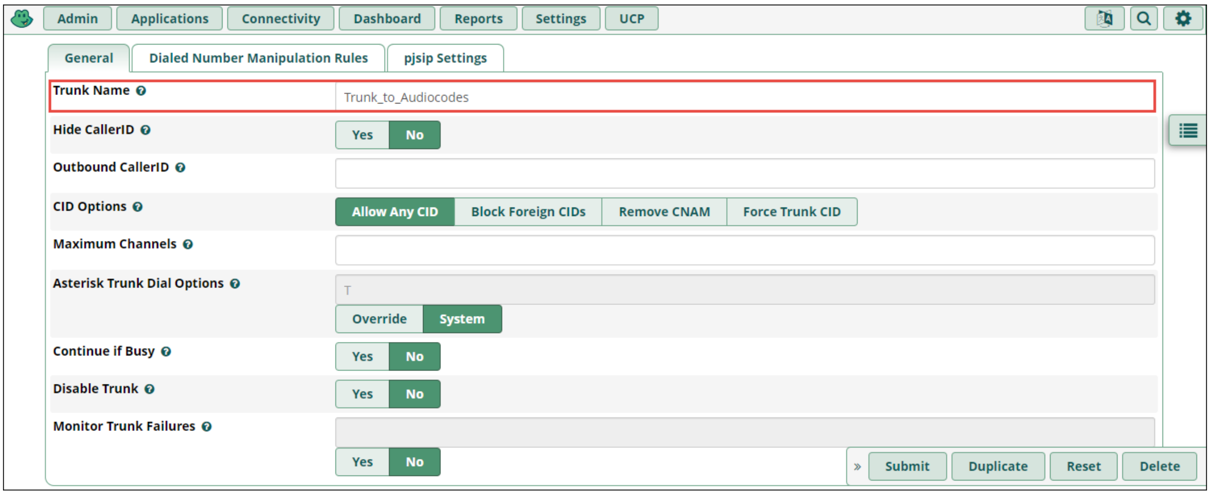

Figure 6: Asterisk Trunk

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

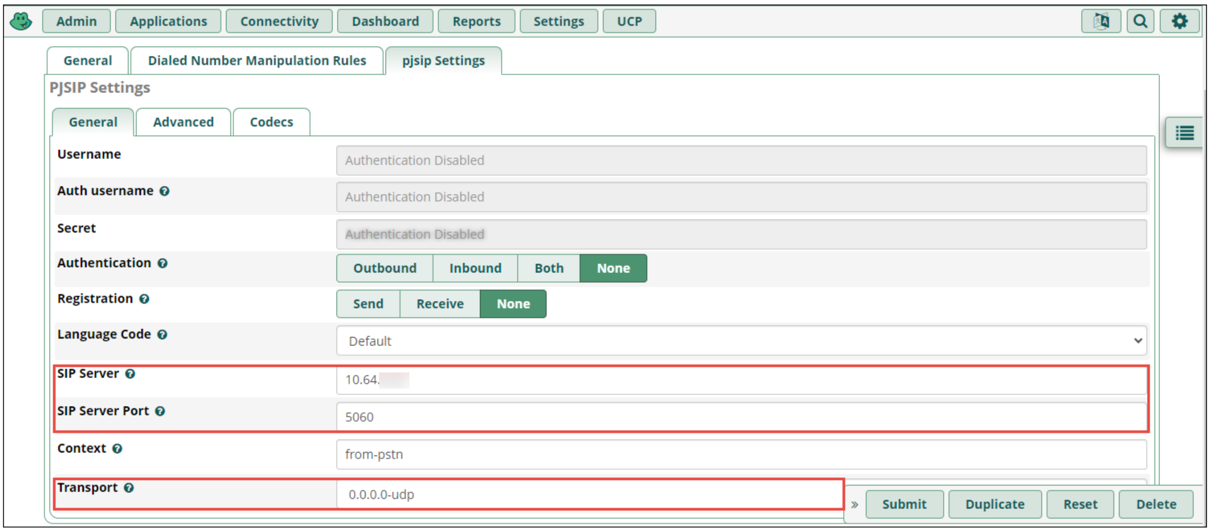

Figure 7: Asterisk Trunk Continuation

Navigate **to Pjsip settings → Codecs**

Enable Ulaw

Click Submit

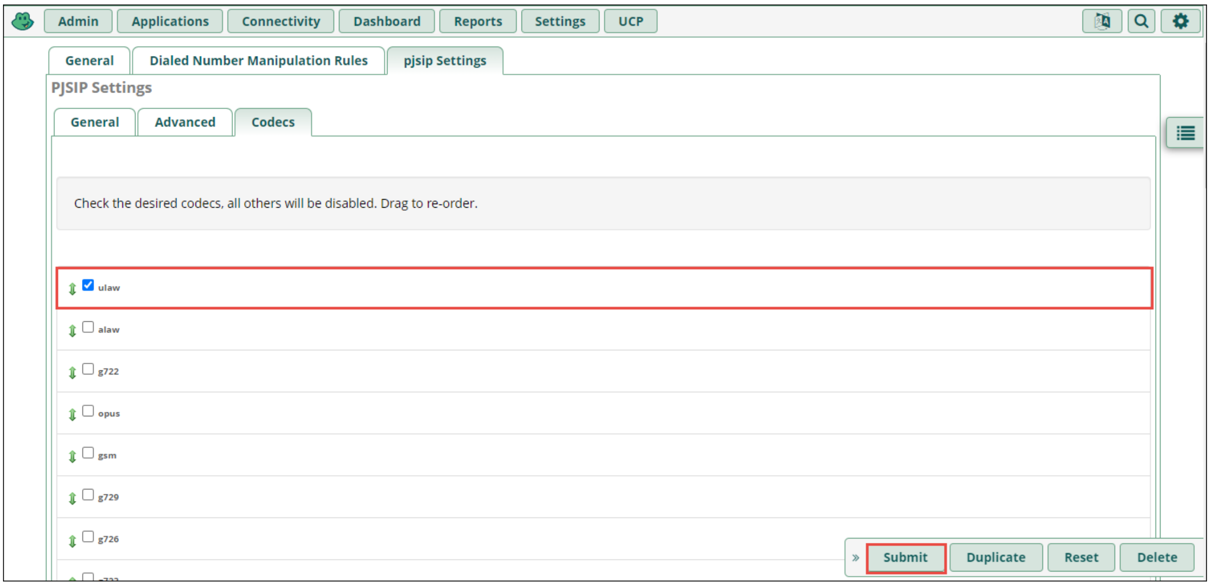

Figure 8: Asterisk Trunk Continuation

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

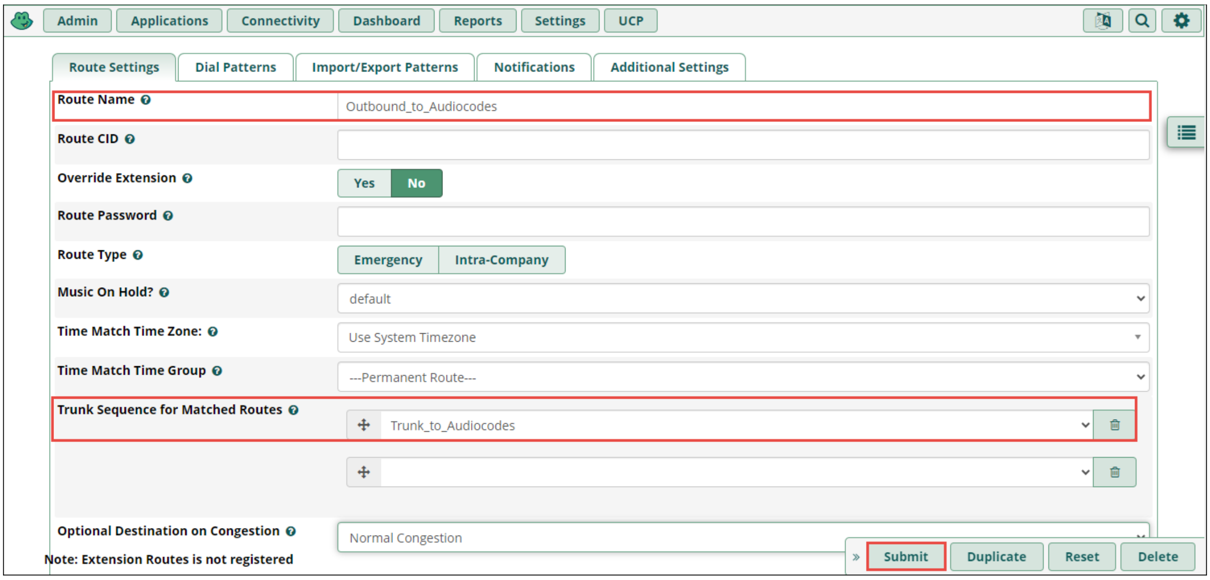

Figure 9: Asterisk Outbound Route

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

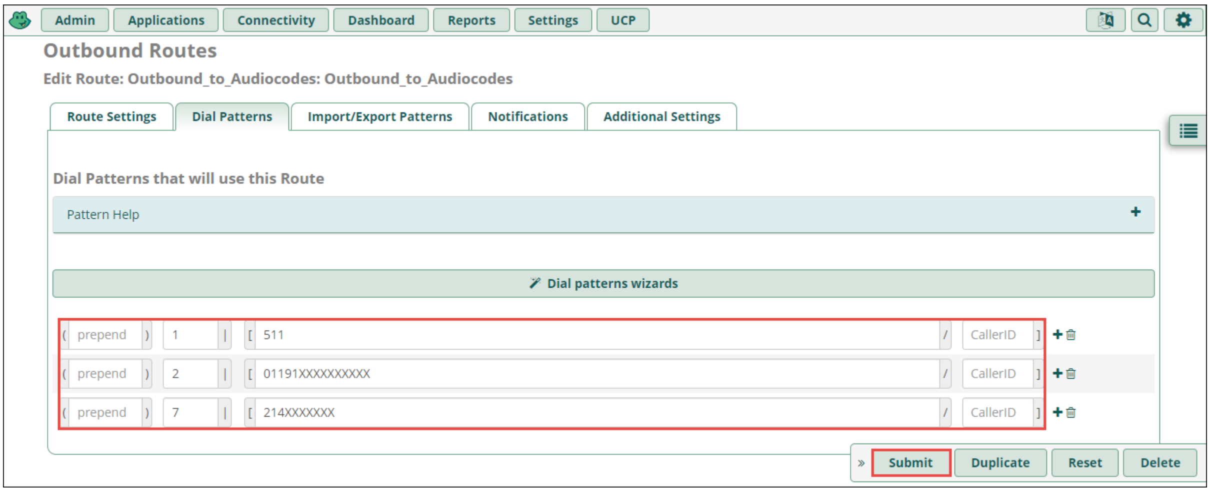

Figure 10: Asterisk Outbound Route Continuation

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

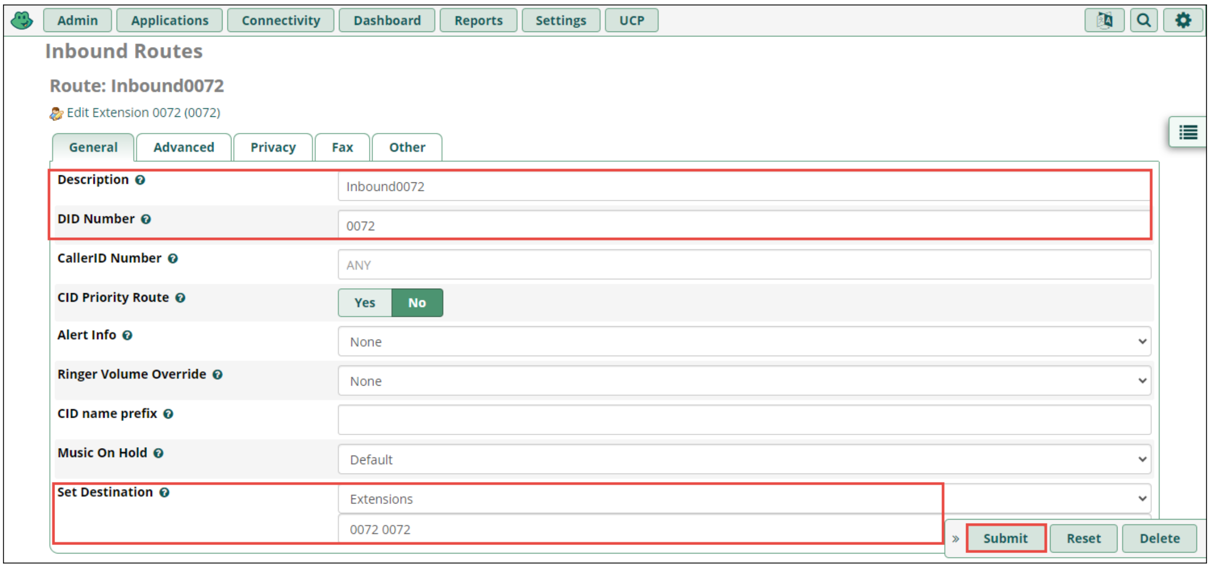

Figure 11: Asterisk Inbound Route

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

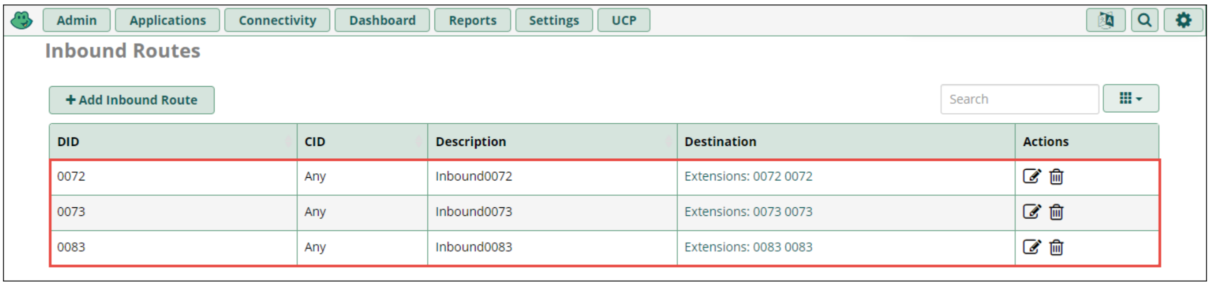

Figure 12: Asterisk inbound Routes List

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**

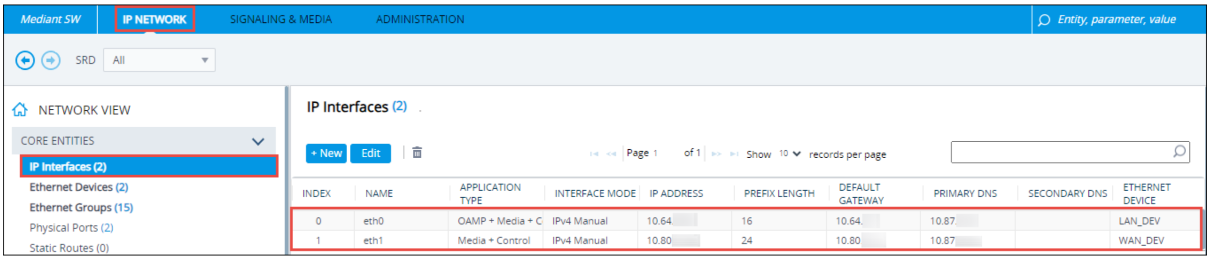

Figure 13: IP Interfaces List

- **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

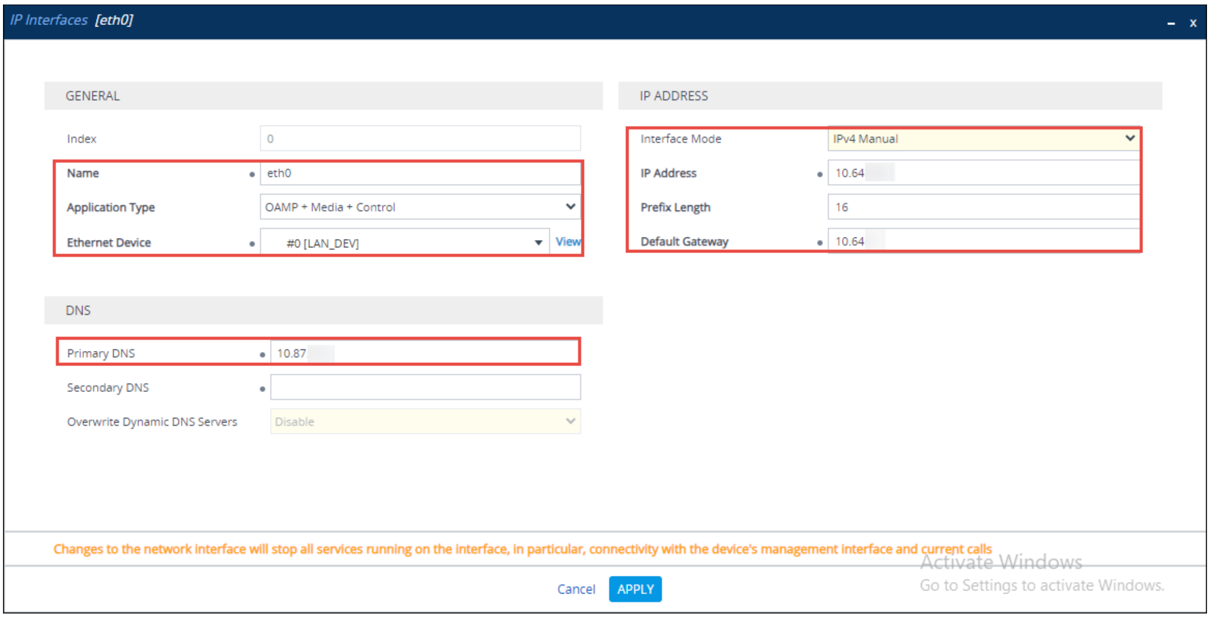

Figure 14: LAN IP Interface

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

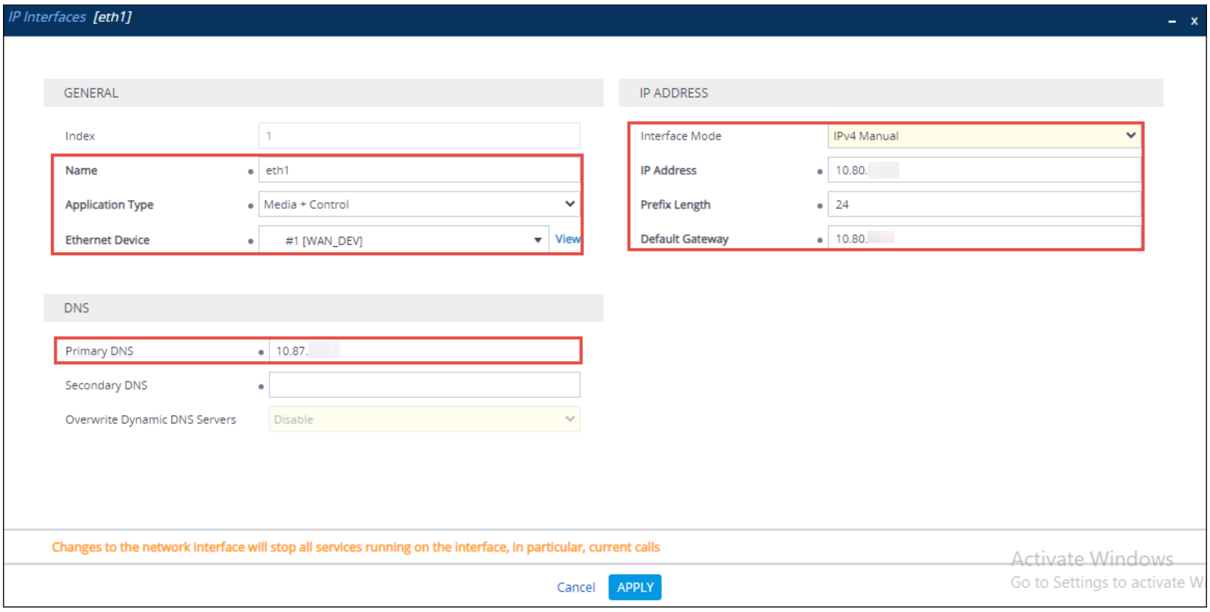

Figure 15: WAN IP Interface

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**

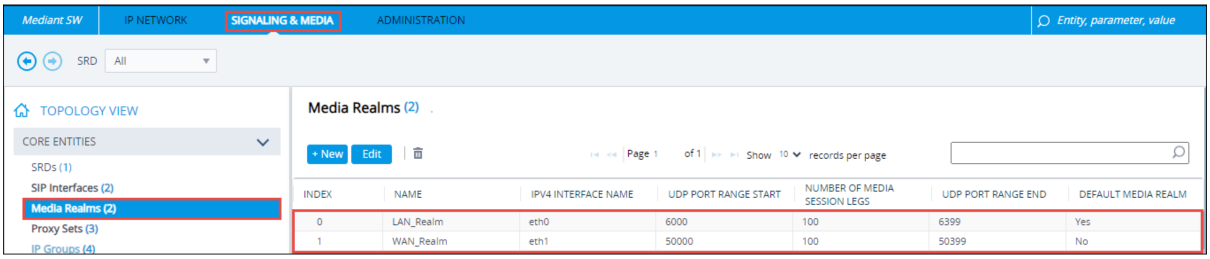

Figure 16: Media Realms List

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

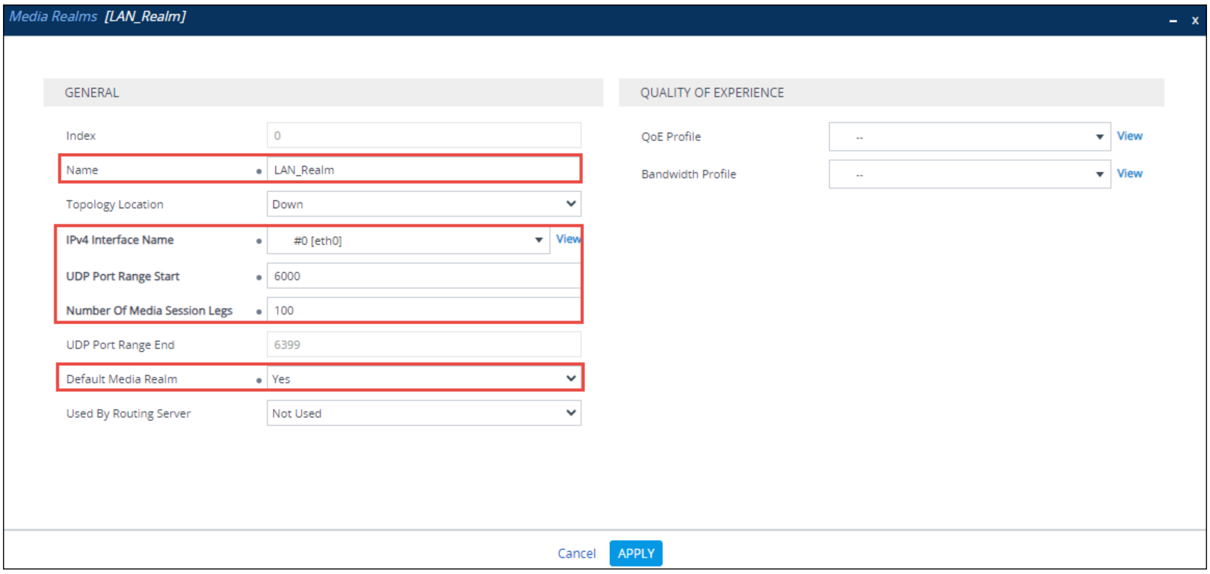

Figure 17: Media Realm for LAN Interface

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

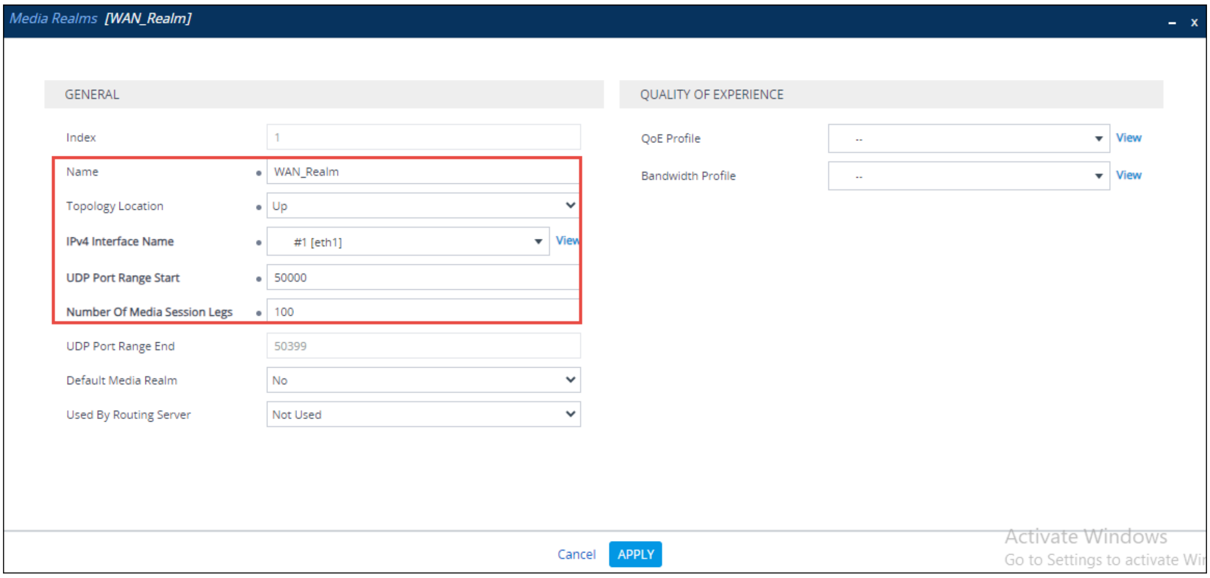

Figure 18: Media Realm for WAN Interface

4.3.3 SRD

Default Signaling Routing Domain (SRD) is used.

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

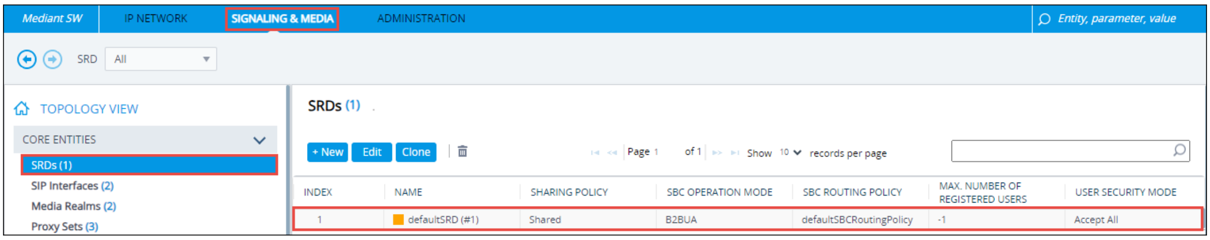

Figure 19: SRD

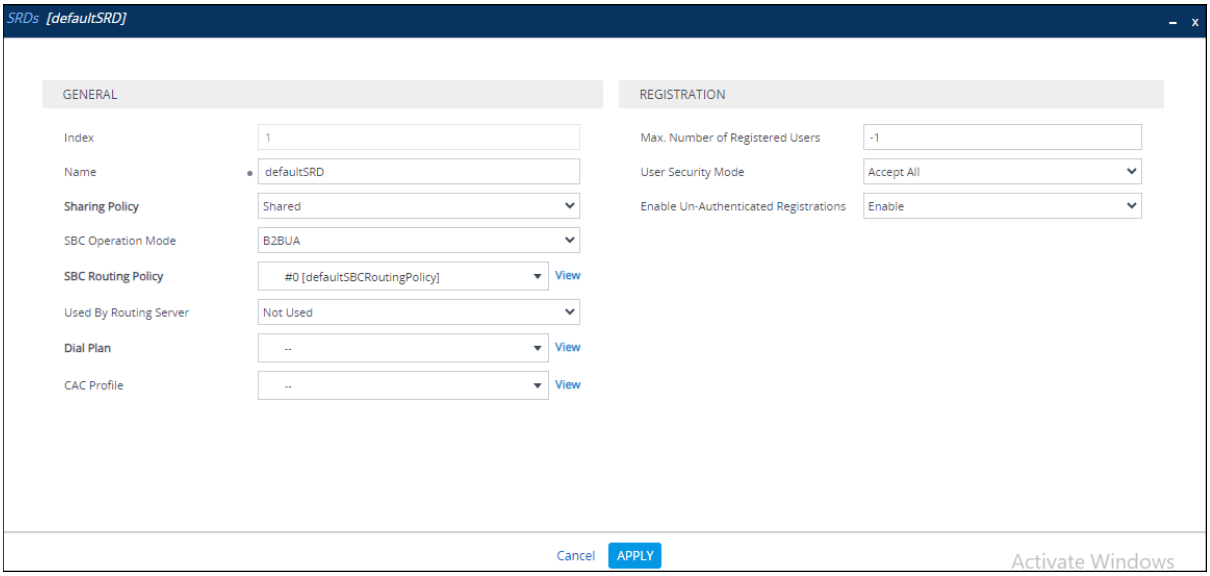

Figure 20: Default SRD

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**

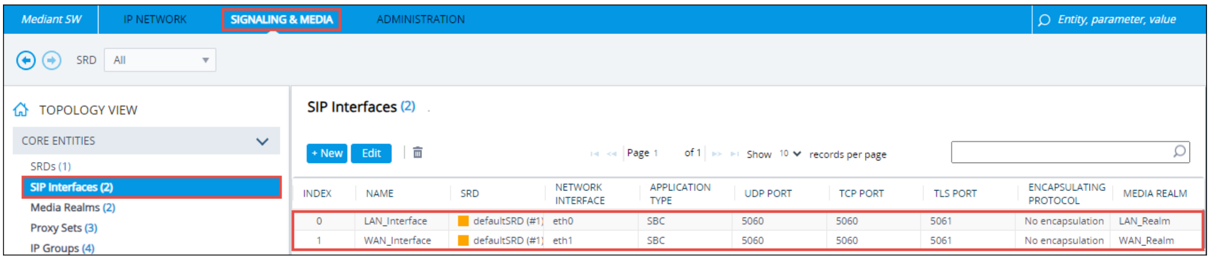

Figure 21: SIP Interfaces List

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

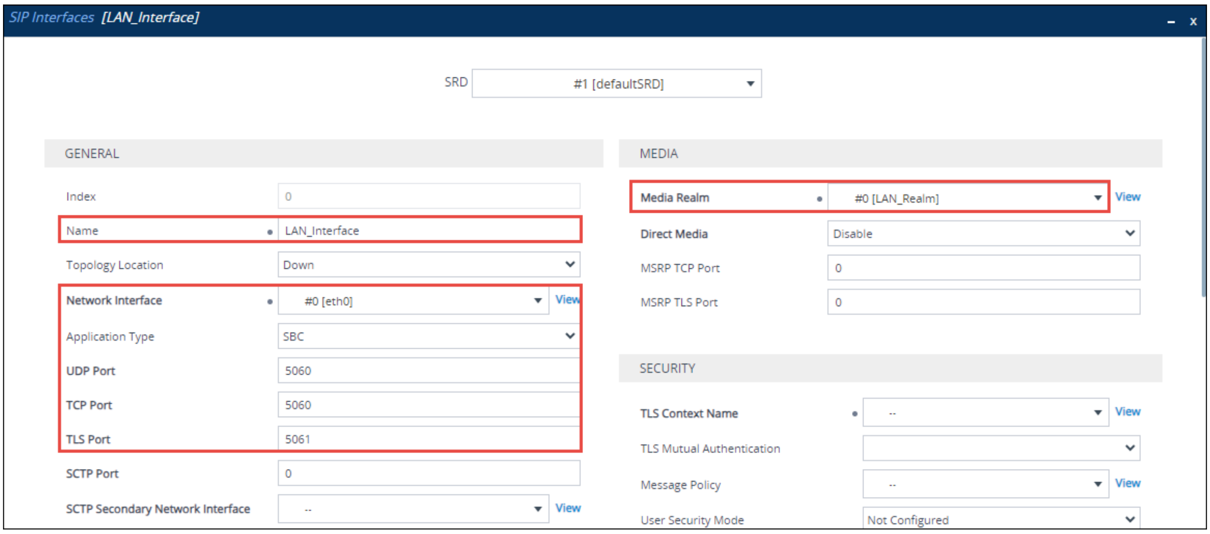

Figure 22: SIP Interface LAN

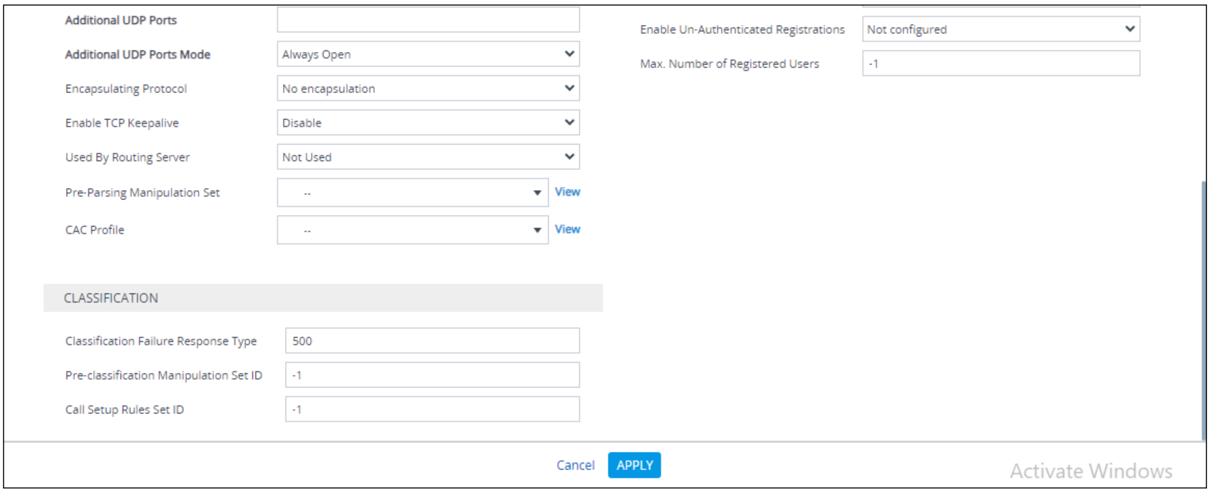

Figure 23: SIP Interface LAN Continuation

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

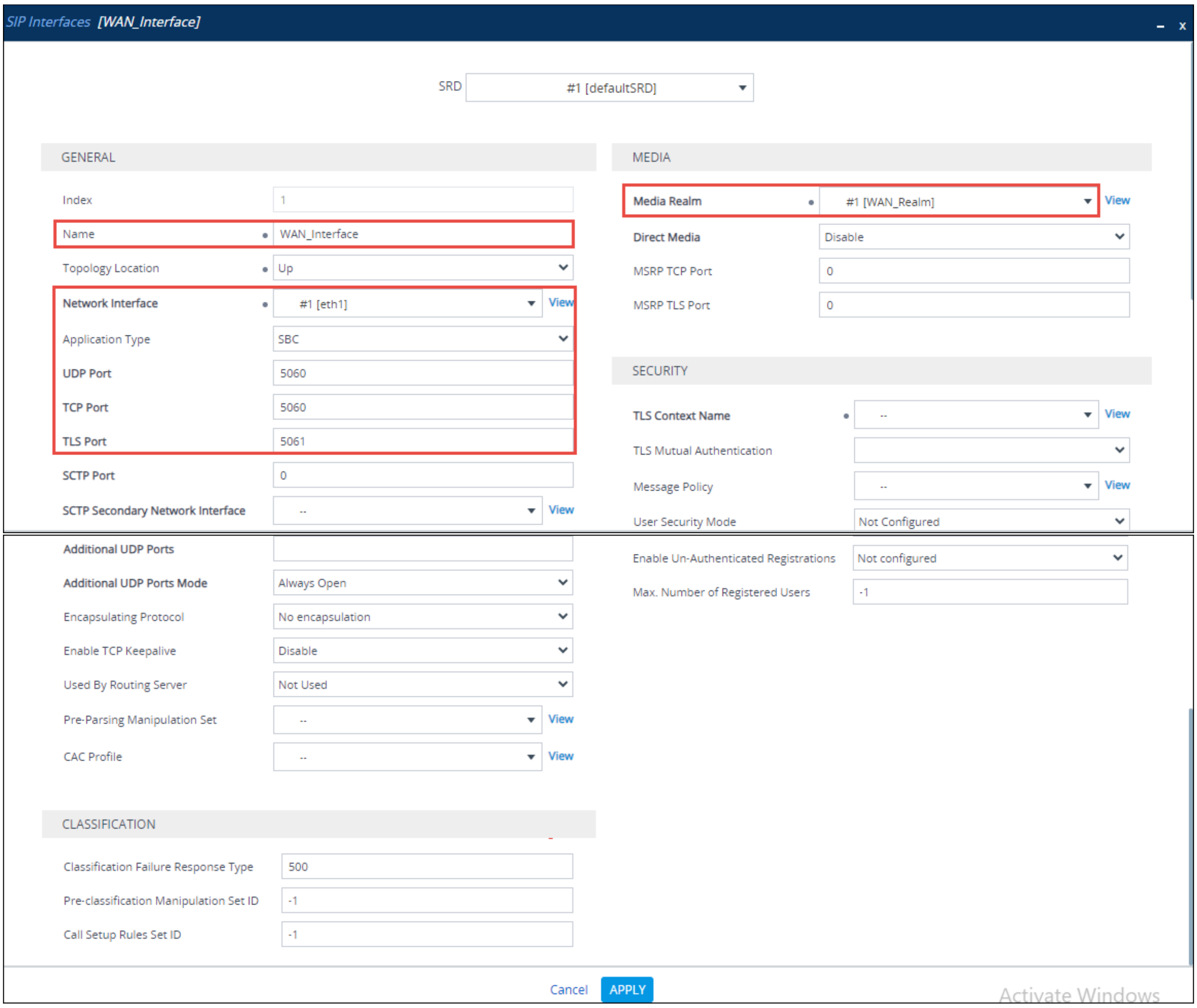

Figure 24: SIP Interface WAN

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**

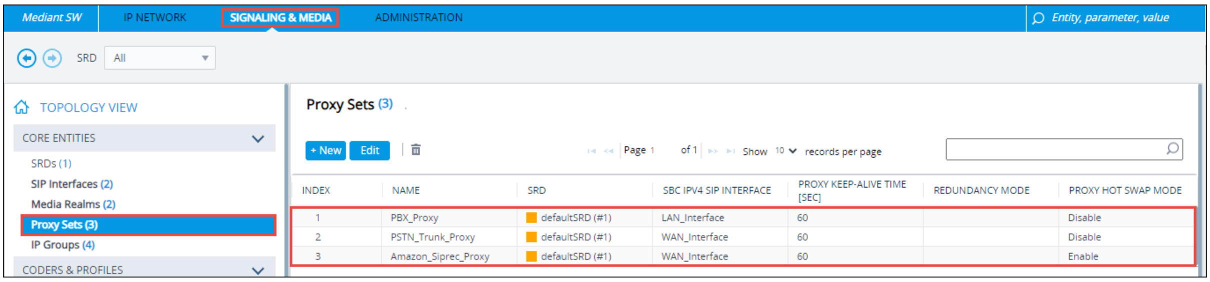

Figure 25: Proxy Sets List

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

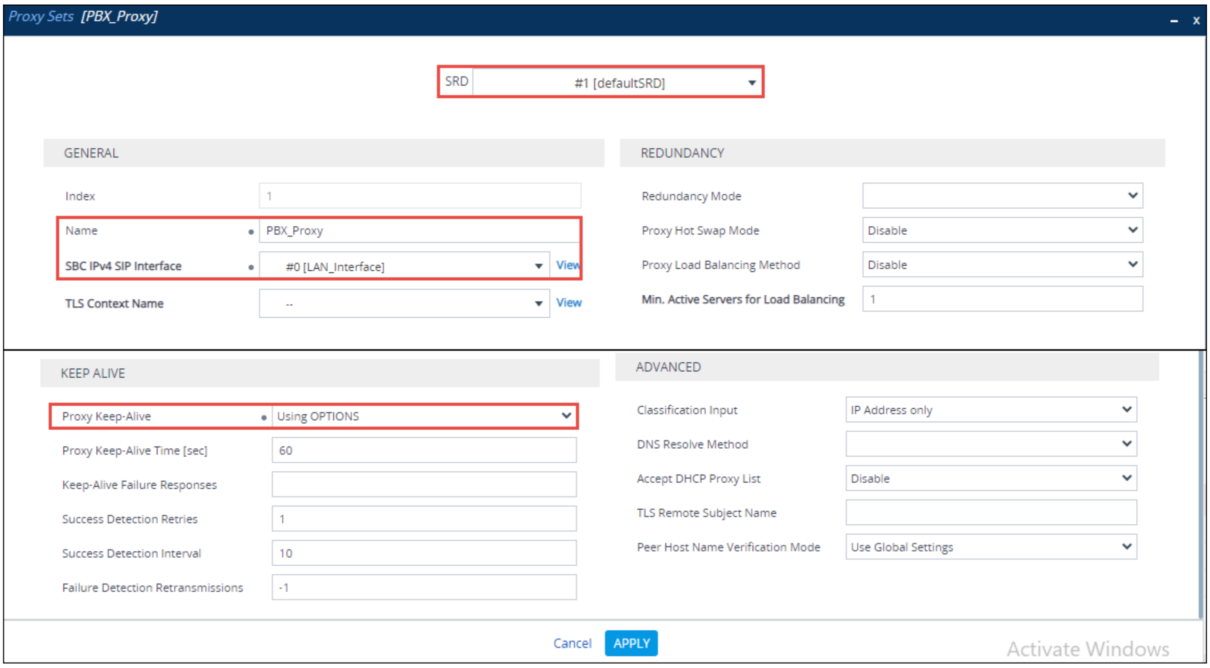

Figure 26: Proxy Set PBX

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

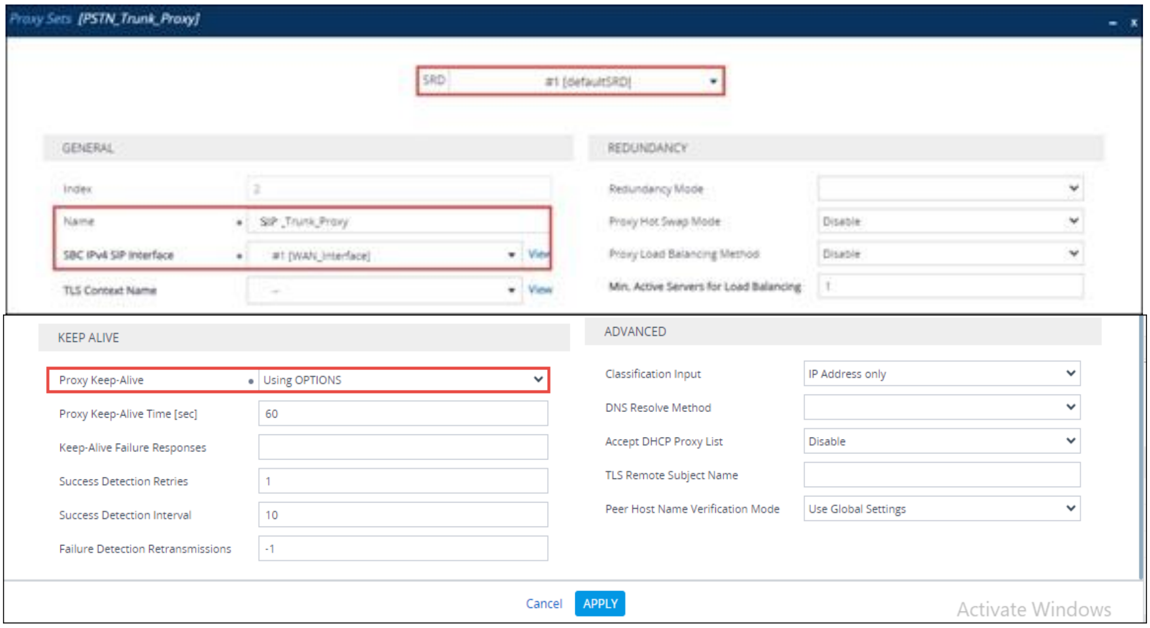

Figure 27: Proxy Set PSTN

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

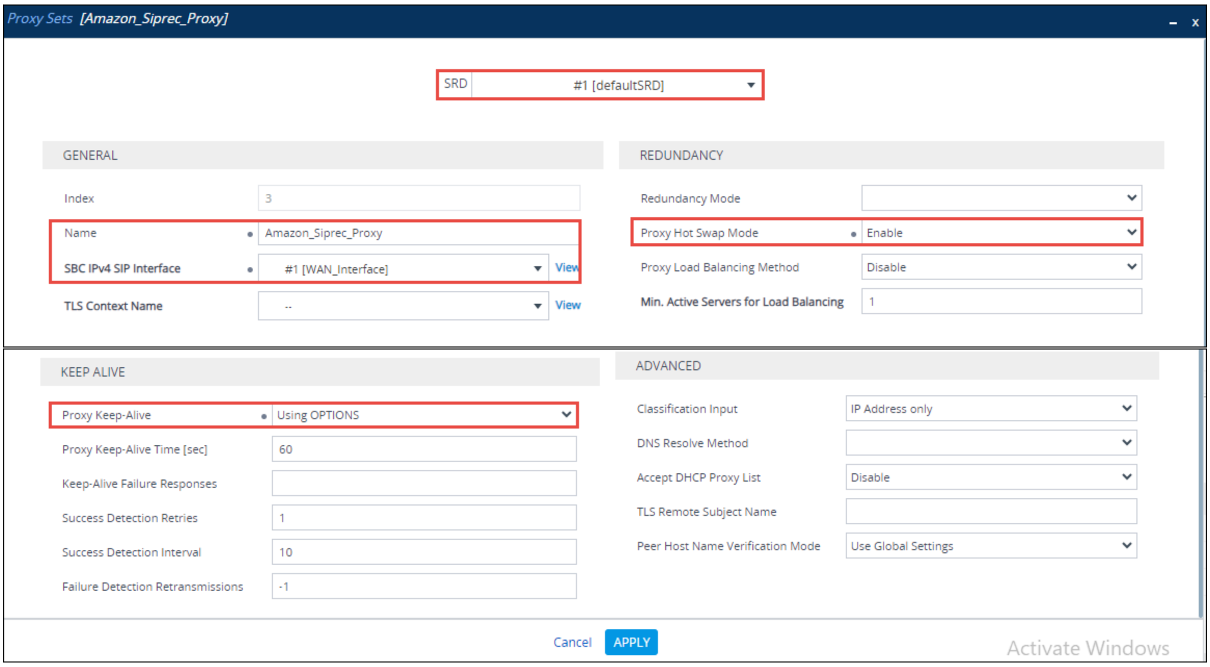

Figure 28: Proxy Set Amazon SIPREC

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

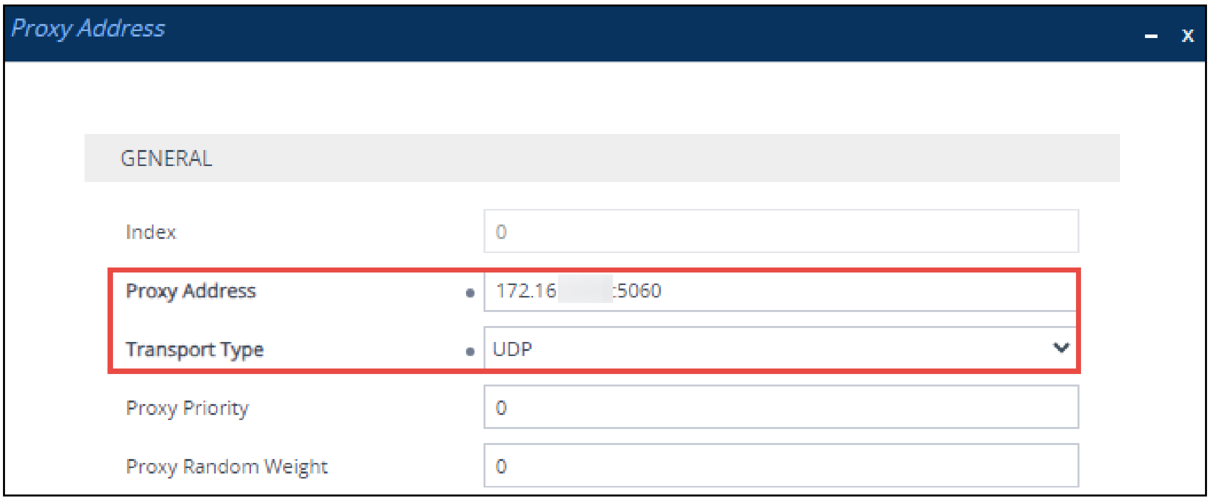

Figure 29: Proxy Address PBX

4.3.6.2 PSTN Proxy Address

Proxy Address: 10.80.X.X:5060
Transport Type: UDP

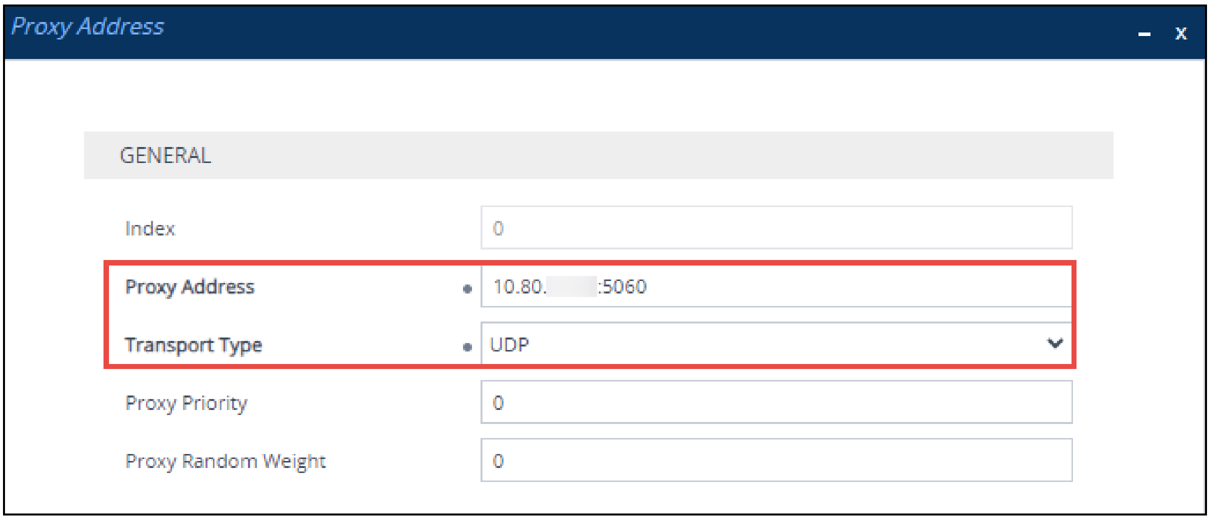

Figure 30: Proxy Address PSTN

4.3.6.3 Amazon Proxy Address

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

Transport Type: UDP

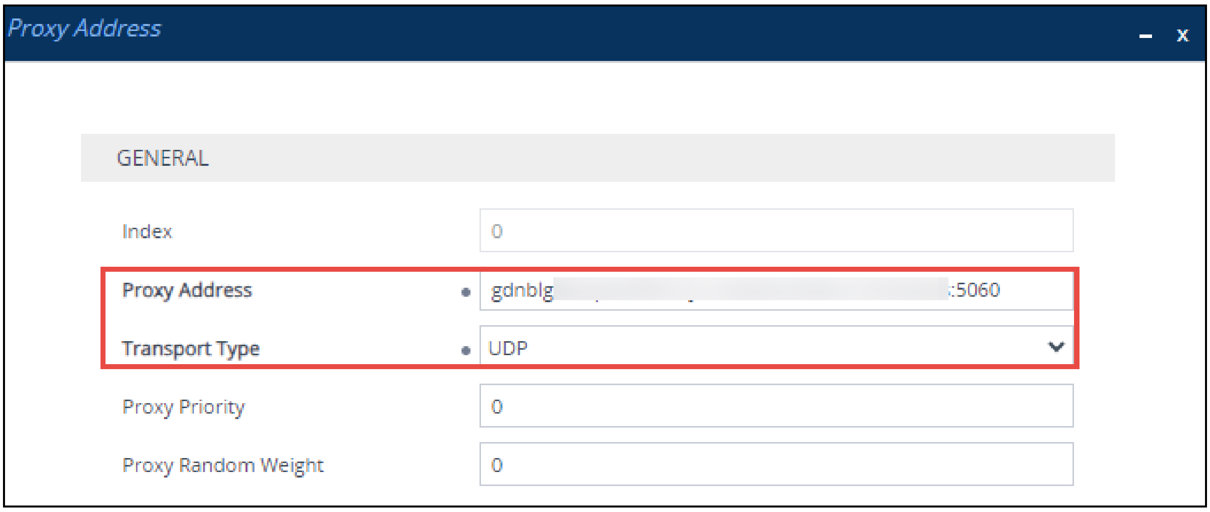

Figure 31: Proxy Address Amazon SIPREC

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**

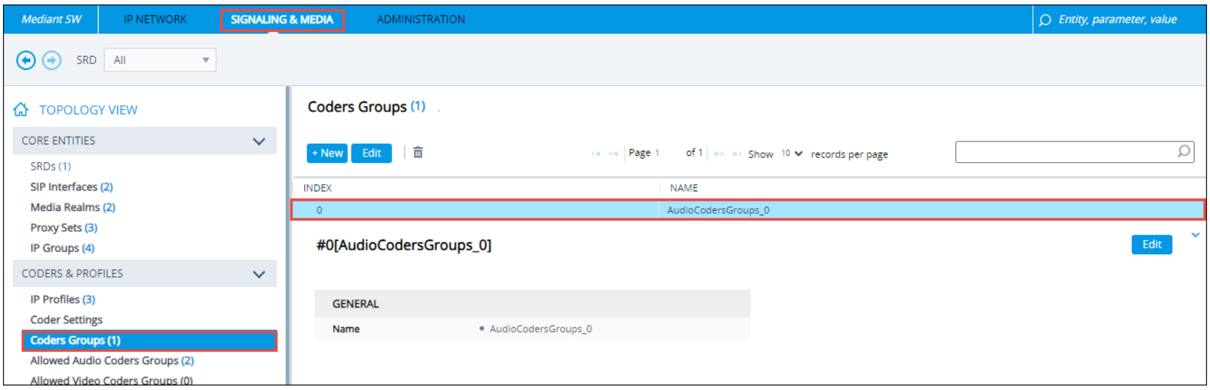

Figure 32: Coders Group

4.3.7.1 Coders Table

Enable **G711ulaw**

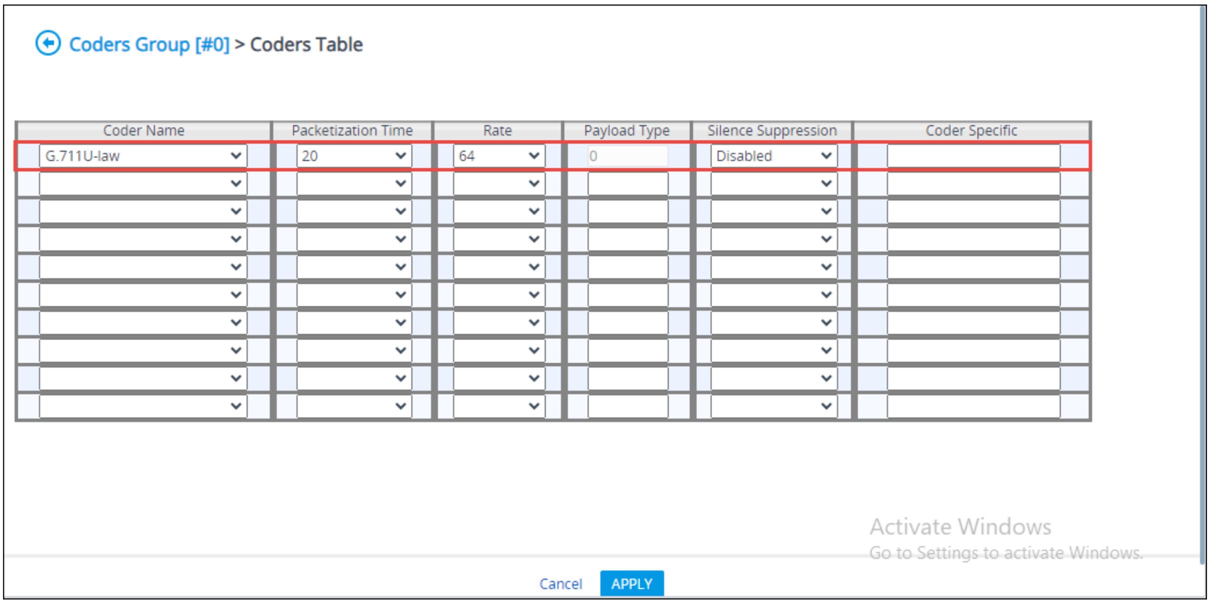

Figure 33: Coders Table

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**

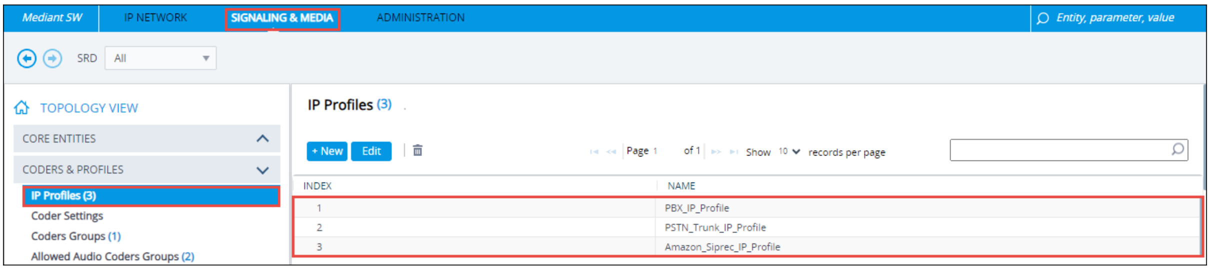

Figure 34: IP Profiles List

4.3.8.1 PBX IP Profile

Name: PBX IP Profile

SBC Media Security Mode: Not Secured

Extension Coders Group: AudioCodersGroups_0

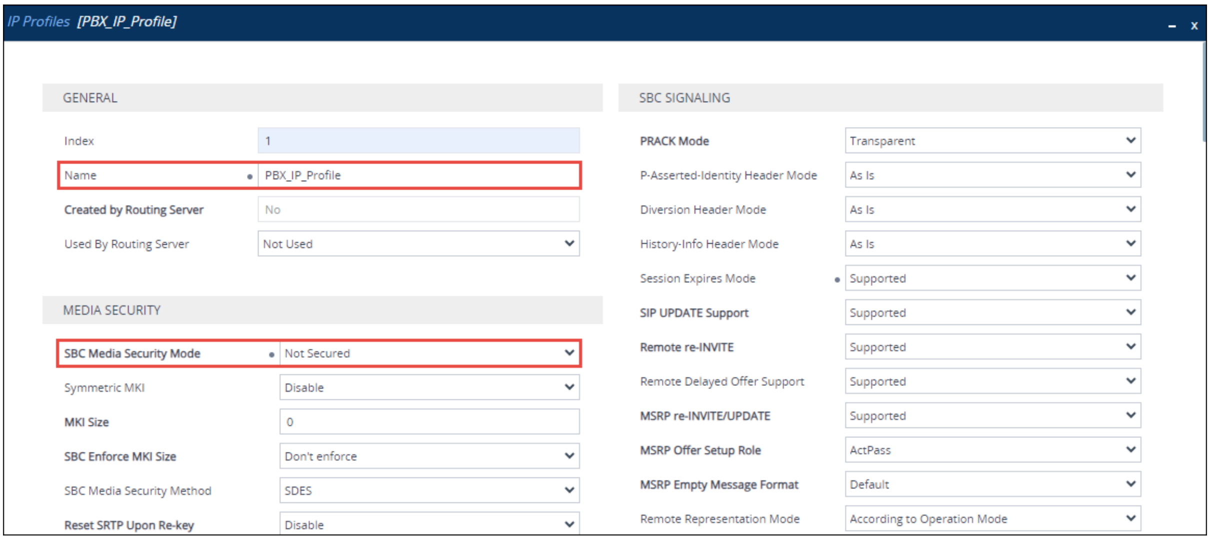

Figure 35: IP Profile PBX

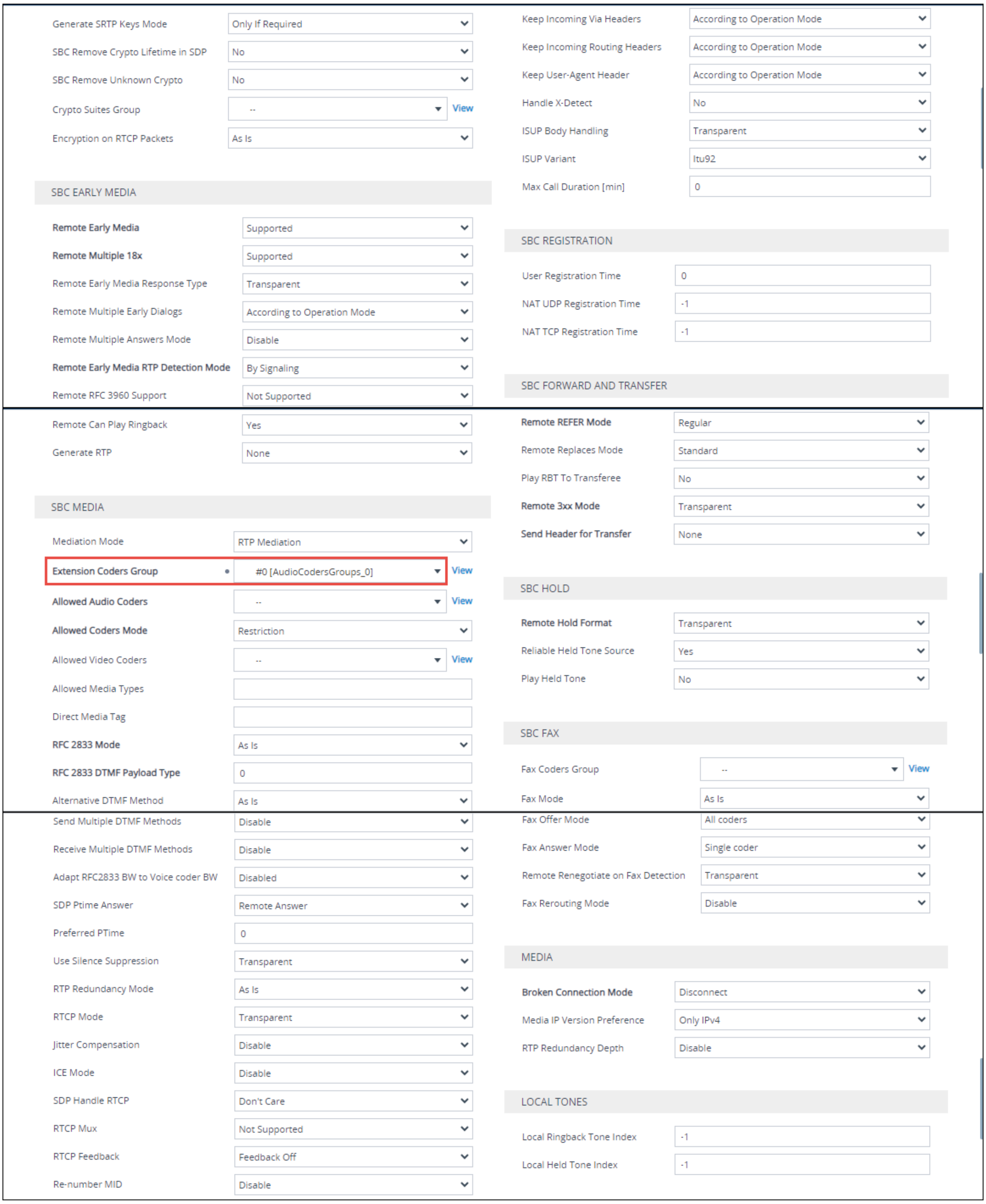

Figure 36: IP Profile PBX Continuation

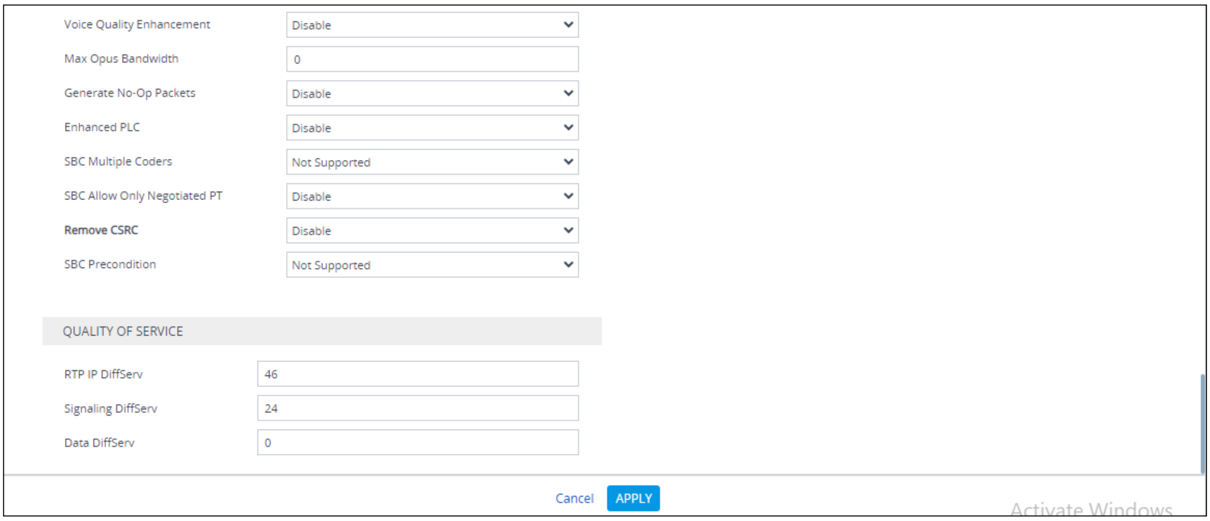

Figure 37: IP Profile PBX Continuation

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

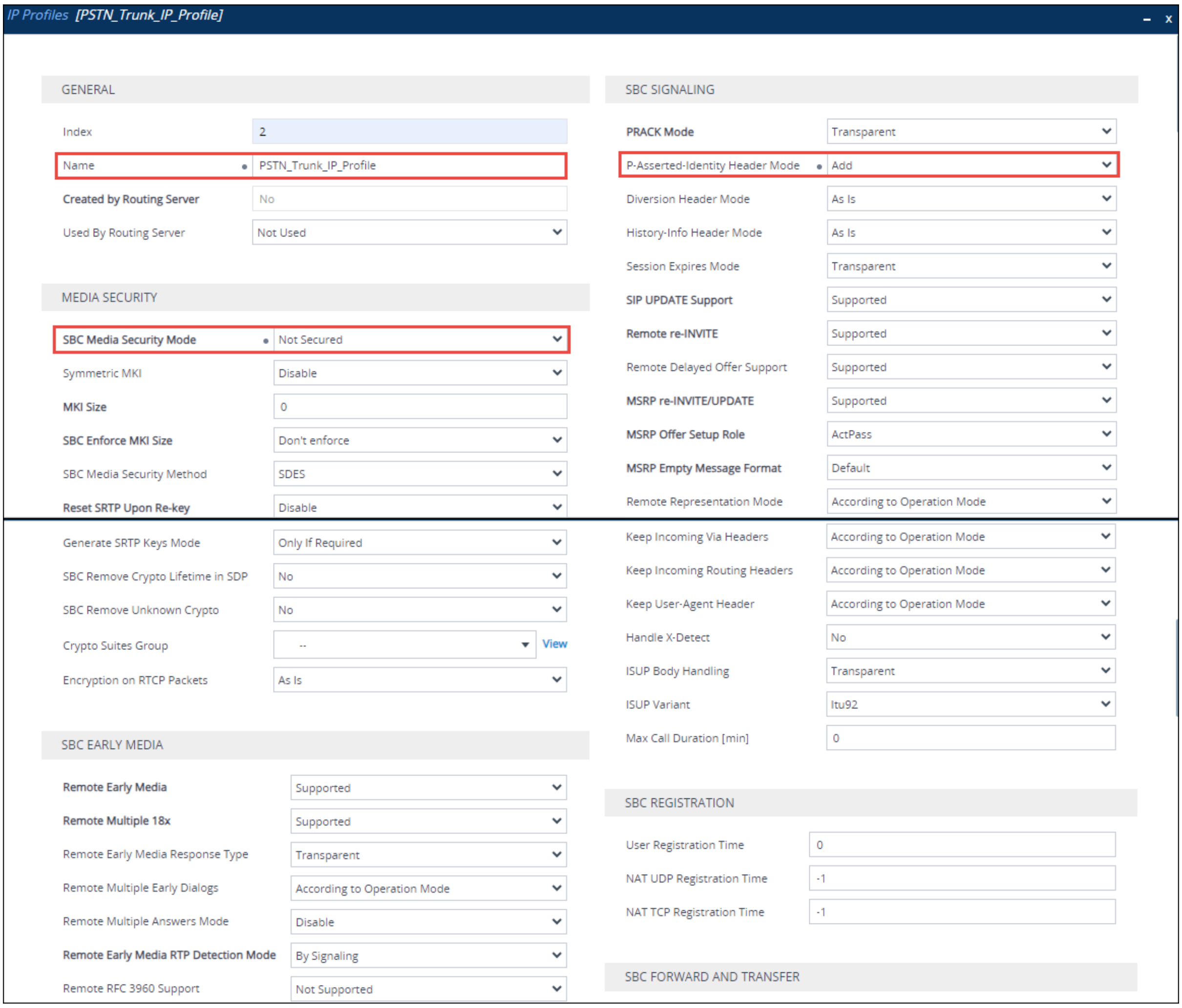

Figure 38: IP Profile PSTN

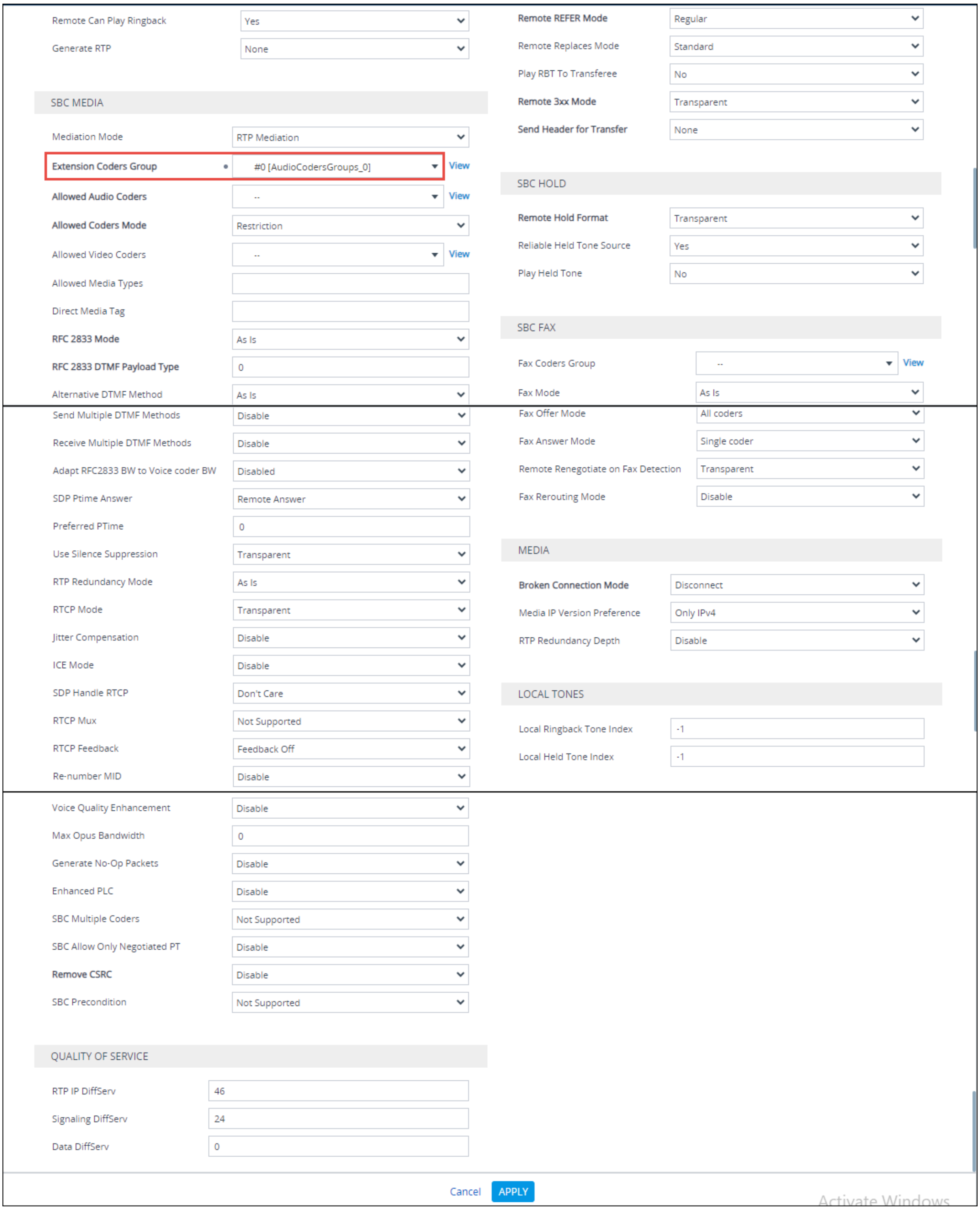

Figure 39: IP Profile PSTN Continuation

4.3.8.3 Amazon SIPREC IP Profile

Name: Amazon Siprec IP Profile

SBC Media Security Mode: Not Secured

Extension Coders Group: AudioCodersGroups_0

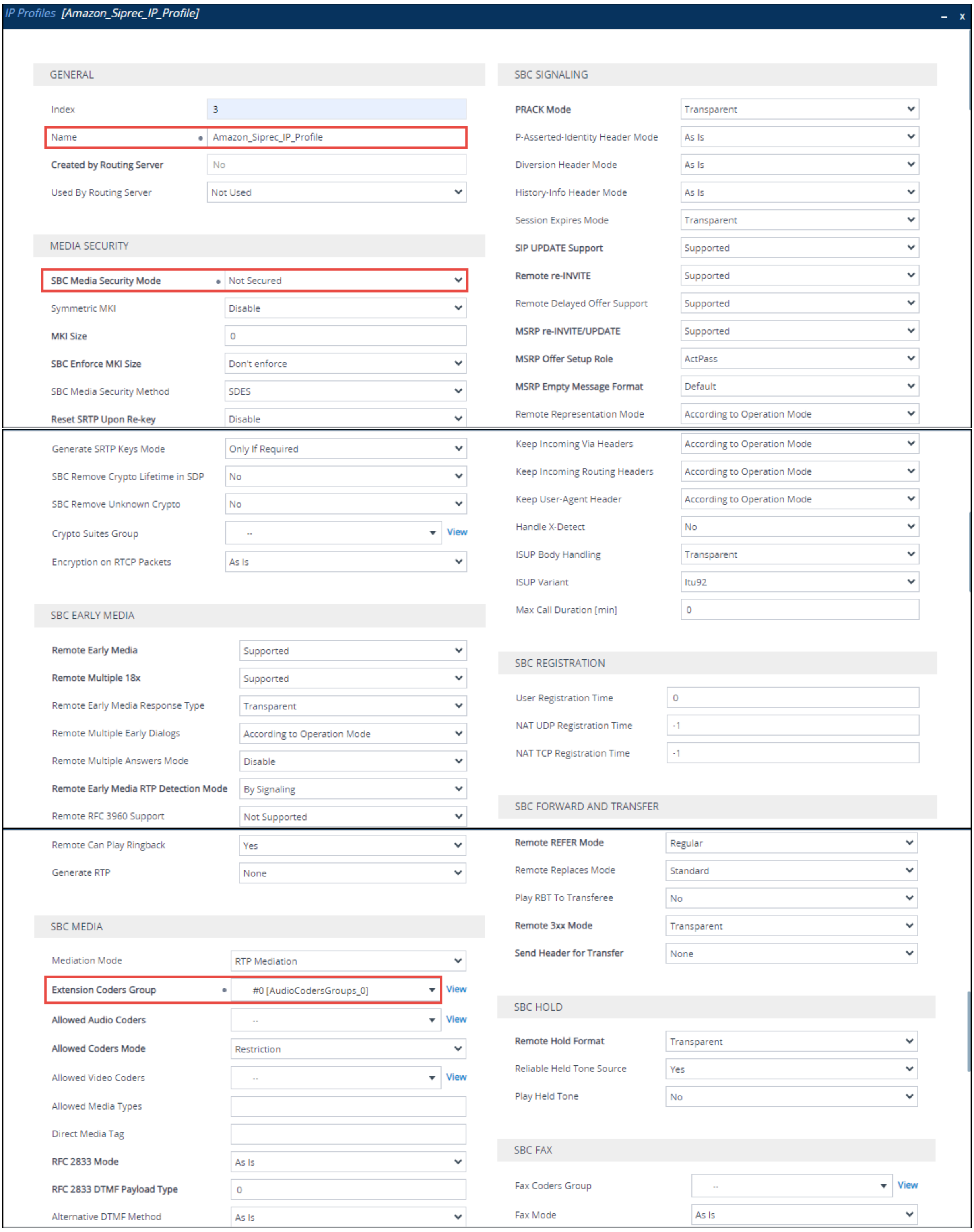

Figure 40: IP Profile Amazon SIPREC

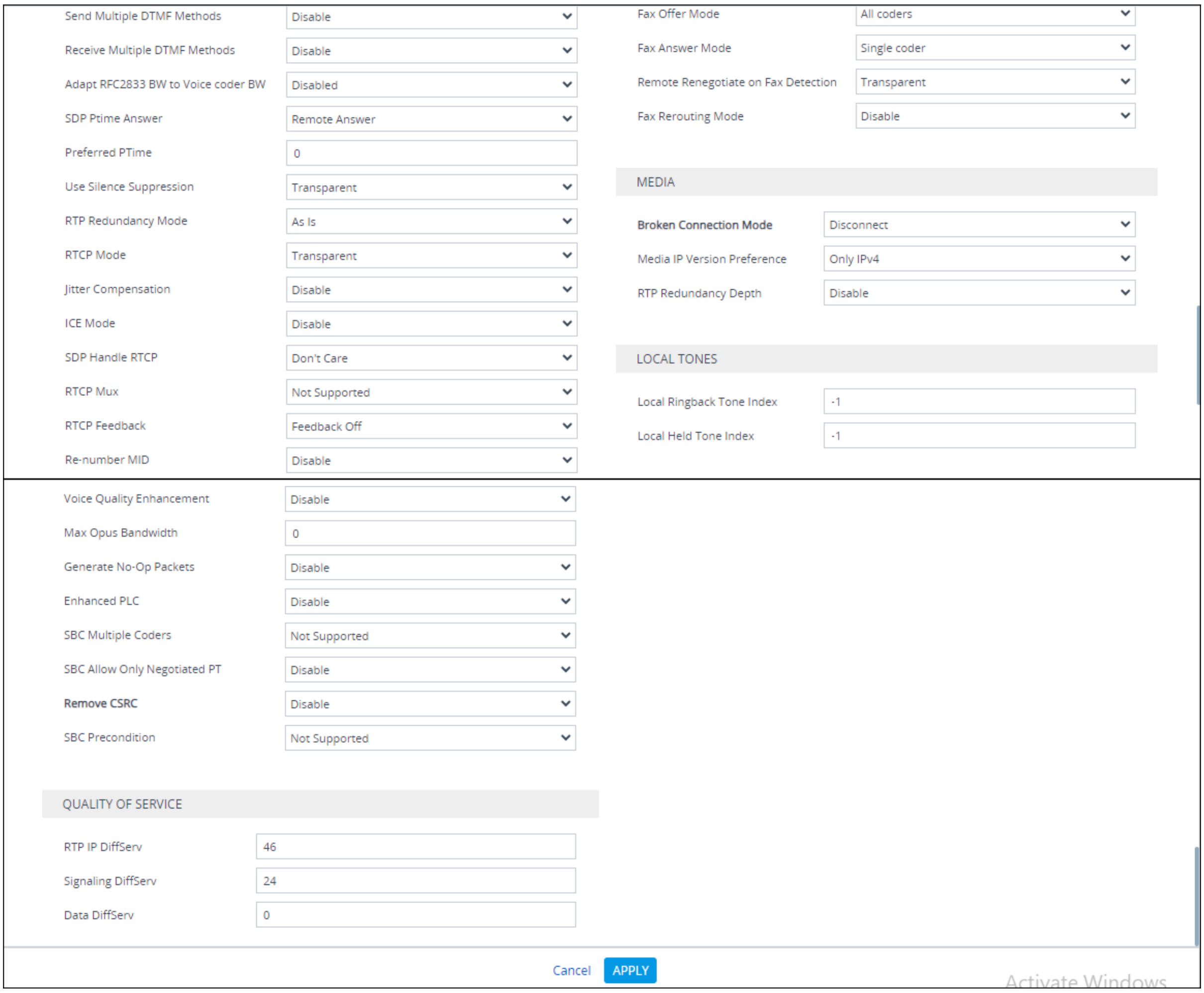

Figure 41: IP Profile Amazon SIPREC Continuation

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**

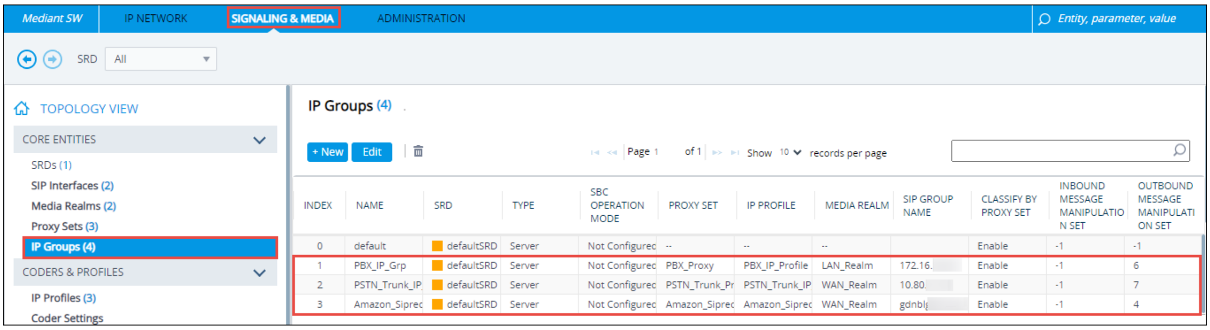

Figure 42: IP Groups List

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)

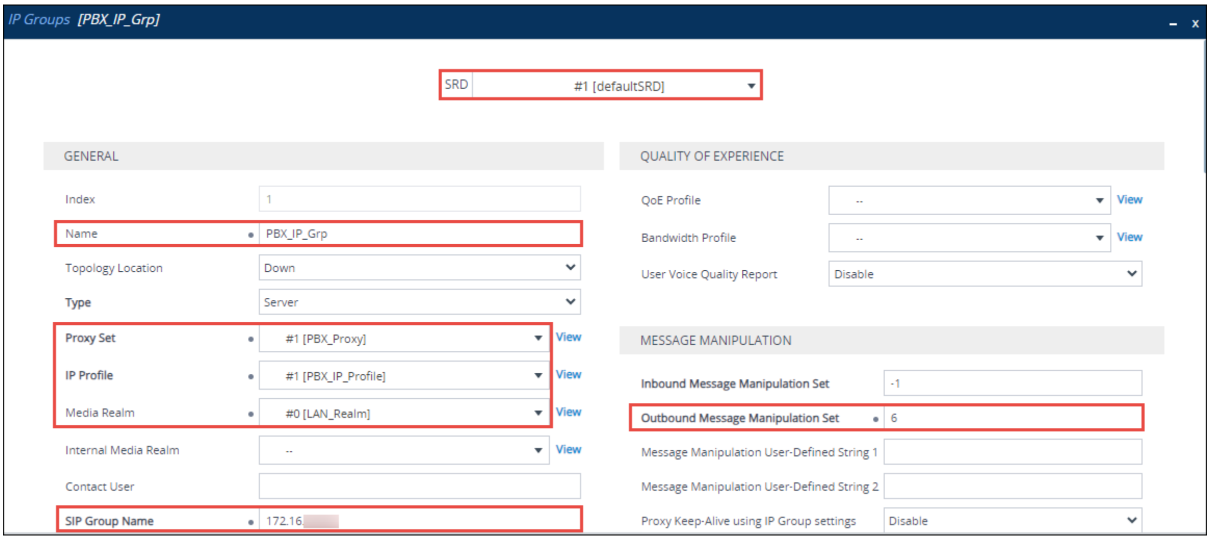

Figure 43: IP Group PBX

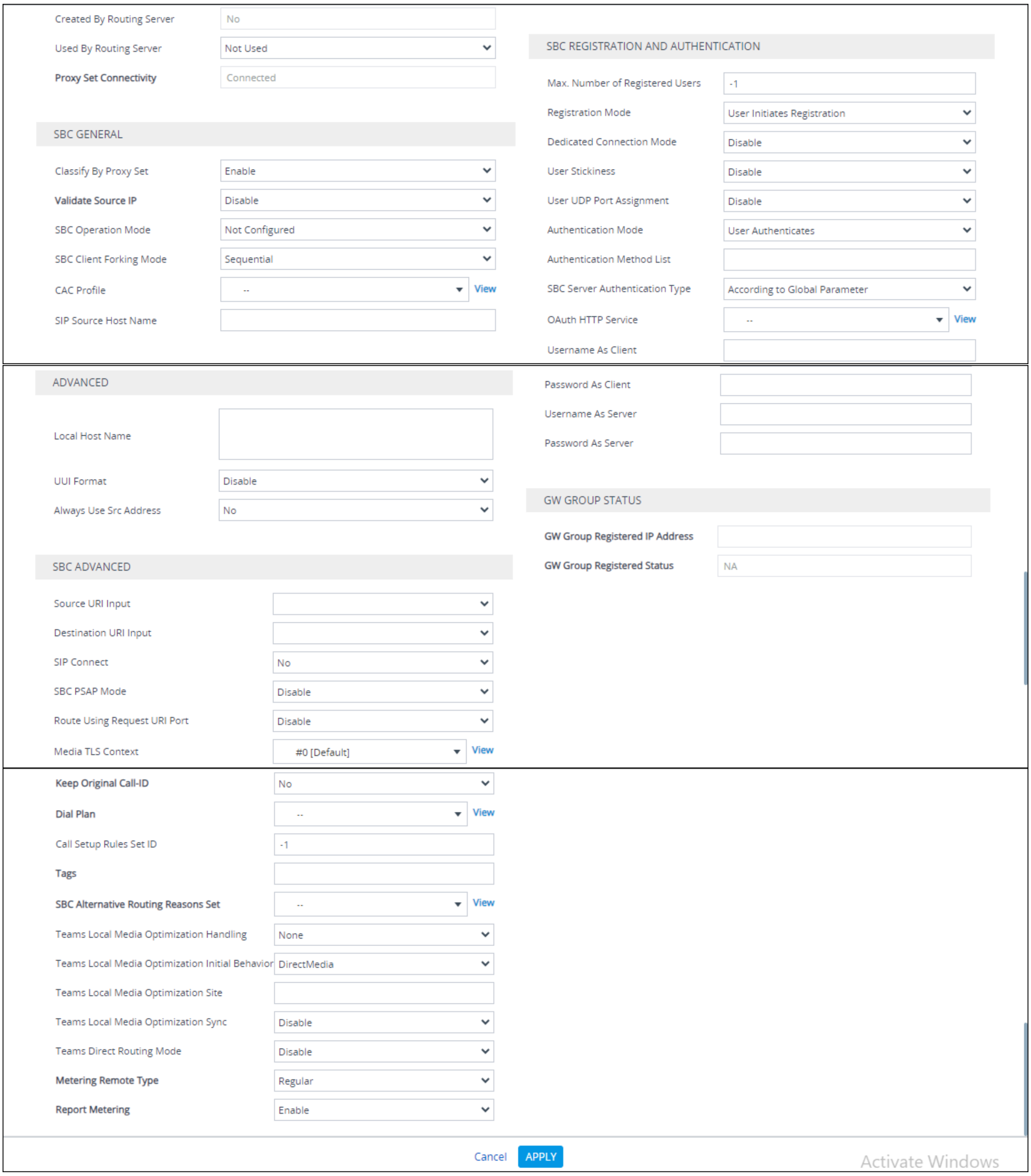

Figure 44: IP group PBX Continuation

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)

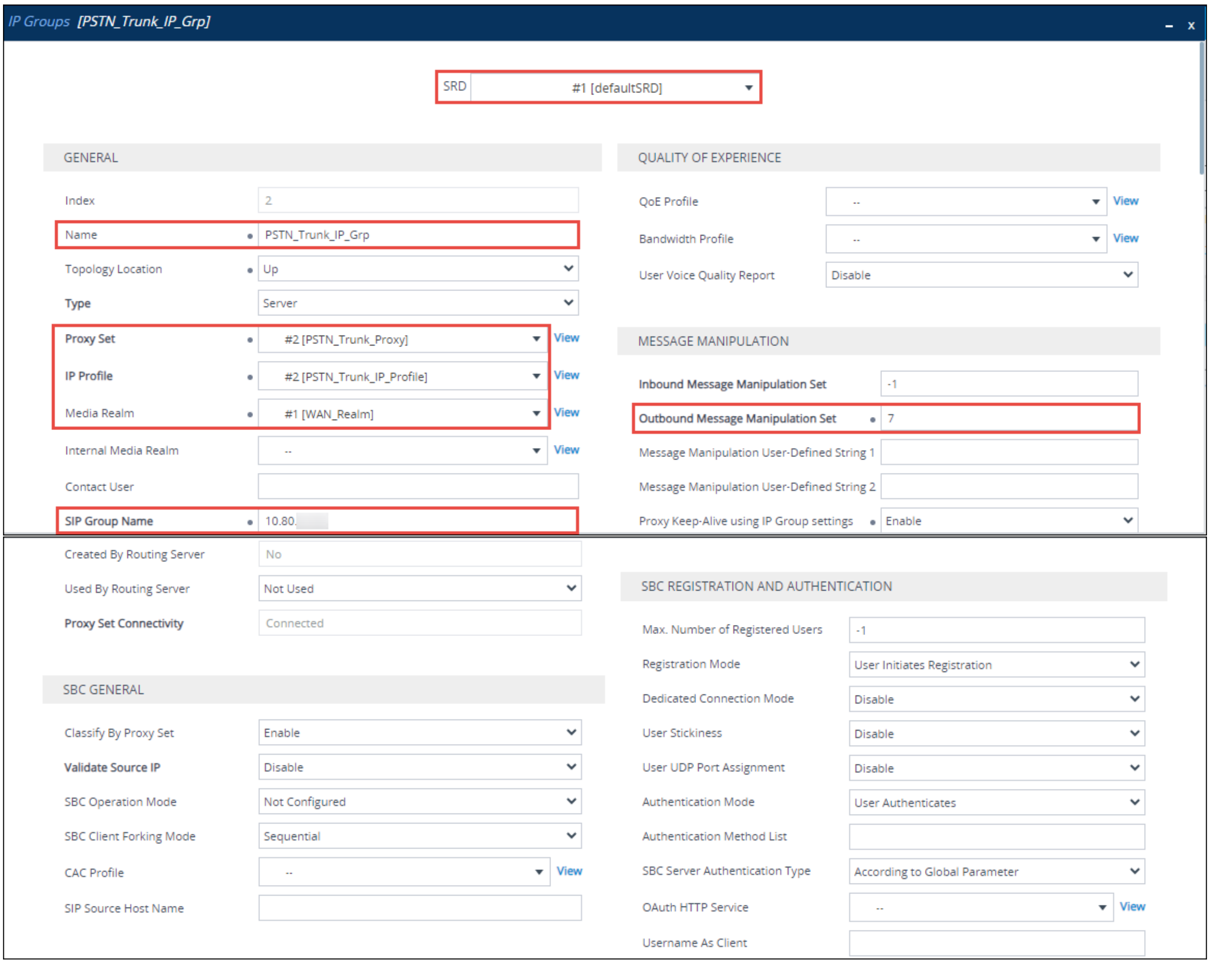

Figure 45: IP Group PSTN

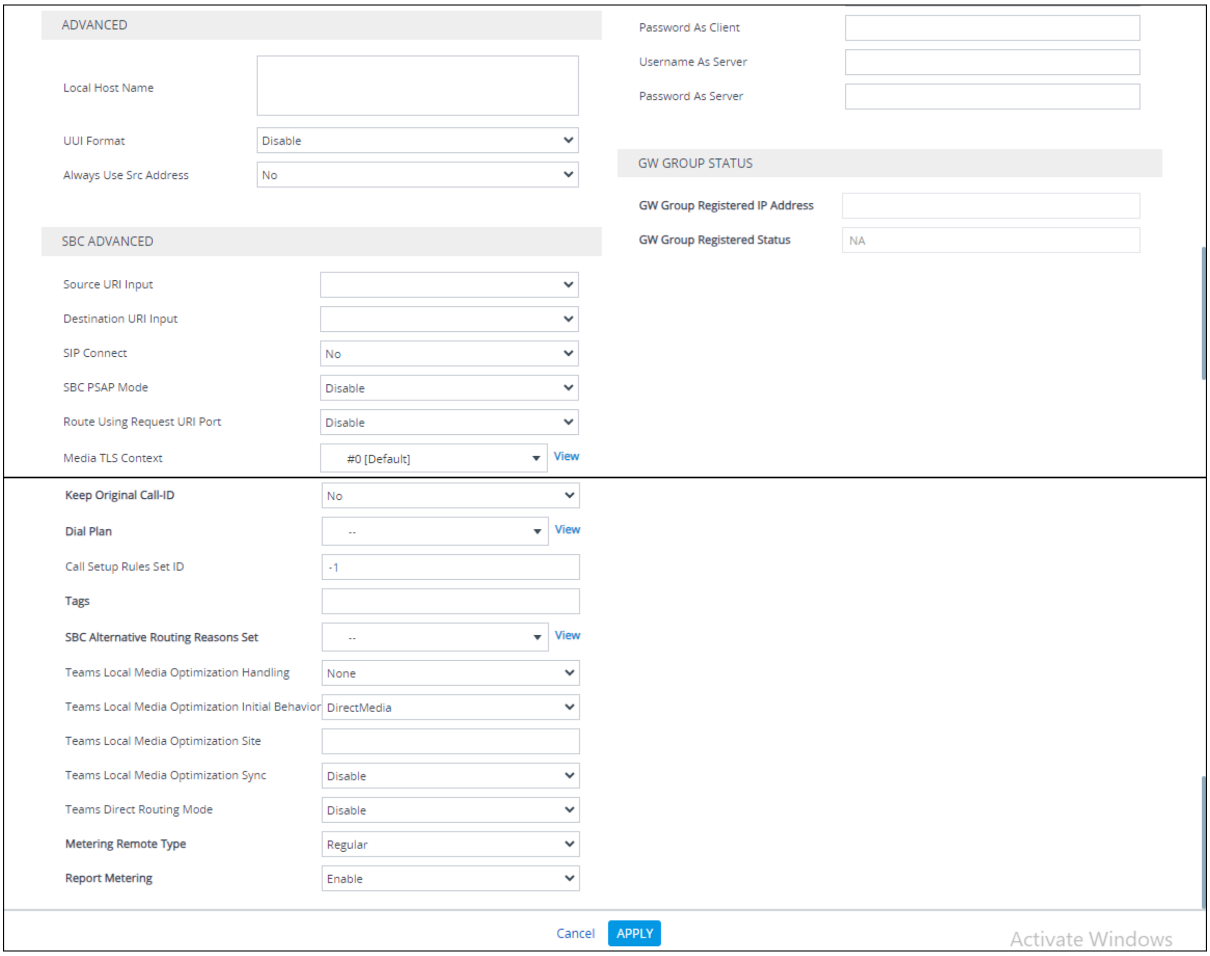

Figure 46: IP Group PSTN Continuation

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: gdnblgxxxxxxxxxxxxxxx.voiceconnector.chime.aws (FQDN of Amazon Chime SDK Voice Connector Trunk)

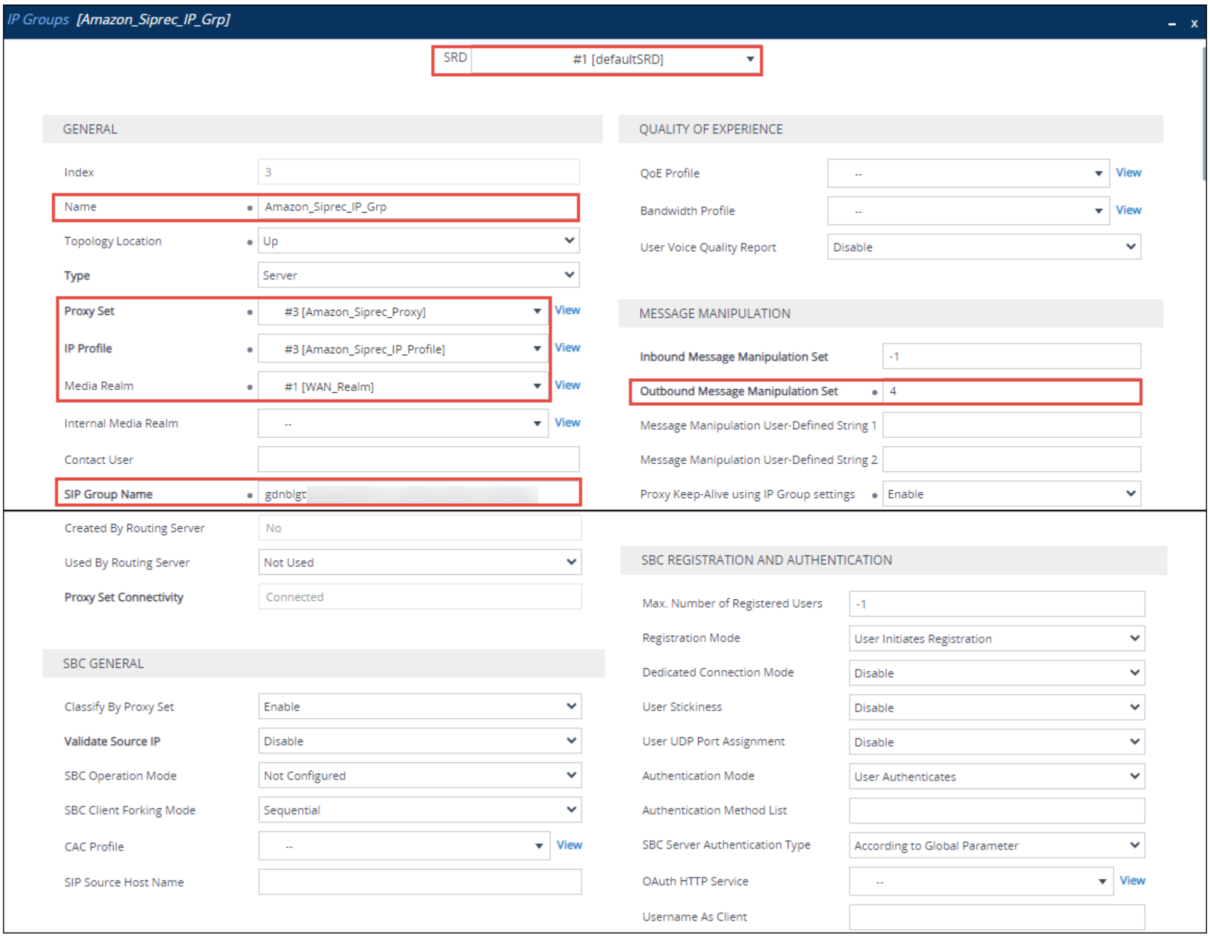

Figure 47: IP Group Amazon Siprec

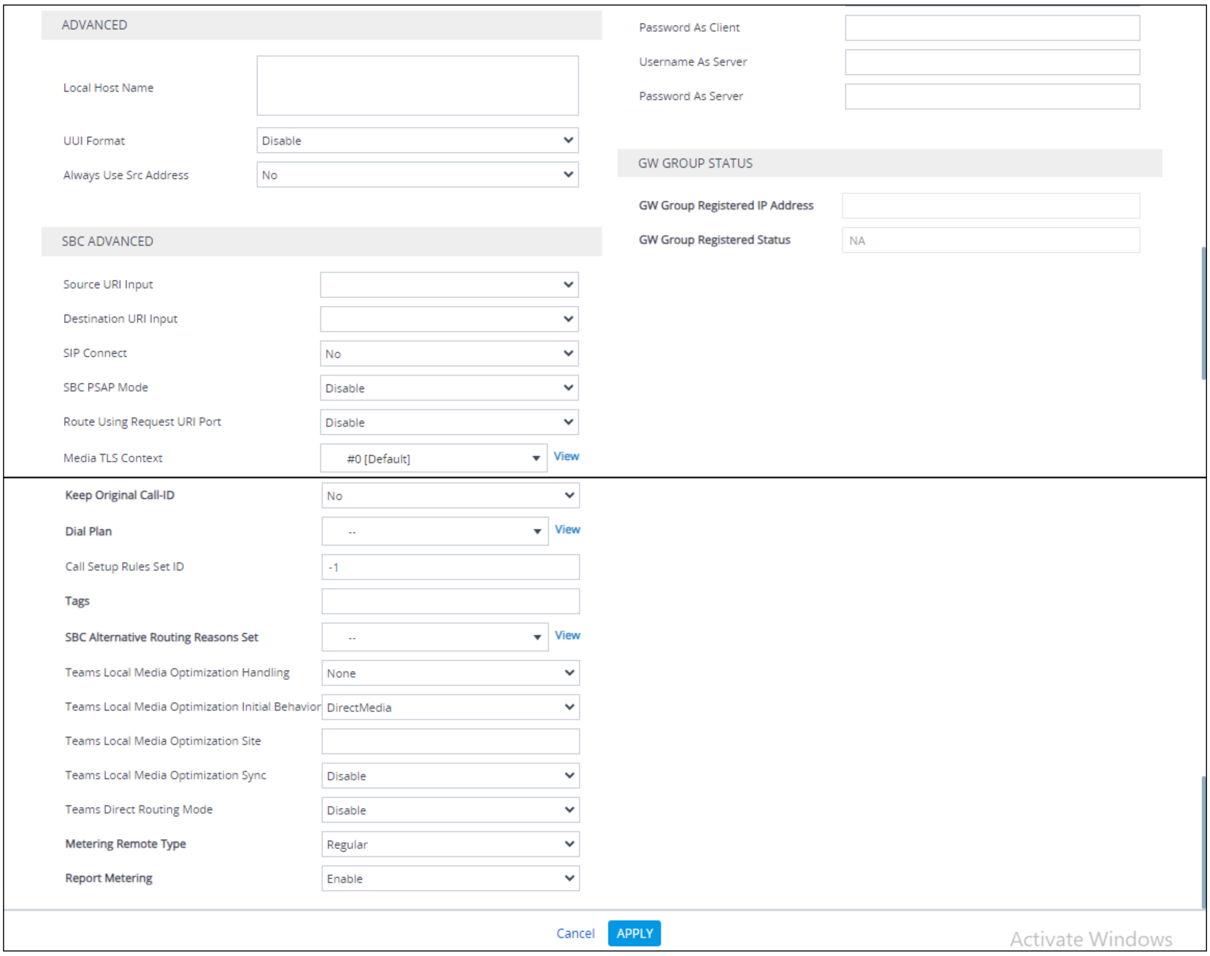

Figure 48: IP Group Amazon SIPREC Continuation

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**

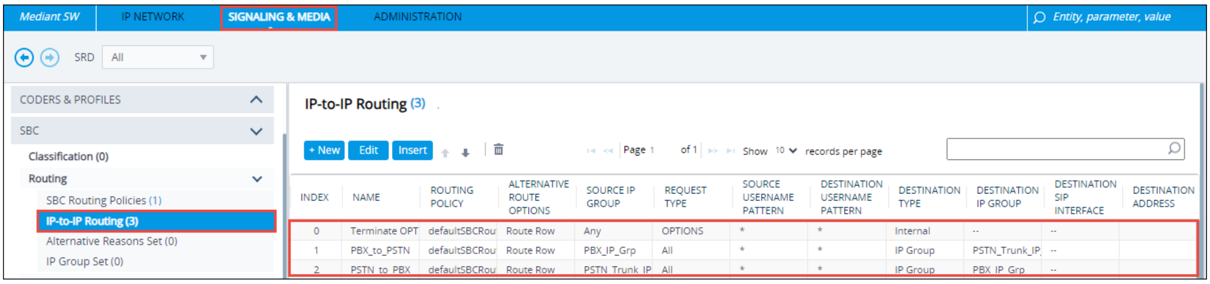

Figure 49: IP-to-IP Routing List

4.3.10.1 Terminate Options

Source IP Group: Any

Request Type: OPTIONS

Destination Type: Internal

Internal Action: Reply(response='200')

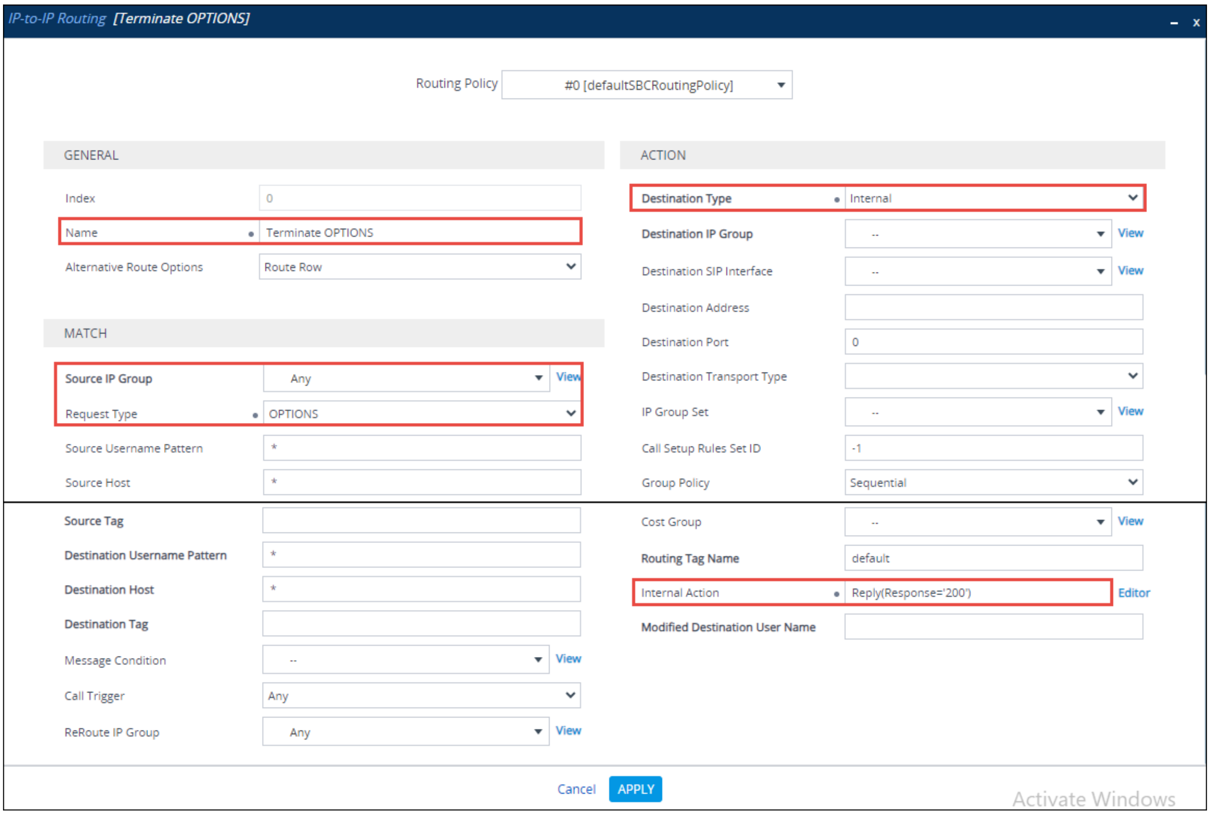

Figure 50: Terminate OPTIONS

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

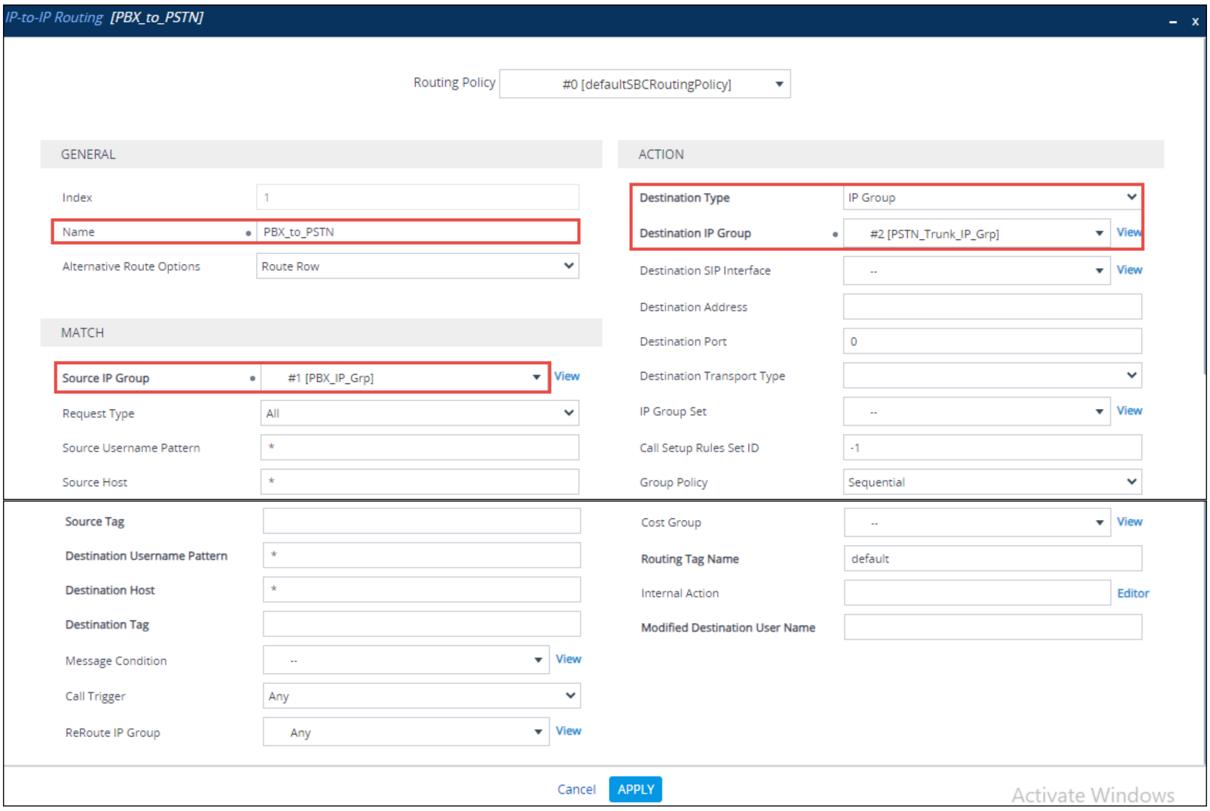

Figure 51: Routing PBX to PSTN

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

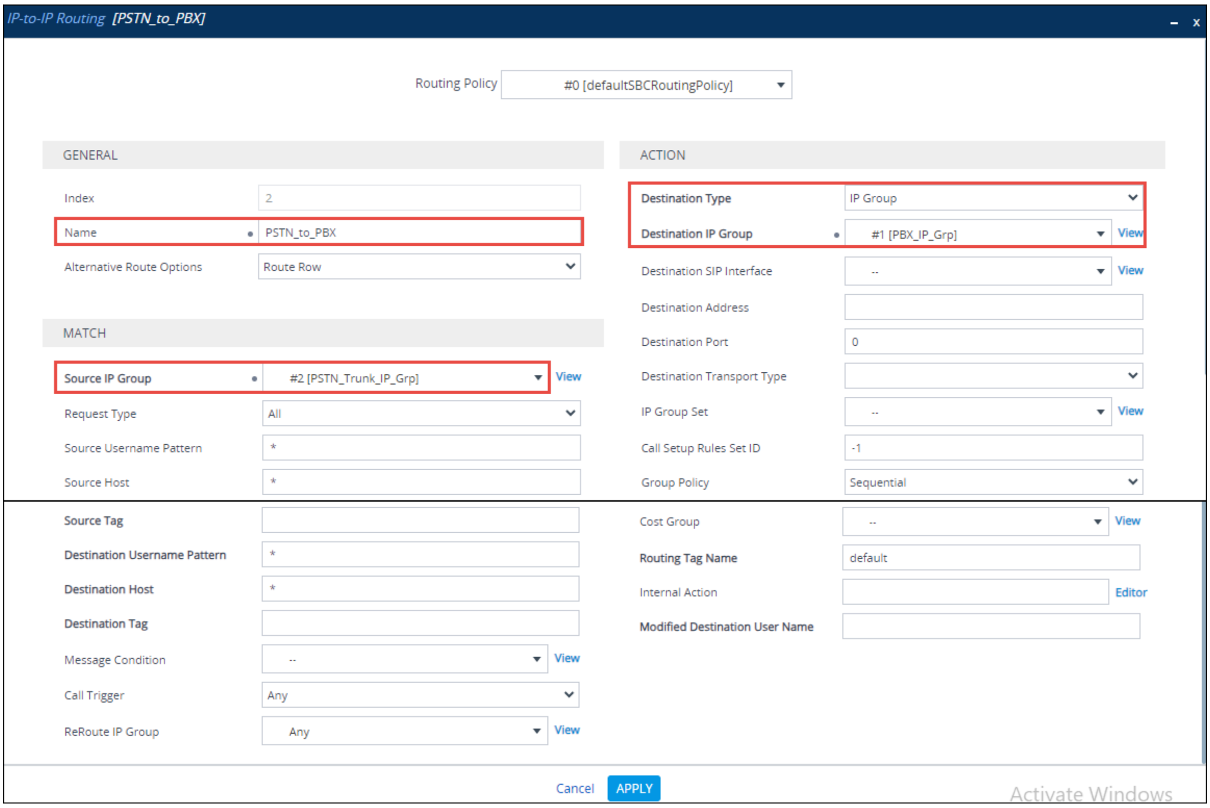

Figure 52: Routing PSTN to PBX

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: gdnblgxxxxxxxxxxxxxxx. voice connector.aws (FQDN of Amazon Chime SDK Voice Connector Trunk)

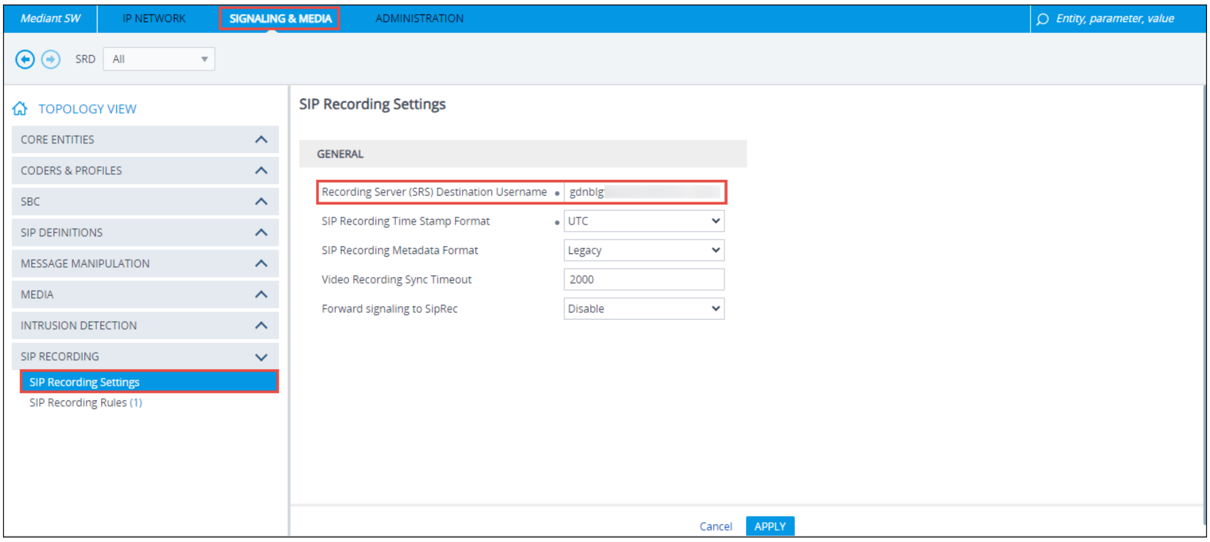

Figure 53: Amazon SIP Recording Settings

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

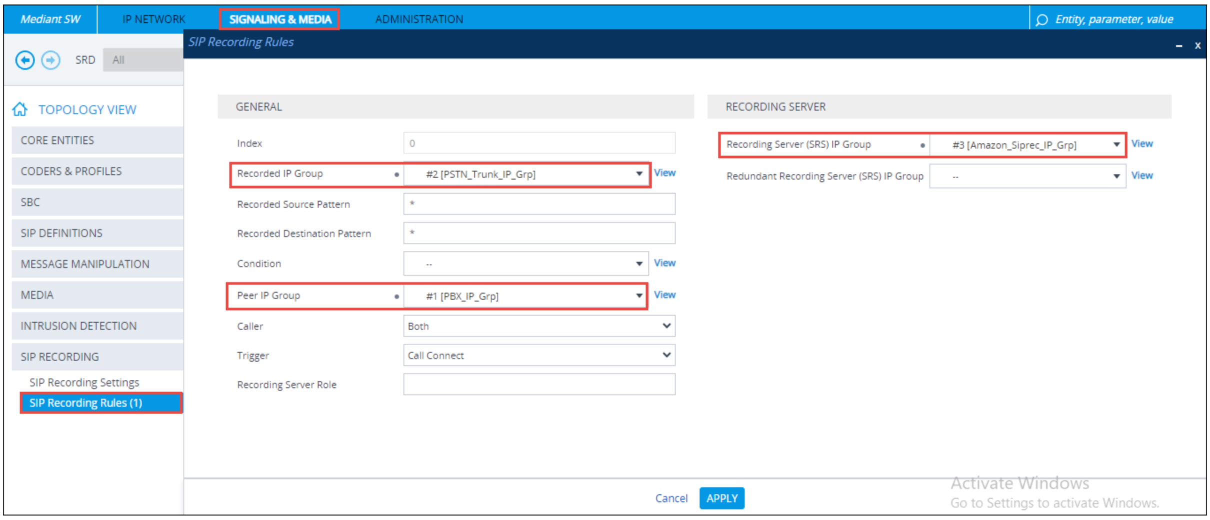

Figure 54: SIP Recording Rules

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

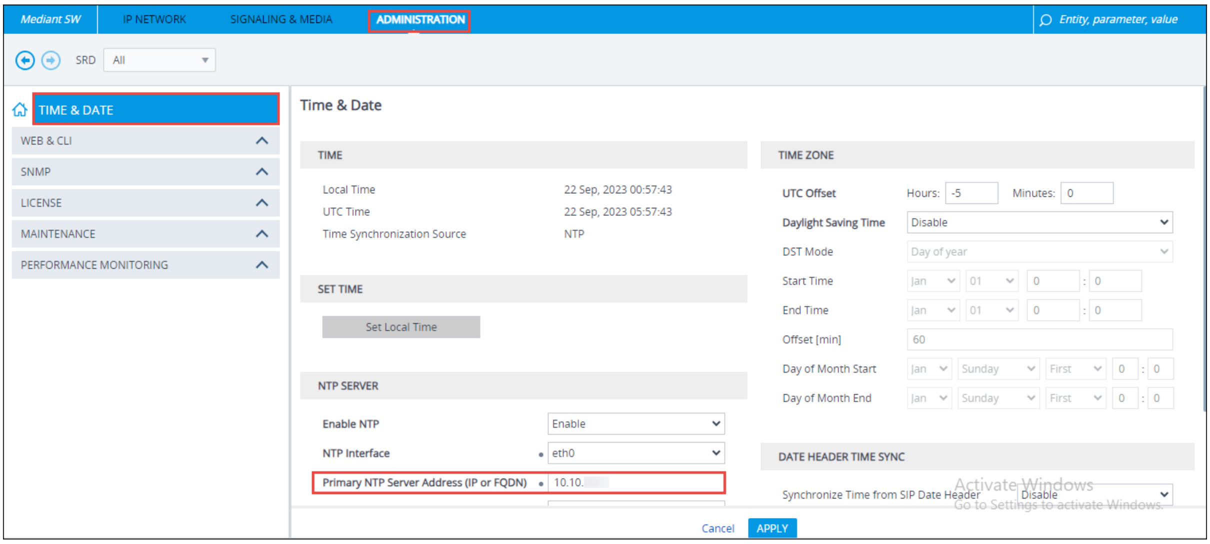

Figure 55: NTP Server

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

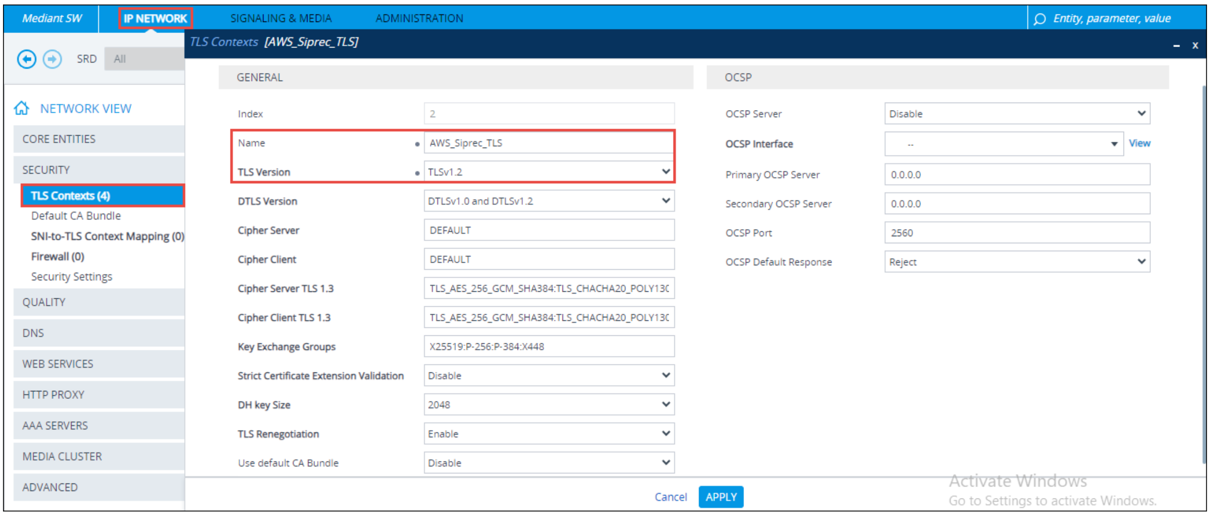

Figure 56: TLS Context

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.

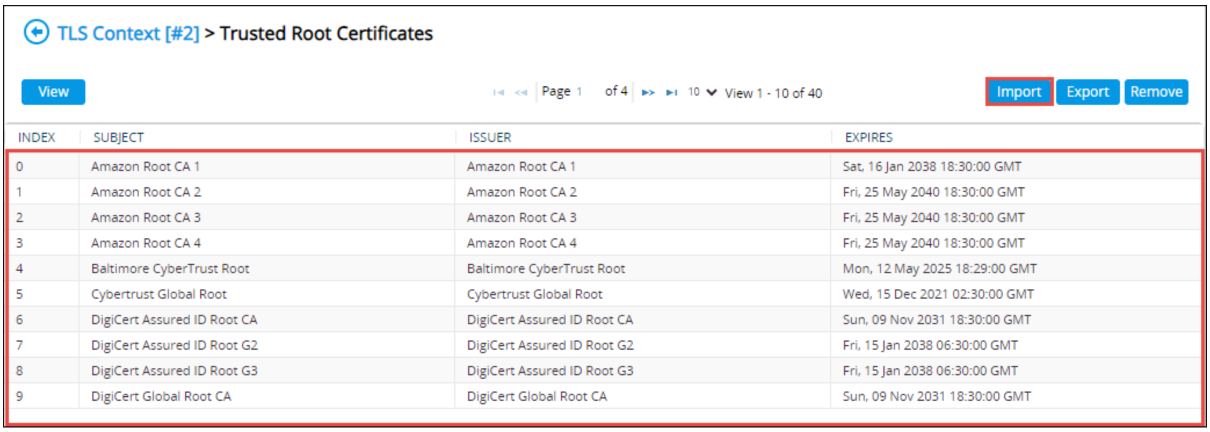

Figure 57: Amazon Trusted Root Certificates

4.3.12.4 SRTP

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

Enable the **Media Security**

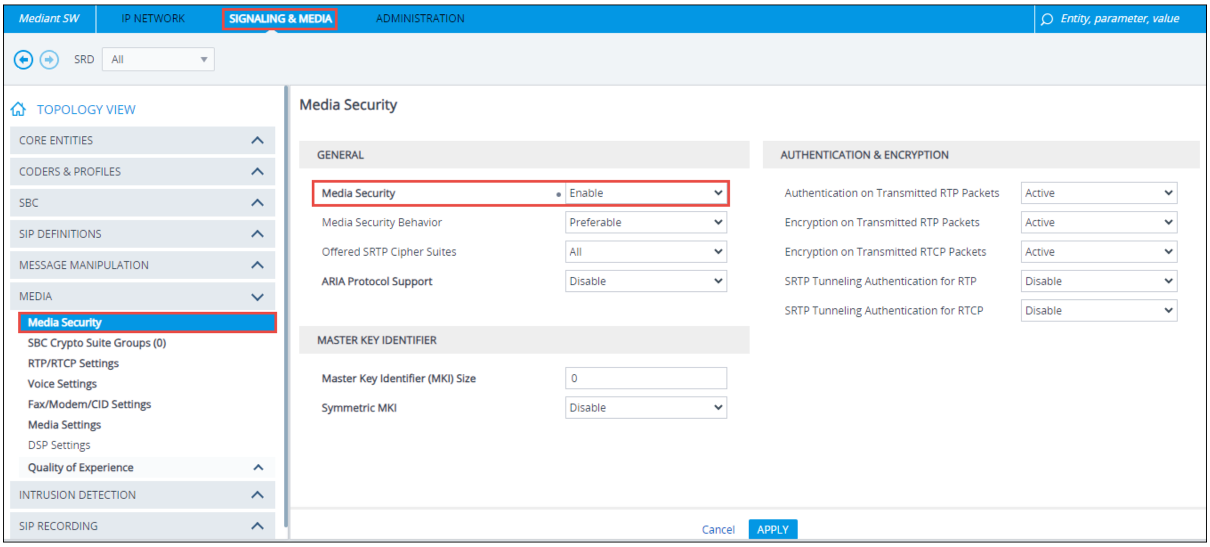

Figure 58: Media Security

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

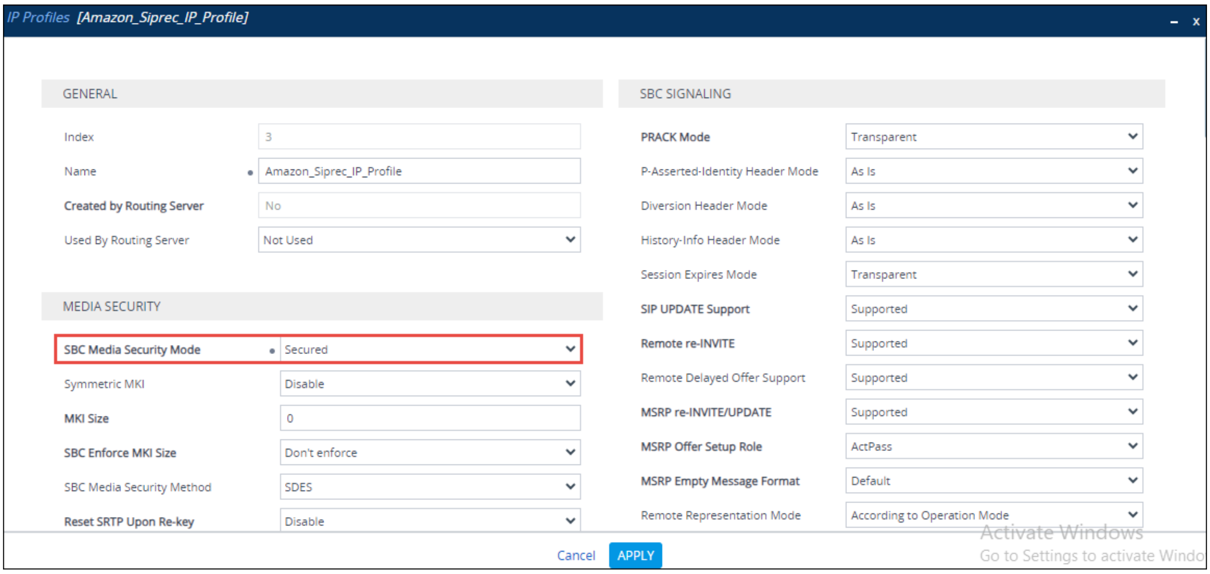

Figure 59: IP Profile TLS

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

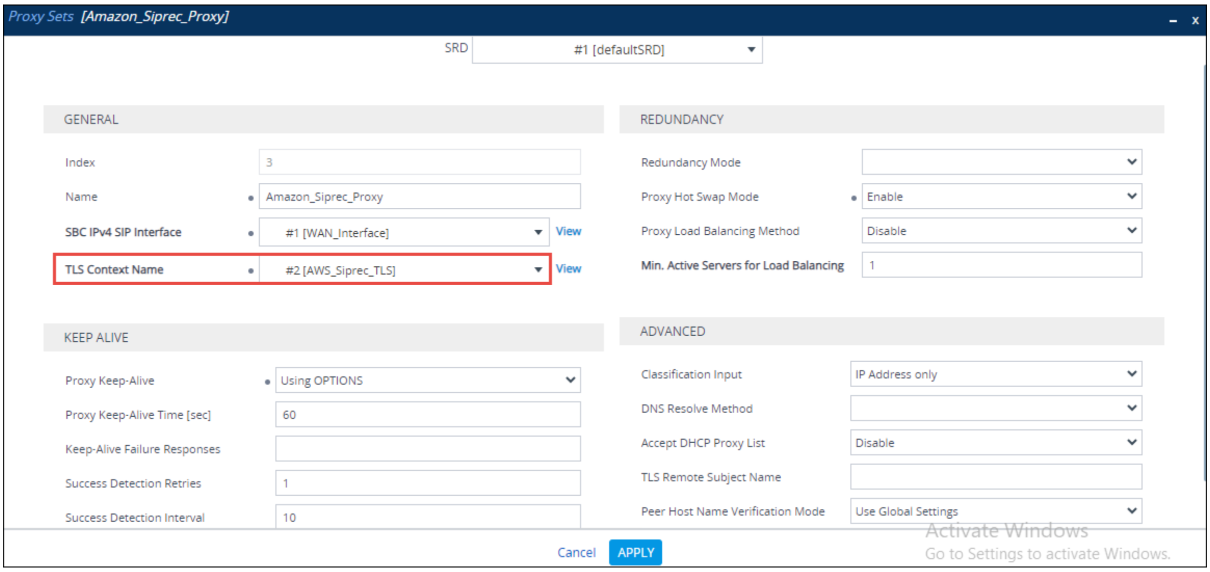

Figure 60: Proxy Set TLS

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: gdnblgxxxxxxxxxxxxxxxx.voiceconnector.chime.aws:5061

Transport Type: TLS

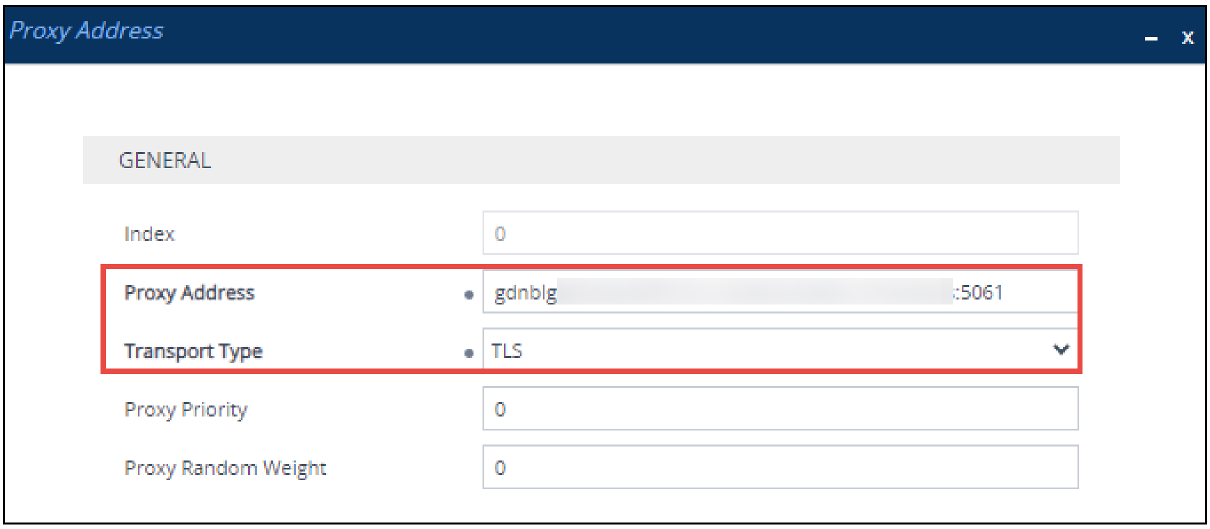

Figure 61: Proxy Address TLS

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**

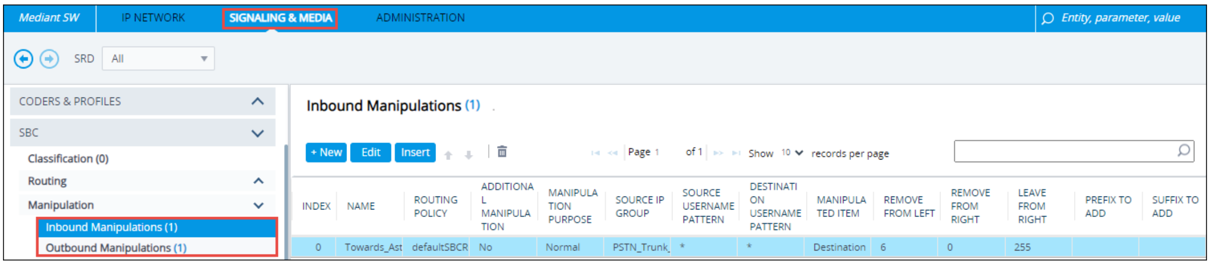

Figure 62: Number Manipulation

4.3.13.1 Inbound Manipulation

Name: Towards PBX

Source IP Group: PSTN Trunk IP Grp

Manipulated Item: Destination

Remove From Left: 6

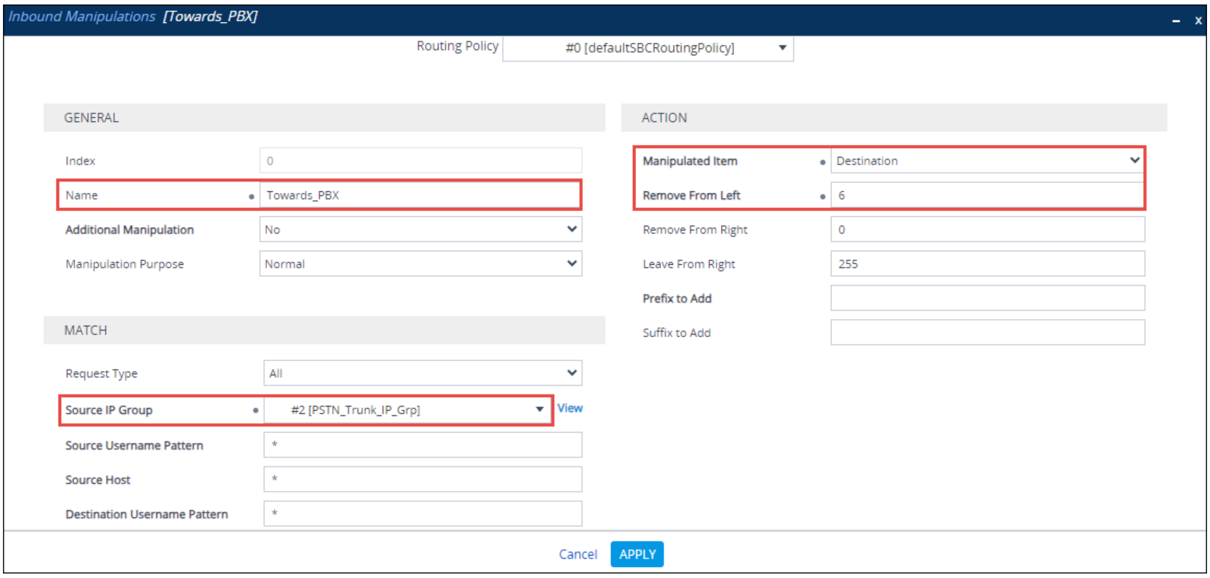

Figure 63: Inbound Manipulation PBX

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

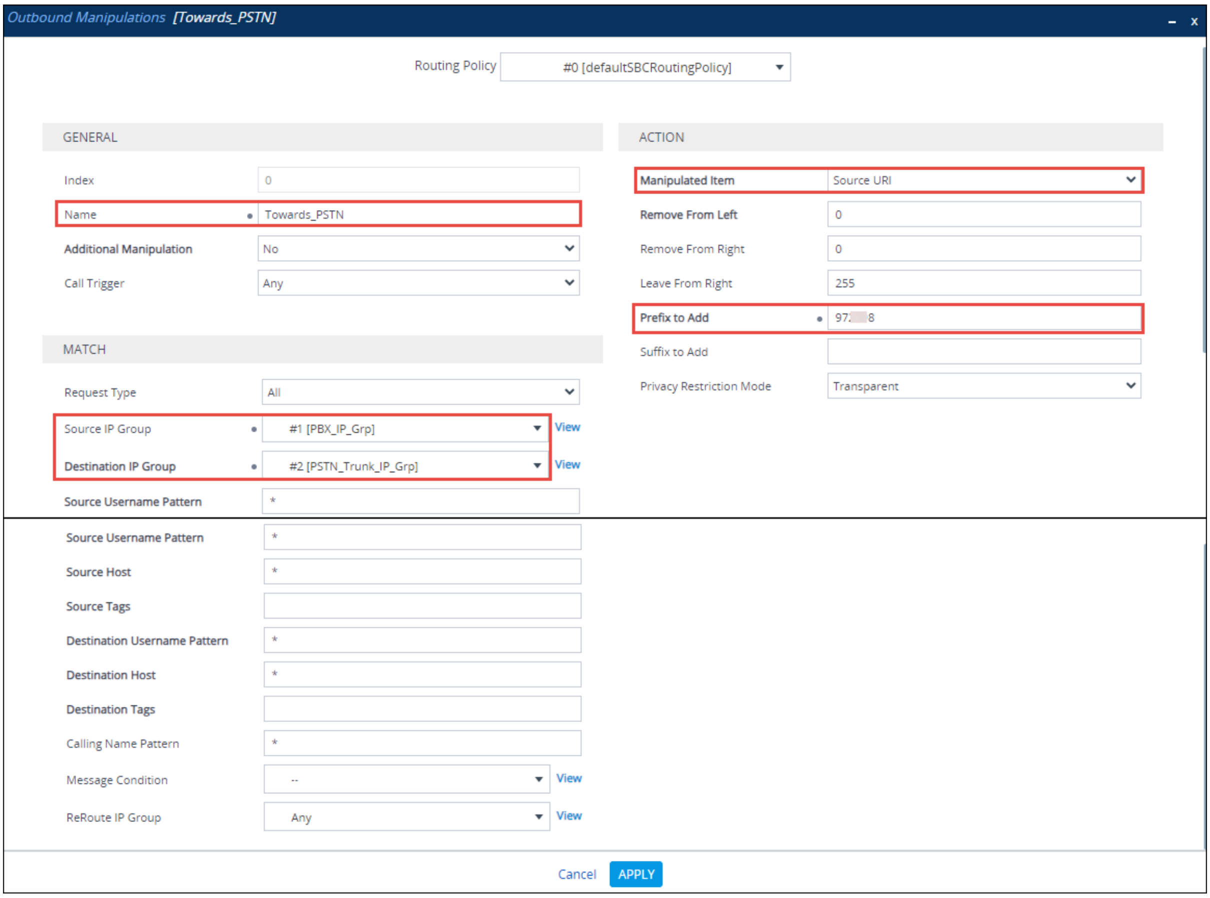

Figure 64: Outbound Manipulation PSTN

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

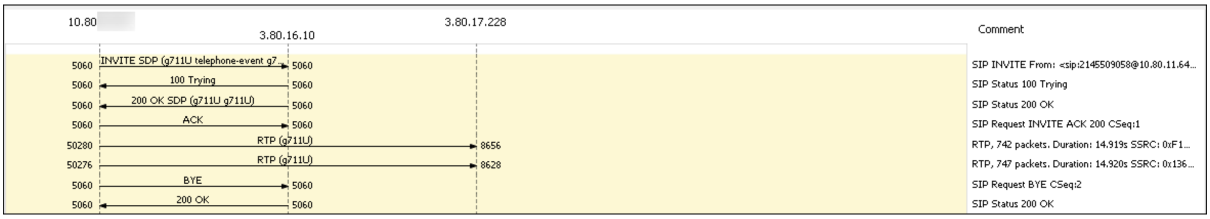

Figure 65: SIPREC sample trace

```
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: 949877094139202380 55@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,UPDA
TE
    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: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
  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:214XXXXXXX@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@gdnblgttwnpzuvf4h1hj1.voiceconnector.chime.aws;user=phone>;tag=Z2p021K
4aZN5B
Call-ID: 949877094139202380 55@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@gdnblgttwnpzuvf4h1hj1.voiceconnector.chime.aws;user=phone>;tag=Z2p021K
4aZN5B
   Call-ID: 949877094139202380550@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@gdnblgttwnpzuvf4h1hj1.voiceconnector.chime.aws;user=phone>;tag=Z2p021K
4aZN5B
   Call-ID: 949877094139202380550@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	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	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User

			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 5) Streaming and recording end when either PBX or PSTN user hangs up		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	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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User Recording 2: PBX User to PSTN User + Music on Hold 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.
4	Outbound hold and resume	PBX User calls external PSTN number. After call	1) Call is connected	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	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 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		Recording 1: PSTN User to PBX User Recording 2: PBX User audio to PSTN User + Music on Hold 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	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 between PSTN and PBX User-2 is captured after transfer 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	Passed	Two call recordings are available in AWS S3 Recording 1: PSTN User to PBX User-1 and PBX User-2 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 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.

			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 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	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 between PSTN and PBX User-2 is captured after transfer 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 transfer completes	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2 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 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.

			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 or PBX User-2 hangs up		
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	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	Passed	Four call recordings are available in AWS S3. 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 Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2 Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1 Recording 4: PBX User-1's audio to PSTN user 2 There is no re-invite from PBX during transfer. Hence, the recording is not paused and the music on hold is captured.

			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		
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	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 between PSTN and PBX User-2 is captured after transfer 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 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	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2 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 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.

			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	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 between PSTN and PBX User-2 is captured after transfer 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 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 or PBX User-2 hangs up	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2 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 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.

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	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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2 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 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.

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	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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2 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 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.

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	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	Four call recordings are available in AWS S3. 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 Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2 Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1 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 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.

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	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	Four call recordings are available in AWS S3. 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 Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2 Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1 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 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.

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	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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 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 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.
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	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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2

			4) RTP is captured when queue accepts call 5) RTP between PSTN and PBX User-2 is captured 6) Inbound caller number and PBX User-1 extension number are captured in the metadata 7) Queue number is captured in the metadata after call transfer 8) There is one call recording per call leg for the duration of the call 9) The timestamps for the recording show accurate start and end times 10) Streaming and recording end when PSTN hangs up		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 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.
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	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	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 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 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) 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		
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	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	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 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 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.

			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		
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	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) RTP between PSTN, PBX User-1, PBX User-2, and PBX User-3 is captured after PBX User-3 is added to the call 6) Inbound caller number and PBX User-1 extension number are captured in the metadata 7) PBX User-2 extension number is added to the metadata after conference starts	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1, PBX User-2, and PBX User-3 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 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.

			8) PBX User-3 extension number is added to the metadata after addition to conference 9) There is one call recording per call leg for the duration of the call 10) The timestamps for the recording show accurate call duration for the entire call 11) Streaming and recording end when either PSTN hangs up or last participant from PBX User-1 and User-2 hangs up		
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	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	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2 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 Meta data information only has PBX User-2's extension and PSTN User number.

			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		
20	Emergency calling	PBX User-1 calls the 511 service	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) 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 511 user hangs up	Passed	Two call recordings are available in AWS S3. Recording 1: 511 User to PBX User Recording 2: PBX User to 511 User Note: - This scenario is locally simulated within Lab environment.
21	Outbound international call	Outbound call from PBX User-1 to international PSTN number	1) Call is connected 2) RTP between PBX Users and PSTN is captured 3) PBX extension number and outbound caller number are captured in the metadata (caller ID capture to be tested)	Passed	Two call recordings are available in AWS S3. Recording 1: International PSTN User to PBX User Recording 2: PBX User to International PSTN User

			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		

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	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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User Recording 2: PBX User to PSTN User + Music on Hold 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.
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	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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 Recording 2: PBX User-1's audio to PSTN User + Music on Hold

		User or PSTN hangs up	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		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	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 between PSTN and PBX User-2 is captured after transfer 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 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	Passed	Two call recordings are available in AWS S3 Recording 1: PSTN User to PBX User-1 and PBX User-2 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 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.

			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	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 between PSTN and PBX User-2 is captured after transfer 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 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 or PBX User-2 hangs up	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2 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 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.

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	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	Four call recordings are available in AWS S3. 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 Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2 Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1 Recording 4: PBX User-1's audio to PSTN user 2 There is no re-invite from PBX during transfer. Hence, the recording is not paused and the music on hold is captured.
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	1) Call is connected 2) RTP between PSTN and PBX User-1 is captured	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2

			3) RTP is not captured between PSTN and PBX User-1 during transfer 4) RTP between PSTN and PBX User-2 is captured after transfer 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 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 or PBX User-2 hangs up		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 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.
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	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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2

			4) RTP between PSTN and PBX User-2 is captured after transfer 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 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 or PBX User-2 hangs up		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 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.
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	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	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2 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

		call participants hangs up	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		User and PBX User-2 + PBX User-2's audio to PSTN User and PBX User-1 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.
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	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	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2 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

		call participants hangs up	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		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.
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	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	Passed	Four call recordings are available in AWS S3. 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 Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2

		participants hangs up	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		Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1 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 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.
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	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.	Passed	Four call recordings are available in AWS S3. 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 Recording 2: PSTN User-1's audio to PBX User-1 and PSTN User-2

		participants hangs up	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		Recording 3: PSTN User-2's audio to PBX User-1 and PSTN User-1 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 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.
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	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	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 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

			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) One recording for the entire call duration is stored 8) The timestamps for the recording show accurate start and end times 9) Streaming and recording end when PSTN hangs up		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.
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	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) RTP between PSTN and PBX User-2 is captured 6) Inbound caller number and PBX User-1 extension number are captured in the metadata 7) Queue number is captured in the metadata after call transfer	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2 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 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.

			8) There is one call recording per call leg for the duration of the call 9) The timestamps for the recording show accurate start and end times 10) Streaming and recording end when PSTN hangs up		
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	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) One recording for the entire call duration is stored 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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 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 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.

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	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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 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 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.
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	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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1, PBX User-2, and PBX User-3 Recording 2: PBX User-1's audio with PSTN User + Music On Hold from PBX User-1 while consulting

		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	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) RTP between PSTN, PBX User-1, PBX User-2, and PBX User-3 is captured after PBX User-3 is added to the call 6) Inbound caller number and PBX User-1 extension number are captured in the metadata 7) PBX User-2 extension number is added to the metadata after conference starts 8) PBX User-3 extension number is added to the metadata after addition to conference 9) There is one call recording per call leg for the duration of the call 10) The timestamps for the recording show accurate call duration for the entire call 11) Streaming and recording end when either PSTN hangs up or last participant from PBX User-1 and User-2 hangs up		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 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.

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	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	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2 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 Meta data information only has PBX User-2's extension and PSTN User number.
20	Emergency calling	PBX User-1 calls the 511 service	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	Two call recordings are available in AWS S3. Recording 1: 511 User to PBX User Recording 2: PBX User to 511 User

			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 511 user hangs up		Note: - This scenario is locally simulated within Lab environment.
21	Outbound international call	Outbound call from PBX User-1 to international PSTN number	1) Call is connected 2) RTP between PBX Users and PSTN is captured 3) PBX extension number and outbound caller number are captured in the metadata (caller ID capture to be tested) 4) There is one call recording 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	Passed	Two call recordings are available in AWS S3. Recording 1: International PSTN User to PBX User Recording 2: PBX User to International PSTN User