Amazon Chime SDK Voice Connector

SIPREC Configuration Guide

FreePBX 16.0.40.4 Asterisk 20.1.0 and Ribbon SBC 5210 v10.01.04-R001

September 2023

Document History

Rev. No.	Date	Description
1.0	August-22-2023	SIPREC Configuration Guide
1.1	September-13-2023	Incorporated the changes based on comments from AWS.

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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 **Ribbon 5210 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 (ESBC). 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 SIPREC reference configuration is illustrated below:

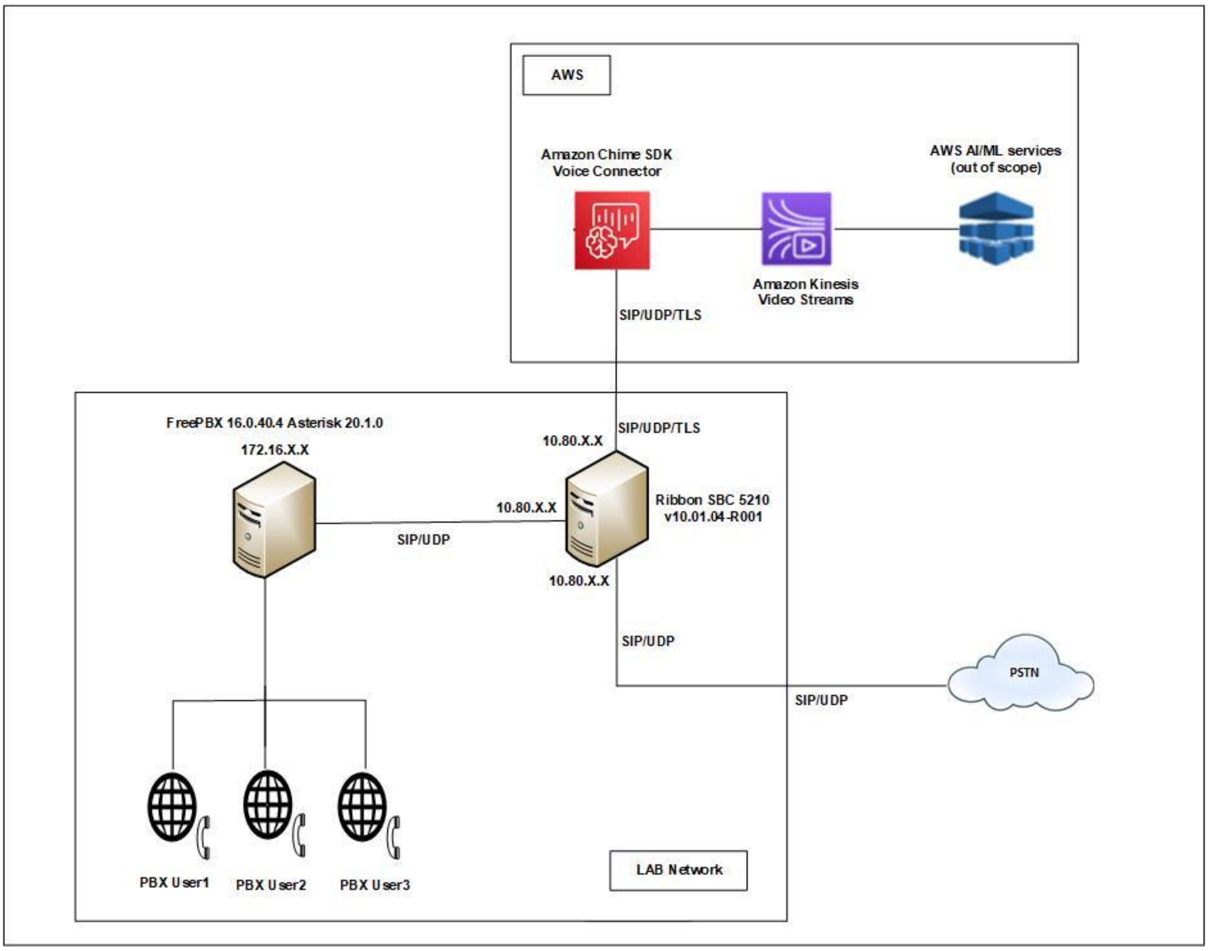


Figure 1: Network Topology

The signaling and media flow is illustrated below:

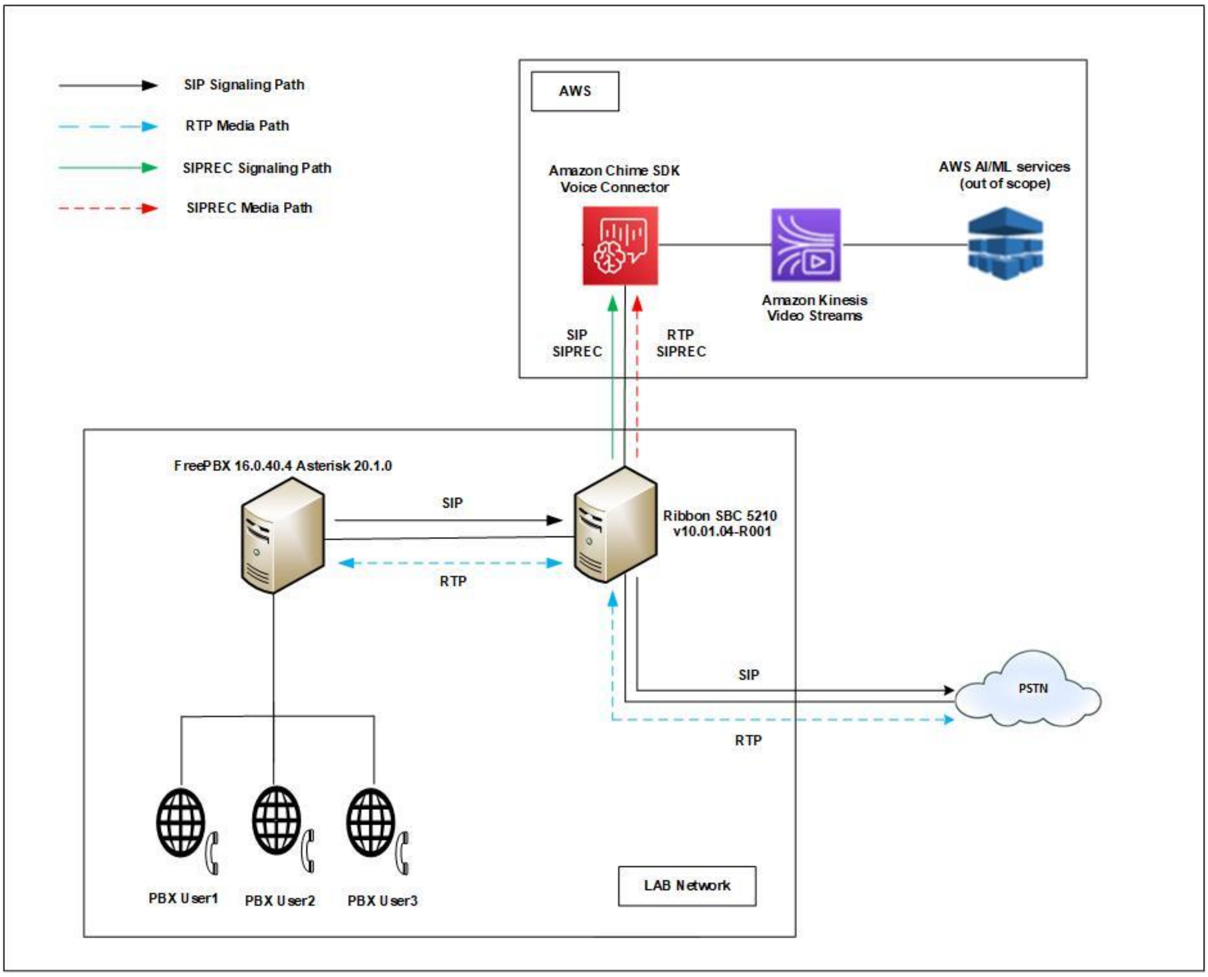


Figure 2: Signaling and Media Flow

2.1 Hardware Components

- VMWare server running ESXi 7.0 or later used for the following virtual machine
    - Asterisk FreePBX
- Ribbon SBC 5210
- Polycom IP Phone(s)
    - VVX 150
    - VVX 201
    - SoundPoint IP 650

2.2 Software Requirements

- FreePBX 16.0.40.4 Asterisk 20.1.0
- Ribbon SBC 5210 v10.01.04-R001

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

- Early media and audio from endpoints are recorded by the SBC for basic Inbound and Outbound calls before call establishment.

- 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.

- Mid call signaling is not observed from PBX for internal Transfer and internal Conference scenarios. Therefore, meta data is not updated for the new parties joined in the call.

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 Ribbon SBC 5210 for SIPREC using SIP Trunking with Amazon Chime SDK Voice Connector.

Table 2 – PBX and ESBC Configuration Steps

Steps	Description	Reference
Step 1	FreePBX Asterisk Configuration	Section 4.2
Step 2	Ribbon SBC 5210 Configuration	Section 4.3
Step 3	Amazon Chime Voice Connector Configuration	Amazon Chime Voice Connector
Step 4	Amazon Chime Kinesis Configuration	Amazon Chime Kinesis Configuration

4.2 FreePBX Asterisk Configuration

The following configuration with screen shots taken from the FreePBX Asterisk system are used to integrate with Ribbon SBC 5210 and it can be customised by the administrators based on their enterprise specifications and requirements.

4.2.1 FreePBX Asterisk Version

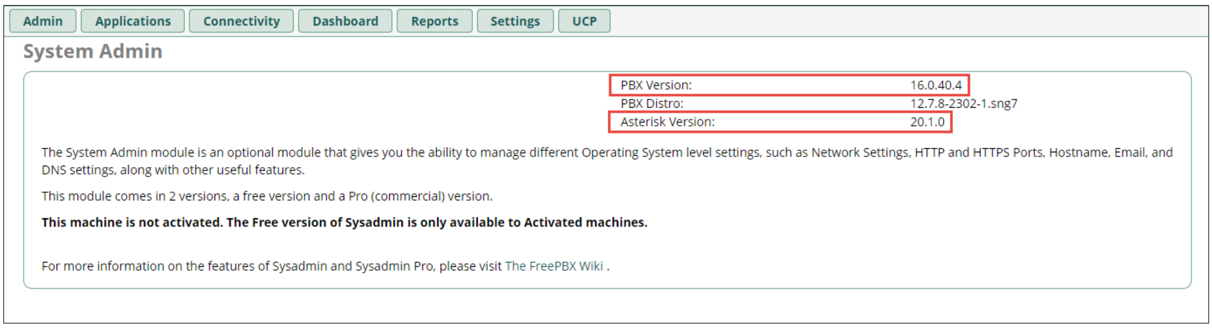


Figure 3: FreePBX Asterisk Version

4.2.2 Extensions

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

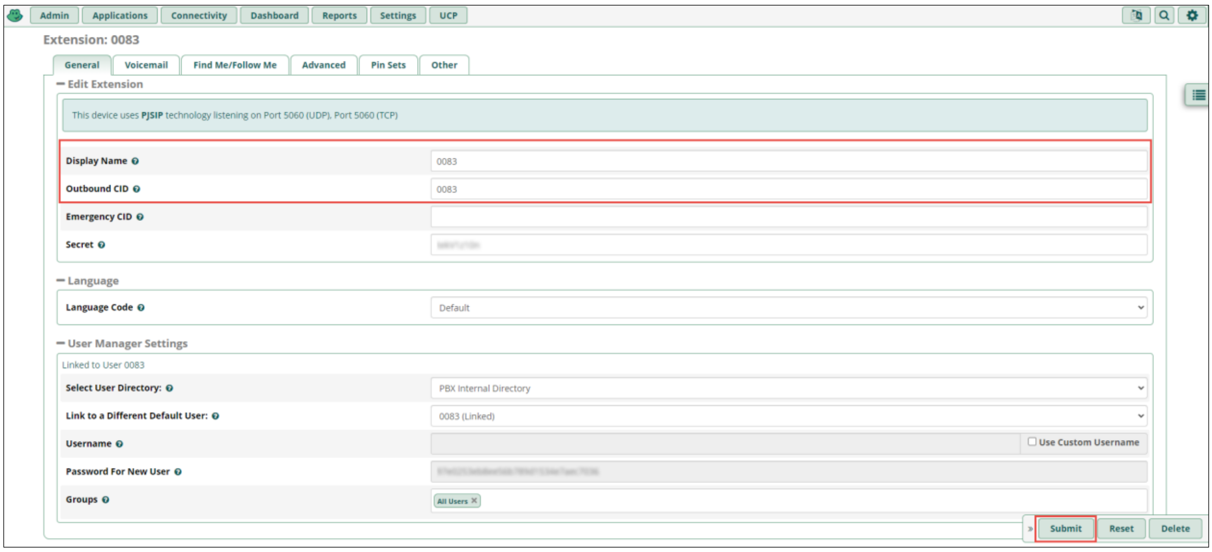


Figure 4: Asterisk Extension

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

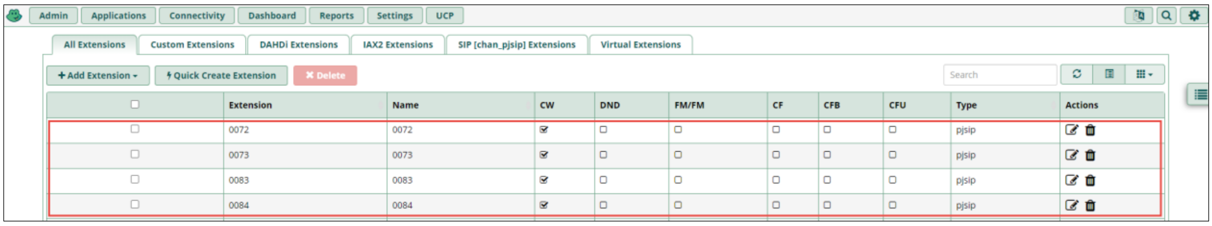


Figure 5: Asterisk Extensions List

4.2.3 Trunk

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

Trunk Name: Enter a name for the Trunk

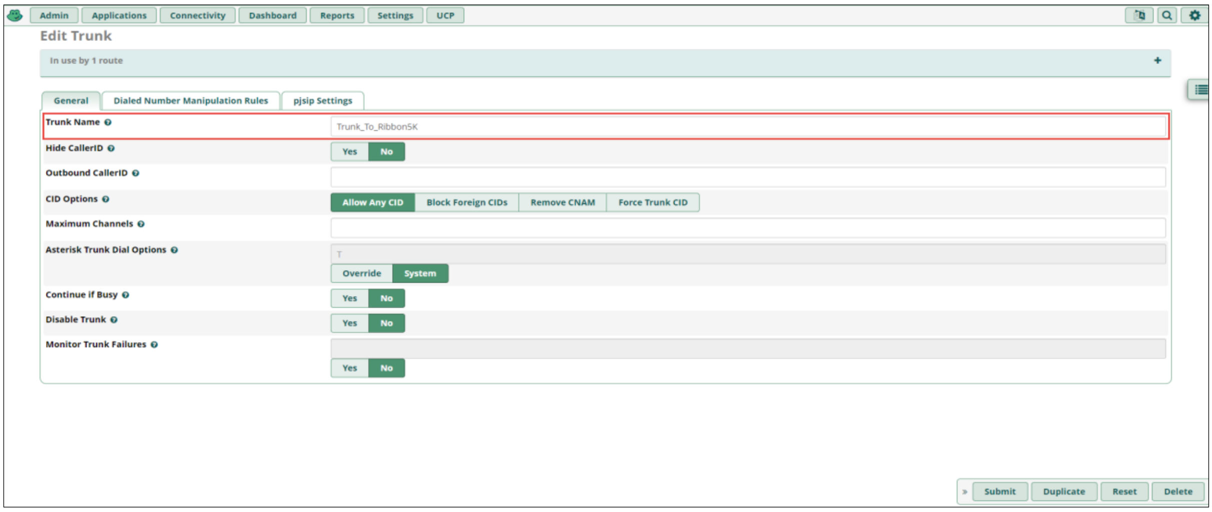


Figure 6: Asterisk Trunk

Navigate to **Pjsip settings → General**

SIP Server: 10.80.X.X (IP of Ribbon 5210 SBC's Network Interface towards the FreePBX Asterisk)

SIP Server Port: 5060

Transport: 0.0.0.0-udp

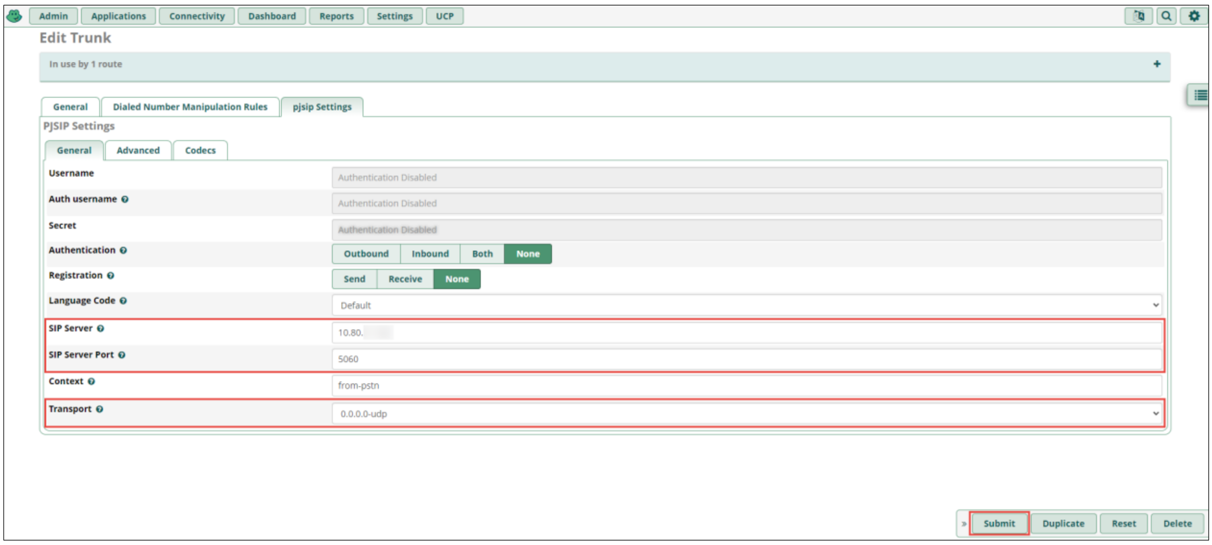


Figure 7: Asterisk Trunk Continuation

Navigate **to Pjsip settings → Codecs**

Enable Ulaw

Click Submit

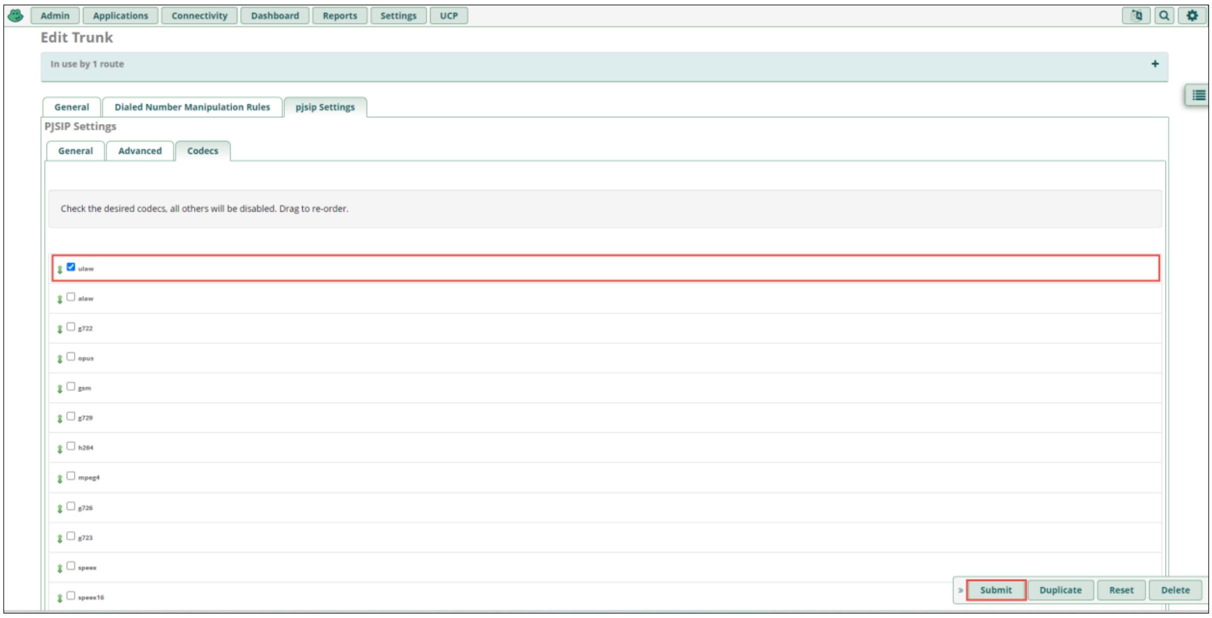


Figure 8: Asterisk Trunk Continuation

4.2.4 Outbound Route

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

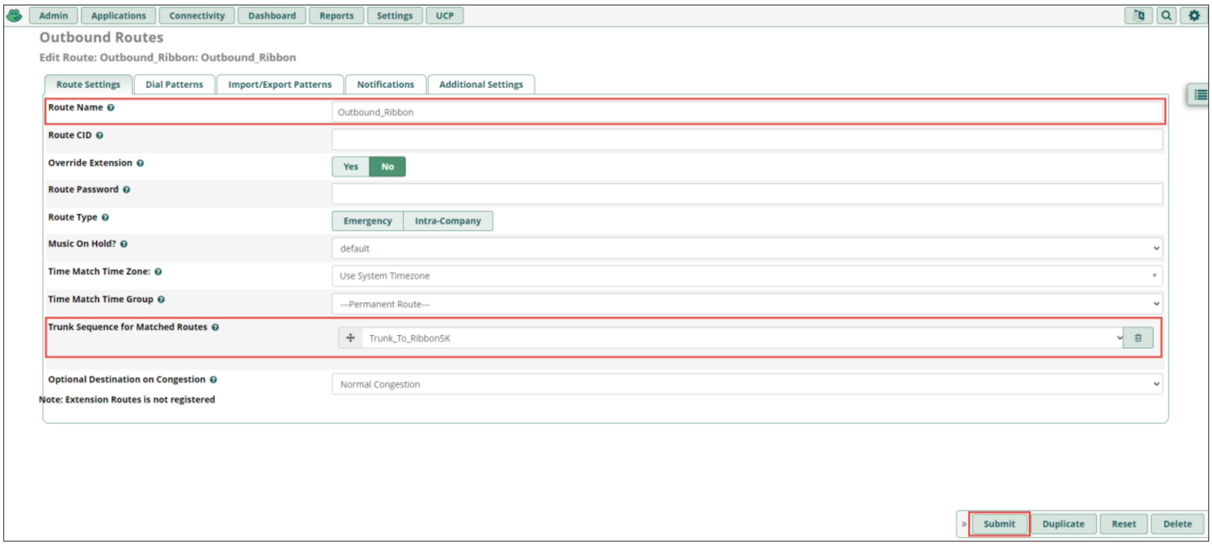


Figure 9: Asterisk Outbound Route

Navigate to **Dial Patterns** and add below patterns. These routing pattern definitions shall be customized based on enterprise requirements.

For PSTN dialing

Prefix: 8

Match Pattern: 214XXXXXXX

For International dialing

Match Pattern: 01191XXXXXXXXXX

For Short code dialing

Prefix: 9

Match Pattern: 411

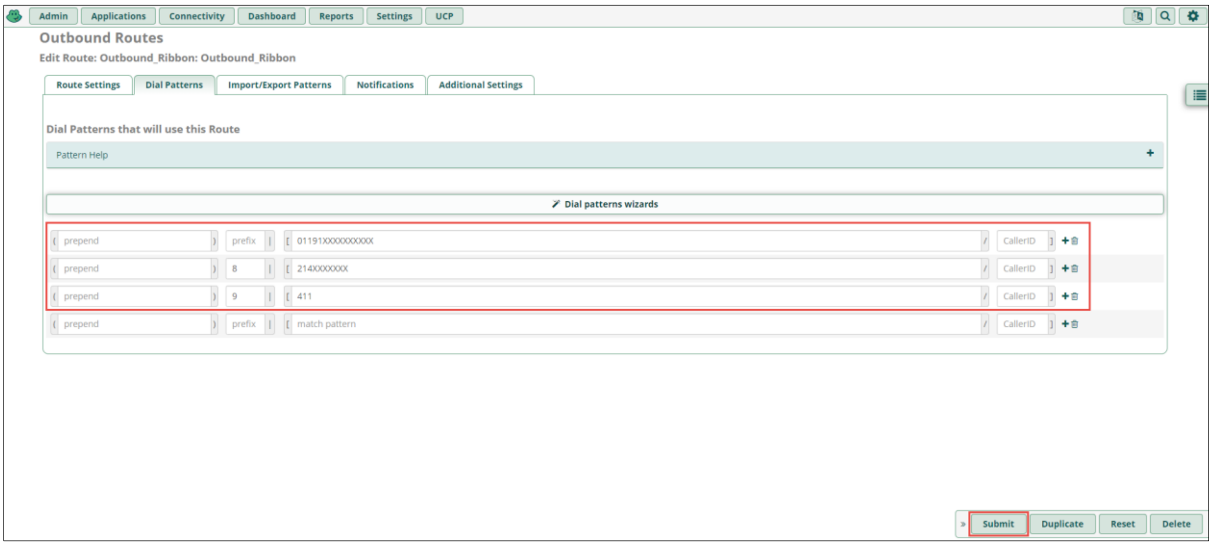


Figure 10: Asterisk Outbound Route Continuation

4.3 Ribbon SBC 5210 Configuration

This section provides a general overview of the configuration along with SIPREC based configuration with Amazon Chime SDK Voice Connector that needs to be performed in Ribbon SBC 5210

4.3.1 Login to Ribbon SBC 5210

- Log into Ribbon SBC 5210 in putty SSH through its Management IP Address
- Enter the "Admin" Username and Password
- To enter configuration mode, enter "configure"
- Configuration is performed in Sonus SBC using the commands listed in the various section below,

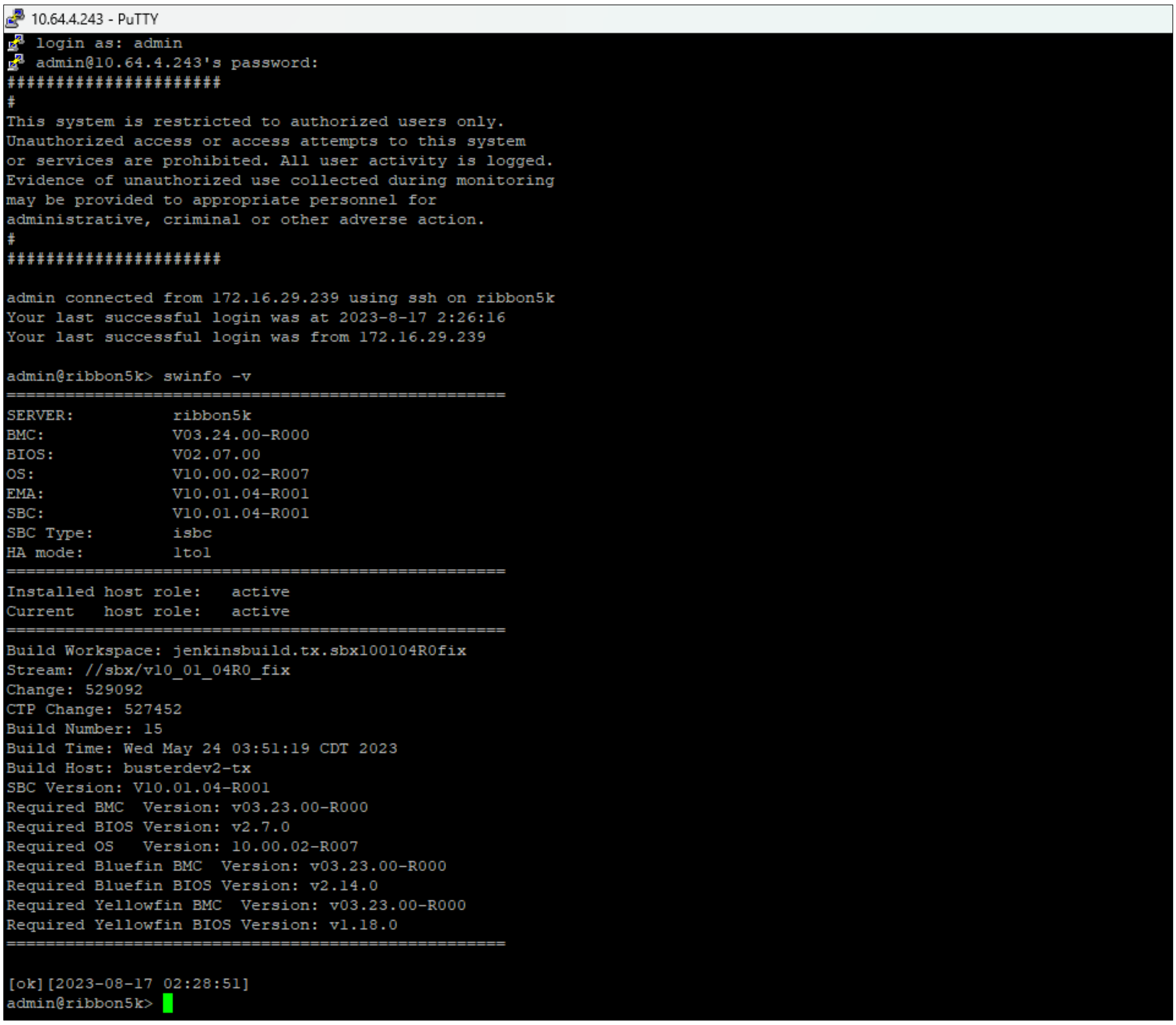


Figure 11: Ribbon SBC Login

4.3.2 Interface Group

Ribbon LAN:

set addressContext default ipInterfaceGroup LAN_IPIG ipInterface LAN_IPI portName pkt0
set addressContext default ipInterfaceGroup LAN_IPIG ipInterface LAN_IPI ipAddress <Interface IP>
set addressContext default ipInterfaceGroup LAN_IPIG ipInterface LAN_IPI prefix 24
set addressContext default ipInterfaceGroup LAN_IPIG ipInterface LAN_IPI mode inService
set addressContext default ipInterfaceGroup LAN_IPIG ipInterface LAN_IPI state enabled
commit

Ribbon WAN:

set addressContext default ipInterfaceGroup WAN_IPIG ipInterface WAN_IPI portName pkt1
set addressContext default ipInterfaceGroup WAN_IPIG ipInterface WAN_IPI ipAddress <Interface IP>
set addressContext default ipInterfaceGroup WAN_IPIG ipInterface WAN_IPI prefix 24
set addressContext default ipInterfaceGroup WAN_IPIG ipInterface WAN_IPI mode inService
set addressContext default ipInterfaceGroup WAN_IPIG ipInterface WAN_IPI state enabled
commit

4.3.3 Zone

Asterisk FreePBX:

set addressContext default zone PBX id 2
commit

PSTN Gateway:

set addressContext default zone PSTN id 5
commit

Amazon Chime Voice Connector:

set addressContext default zone AWS id 4
commit

4.3.4 Sip Signaling port

Asterisk FreePBX:

set addressContext default zone PBX sipSigPort 1 ipInterfaceGroupName LAN_IPIG
set addressContext default zone PBX sipSigPort 1 ipAddressV4 < Interface IP>
set addressContext default zone PBX sipSigPort 1 portNumber 5060
set addressContext default zone PBX sipSigPort 1 mode inService
set addressContext default zone PBX sipSigPort 1 state enabled
set addressContext default zone PBX sipSigPort 1 transportProtocolsAllowed sip-udp
commit

PSTN Gateway:

set addressContext default zone PSTN sipSigPort 5 ipInterfaceGroupName WAN_IPIG
set addressContext default zone PSTN sipSigPort 5 ipAddressV4 <WAN Interface IP>
set addressContext default zone PSTN sipSigPort 5 portNumber 5060
set addressContext default zone PSTN sipSigPort 5 mode inService
set addressContext default zone PSTN sipSigPort 5 state enabled
set addressContext default zone PSTN sipSigPort 5 transportProtocolsAllowed sip-udp
commit

Amazon Chime Voice Connector:

set addressContext default zone AWS sipSigPort 3 ipInterfaceGroupName LAN_IPIG
set addressContext default zone AWS sipSigPort 3 ipAddressV4 <LAN Interface IP>
set addressContext default zone AWS sipSigPort 3 portNumber 5062
set addressContext default zone AWS sipSigPort 3 mode inService
set addressContext default zone AWS sipSigPort 3 state enabled
set addressContext default zone AWS sipSigPort 3 siprec enabled
set addressContext default zone AWS sipSigPort 3 transportProtocolsAllowed sip-udp
commit

4.3.5 SIP Trunk

Asterisk FreePBX:

set addressContext default zone PBX sipTrunkGroup PBX_TG media mediaIpInterfaceGroupName LAN_IPIG
set addressContext default zone PBX sipTrunkGroup PBX_TG state enabled
set addressContext default zone PBX sipTrunkGroup PBX_TG mode inService
set addressContext default zone PBX sipTrunkGroup PBX_TG policy digitParameterHandling numberingPlan NANP_ACCESS
set addressContext default zone PBX sipTrunkGroup PBX_TG policy digitParameterHandling egressDmPmRule Rule_Digit_to_Ext
set addressContext default zone PBX sipTrunkGroup PBX_TG policy callRouting elementRoutingPriority TG_ERP
set addressContext default zone PBX sipTrunkGroup PBX_TG ingressIpPrefix <IP address of PBX> 32
set addressContext default zone PBX sipTrunkGroup PBX_TG policy signaling ipSignalingProfile PBX_IPSP
set addressContext default zone PBX sipTrunkGroup PBX_TG policy media packetServiceProfile PBX_PSP
set addressContext default zone PBX sipTrunkGroup PBX_TG signaling messageManipulation outputAdapterProfile removephonecontext
commit

PSTN Gateway:

set addressContext default zone PSTN sipTrunkGroup PSTN_TG media mediaIpInterfaceGroupName WAN_IPIG
set addressContext default zone PSTN sipTrunkGroup PSTN_TG state enabled
set addressContext default zone PSTN sipTrunkGroup PSTN_TG mode inService
set addressContext default zone PSTN sipTrunkGroup PSTN_TG policy digitParameterHandling numberingPlan NANP_ACCESS
set addressContext default zone PSTN sipTrunkGroup PSTN_TG policy digitParameterHandling egressDmPmRule Rule_Digit_to_Ext

```
set addressContext default zone PSTN sipTrunkGroup PSTN_TG ingressIpPrefix <IP address
of PSTN Gateway> 32
set addressContext default zone PSTN sipTrunkGroup PSTN_TG policy signaling
ipSignalingProfile PSTN_IPSP
set addressContext default zone PSTN sipTrunkGroup PSTN_TG policy media
packetServiceProfile PSTN_PSP
set addressContext default zone PSTN sipTrunkGroup PSTN_TG signaling
messageManipulation inputAdapterProfile Digitmanip
commit
```

Amazon Chime Voice Connector:

```
set addressContext default zone AWS sipTrunkGroup AWS_TG media
mediaIpInterfaceGroupName LAN_IPIG
set addressContext default zone AWS sipTrunkGroup AWS_TG state enabled
set addressContext default zone AWS sipTrunkGroup AWS_TG mode inService
set addressContext default zone AWS sipTrunkGroup AWS_TG policy
digitParameterHandling numberingPlan NANP_ACCESS
set addressContext default zone AWS sipTrunkGroup AWS_TG policy
digitParameterHandling egressDmPmRule Rule_Digit_to_Ext
set addressContext default zone AWS sipTrunkGroup AWS_TG policy callRouting
elementRoutingPriority DEFAULT_IP
set addressContext default zone AWS sipTrunkGroup AWS_TG ingressIpPrefix 0.0.0.0 0
set addressContext default zone AWS sipTrunkGroup AWS_TG signaling transportPreference
preference1 udp
set addressContext default zone AWS sipTrunkGroup AWS_TG policy signaling
ipSignalingProfile AWS_IPSP
set addressContext default zone AWS sipTrunkGroup AWS_TG policy media
packetServiceProfile AWS_PSP
set addressContext default zone AWS sipTrunkGroup AWS_TG signaling
messageManipulation outputAdapterProfile AC_Req_Uri
commit
```

4.3.6 IP Peer

Asterisk FreePBX:

```
set addressContext default zone PBX ipPeer PBX_IPP ipAddress <IP address of PBX>
set addressContext default zone PBX ipPeer PBX_IPP ipPort 5060
set addressContext default zone PBX ipPeer PBX_IPP policy description PBX_IPP
set addressContext default zone PBX ipPeer PBX_IPP policy sip fqdn ""
set addressContext default zone PBX ipPeer PBX_IPP policy sip fqdnPort 0
set addressContext default zone PBX ipPeer PBX_IPP pathCheck profile PBX
set addressContext default zone PBX ipPeer PBX_IPP pathCheck hostName ""
set addressContext default zone PBX ipPeer PBX_IPP pathCheck hostPort 5060
set addressContext default zone PBX ipPeer PBX_IPP pathCheck state enabled
set addressContext default zone PBX ipPeer PBX_IPP pathCheck statusUpdateSupport
enabled
commit
```

PSTN Gateway:

```
set addressContext default zone PSTN ipPeer PSTN_IPP ipAddress <IP Address of PSTN
Gateway>
set addressContext default zone PSTN ipPeer PSTN_IPP ipPort 5060
set addressContext default zone PSTN ipPeer PSTN_IPP policy description PSTN_IPP
set addressContext default zone PSTN ipPeer PSTN_IPP policy sip fqdn ""
set addressContext default zone PSTN ipPeer PSTN_IPP policy sip fqdnPort 0
set addressContext default zone PSTN ipPeer PSTN_IPP pathCheck profile PSTN
set addressContext default zone PSTN ipPeer PSTN_IPP pathCheck hostName ""
set addressContext default zone PSTN ipPeer PSTN_IPP pathCheck hostPort 5060
set addressContext default zone PSTN ipPeer PSTN_IPP pathCheck state enabled
set addressContext default zone PSTN ipPeer PSTN_IPP pathCheck statusUpdateSupport
enabled
commit
```

Amazon Chime Voice Connector:

```
set addressContext default zone AWS ipPeer AWS_IPP ipPort 0
set addressContext default zone AWS ipPeer AWS_IPP policy description AWS_IPP
set addressContext default zone AWS ipPeer AWS_IPP policy sip fqdn <FQDN of Amazon
Chime SDK Voice Connector>
set addressContext default zone AWS ipPeer AWS_IPP policy sip fqdnPort 0
set addressContext default zone AWS ipPeer AWS_IPP pathCheck profile AWS
set addressContext default zone AWS ipPeer AWS_IPP pathCheck hostName <FQDN of
Amazon Chime SDK Voice Connector>
set addressContext default zone AWS ipPeer AWS_IPP pathCheck hostPort 0
set addressContext default zone AWS ipPeer AWS_IPP pathCheck state enabled
set addressContext default zone AWS ipPeer AWS_IPP pathCheck statusUpdateSupport
enabled
commit
```

4.3.7 PathCheck Profile

Asterisk FreePBX:

```
set profiles services pathCheckProfile PBX protocol sipOptions
set profiles services pathCheckProfile PBX sendInterval 30
set profiles services pathCheckProfile PBX replyTimeoutCount 3
set profiles services pathCheckProfile PBX recoveryCount 3
set profiles services pathCheckProfile PBX transportPreference preference1 udp
commit
```

PSTN Gateway:

```
set profiles services pathCheckProfile PSTN protocol sipOptions
set profiles services pathCheckProfile PSTN sendInterval 30
set profiles services pathCheckProfile PSTN replyTimeoutCount 3
set profiles services pathCheckProfile PSTN recoveryCount 3
set profiles services pathCheckProfile PSTN transportPreference preference1 udp
commit
```

Amazon Chime Voice Connector:

```
set profiles services pathCheckProfile AWS protocol sipOptions
set profiles services pathCheckProfile AWS sendInterval 30
set profiles services pathCheckProfile AWS replyTimeoutCount 3
set profiles services pathCheckProfile AWS recoveryCount 3
set profiles services pathCheckProfile AWS transportPreference preference1 udp
commit
```

4.3.8 Signaling Profile

Asterisk FreePBX:

```
set profiles signaling ipSignalingProfile PBX_IPSP
set profiles signaling ipSignalingProfile PBX_IPSP egressIpAttributes transport type1 udp
set profiles signaling ipSignalingProfile PBX_IPSP commonIpAttributes flags
includeTransportTypeInContactHeader enable
commit
```

PSTN Gateway:

```
set profiles signaling ipSignalingProfile PSTN_IPSP
set profiles signaling ipSignalingProfile PSTN_IPSP egressIpAttributes transport type1 tcp
set profiles signaling ipSignalingProfile PSTN_IPSP commonIpAttributes flags
includeTransportTypeInContactHeader enable
commit
```

Amazon Chime Voice Connector:

```
set profiles signaling ipSignalingProfile AWS_IPSP
set profiles signaling ipSignalingProfile AWS_IPSP egressIpAttributes transport type1
tlsOverTcp
set profiles signaling ipSignalingProfile AWS_IPSP commonIpAttributes flags
includeTransportTypeInContactHeader enable
set profiles signaling ipSignalingProfile AWS_IPSP egressIpAttributes
numberGlobalizationProfile DEFAULT_IP
commit
```

4.3.9 Codec

```
set profiles media codecEntry G711-DEFAULT codec g711
set profiles media codecEntry G711-DEFAULT dtmf relay rfc2833
set profiles media codecEntry G711-DEFAULT packetSize 20
commit
```

4.3.10 Packet Service Profile

Asterisk FreePBX:

```
set profiles media packetServiceProfile PBX_PSP
set profiles media packetServiceProfile PBX_PSP codec codecEntry1 G711-DEFAULT
set profiles media packetServiceProfile PBX_PSP preferredRtpPayloadTypeForDtmfRelay
101
```

set profiles media packetServiceProfile PBX_PSP silenceInsertionDescriptor g711SidRtpPayloadType 13
set profiles media packetServiceProfile PBX_PSP silenceInsertionDescriptor heartbeat enable
commit

PSTN Gateway:

set profiles media packetServiceProfile PSTN_PSP
set profiles media packetServiceProfile PSTN_PSP codec codecEntry1 G711-DEFAULT
set profiles media packetServiceProfile PSTN_PSP preferredRtpPayloadTypeForDtmfRelay 101
set profiles media packetServiceProfile PSTN_PSP silenceInsertionDescriptor g711SidRtpPayloadType 13
set profiles media packetServiceProfile PSTN_PSP silenceInsertionDescriptor heartbeat enable
commit

Amazon Chime Voice Connector:

set profiles media packetServiceProfile AWS_PSP
set profiles media packetServiceProfile AWS_PSP codec codecEntry1 G711-DEFAULT
set profiles media packetServiceProfile AWS_PSP preferredRtpPayloadTypeForDtmfRelay 101
set profiles media packetServiceProfile AWS_PSP silenceInsertionDescriptor g711SidRtpPayloadType 13
set profiles media packetServiceProfile AWS_PSP silenceInsertionDescriptor heartbeat enable
set profiles media packetServiceProfile AWS_PSP secureRtpRtcp cryptoSuiteProfile DEFAULT
set profiles media packetServiceProfile AWS_PSP secureRtpRtcp flags enableSrtp enable
commit

4.3.11 Routing Label

Asterisk FreePBX:

set global callRouting routingLabel PBX_RL overflowNumber ""
set global callRouting routingLabel PBX_RL overflowNOA none
set global callRouting routingLabel PBX_RL overflowNPI none
set global callRouting routingLabel PBX_RL routePrioritizationType sequence
set global callRouting routingLabel PBX_RL action routes
set global callRouting routingLabel PBX_RL numRoutesPerCall 10
set global callRouting routingLabel PBX_RL routingLabelRoute 1 routeType trunkGroup
set global callRouting routingLabel PBX_RL routingLabelRoute 1 trunkGroup PBX_TG
set global callRouting routingLabel PBX_RL routingLabelRoute 1 ipPeer PBX_IPP
set global callRouting routingLabel PBX_RL routingLabelRoute 1 proportion 0
set global callRouting routingLabel PBX_RL routingLabelRoute 1 cost 1000000
set global callRouting routingLabel PBX_RL routingLabelRoute 1 inService inService
set global callRouting routingLabel PBX_RL routingLabelRoute 1 testing normal
commit

PSTN Gateway:

set global callRouting routingLabel PSTN_RL overflowNumber ""
set global callRouting routingLabel PSTN_RL overflowNOA none
set global callRouting routingLabel PSTN_RL overflowNPI none
set global callRouting routingLabel PSTN_RL routePrioritizationType sequence

set global callRouting routingLabel PSTN_RL action routes
set global callRouting routingLabel PSTN_RL numRoutesPerCall 10
set global callRouting routingLabel PSTN_RL routingLabelRoute 1 routeType trunkGroup
set global callRouting routingLabel PSTN_RL routingLabelRoute 1 trunkGroup PSTN_TG
set global callRouting routingLabel PSTN_RL routingLabelRoute 1 ipPeer PSTN_IPP
set global callRouting routingLabel PSTN_RL routingLabelRoute 1 proportion 0
set global callRouting routingLabel PSTN_RL routingLabelRoute 1 cost 1000000
set global callRouting routingLabel PSTN_RL routingLabelRoute 1 inService inService
set global callRouting routingLabel PSTN_RL routingLabelRoute 1 testing normal
commit

4.3.12 Route

set global callRouting route trunkGroup PBX_TG RIBBON5210 standard Sonus_NULL 1 all all ALL none Sonus_NULL routingLabel PSTN_RL
set global callRouting route trunkGroup PSTN_TG RIBBON5210 standard Sonus_NULL 1 all all ALL none Sonus_NULL routingLabel PBX_RL
commit

4.3.13 Digit Manipulations

Phones at PBX are set with 4-digit extensions, and they are translated to 10-digit DID at SBC to pass-through the calls successfully towards PSTN network and vice versa. It is done using digit manipulation commands mentioned below:

Note: Manipulations are subjective and may vary based on the Enterprise customers requirement

1. **For Inbound calls from PSTN (Mandatory)**

Criteria called "Digit_to_Ext" is created to manipulate the 10-digit extension from PSTN to 4-digit in "To" header.

set profiles digitParameterHandling dmPmCriteria Digit_to_Ext criteriaType digit
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitType calledNumber
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext parameterPresenceCheck exists
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria egressFlag value send
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria egressFlag operation ignore
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria digitMatch value startDigitPosition 0
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria digitMatch value numberOfDigits 6
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria digitMatch value matchValue < Prefix that needs to be stripped for this manipulation>
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria digitMatch operation equals

```
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria natureOfAddress value 950
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria natureOfAddress operation ignore
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria numberingPlanIndicator value data
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria numberingPlanIndicator operation ignore
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria numberLength value 0
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria numberLength operation ignore
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria presentationMatch value none
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria presentationMatch operation ignore
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria screeningMatch value none
set profiles digitParameterHandling dmPmCriteria Digit_to_Ext digitCriteria screeningMatch operation ignore
commit
```

Below subRule is set to strip first 6 digits of the number in “To” header when applied to Digit_to_Ext criteria.

```
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 criteria Digit_to_Ext
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 ruleType digit
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation numberType calledNumber
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation numberParameterManipulation natureOfAddress none
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation numberParameterManipulation numberingPlanIndicator none
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation numberParameterManipulation numberLength noInput
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation numberParameterManipulation presentation none
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation numberParameterManipulation screening none
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation numberParameterManipulation includeInEgress none
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation digitStringManipulation startDigitPosition 0
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation digitStringManipulation numberOfDigits 6
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation digitStringManipulation replacement type constant
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation digitStringManipulation replacement digitString calledNumber
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation digitStringManipulation replacement startDigitPosition 0
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0 digitManipulation digitStringManipulation replacement numberOfDigits 0
```

```
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0
digitManipulation digitStringManipulation replacement value ""
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 0
digitManipulation digitStringManipulation action none
commit
```

2. **For Outbound calls to PSTN (Mandatory)**

Criteria called "EXT_to_DIGIT" is created to manipulate the 4-digit extension from PBX to 10-digit in "From" header.

```
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT criteriaType digit
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitType callingNumber
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT parameterPresenceCheck
exists
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria egressFlag
value send
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria egressFlag
operation ignore
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria digitMatch
value startDigitPosition 0
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria digitMatch
value numberOfDigits 3
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria digitMatch
value matchValue <First 3 common digits of extension pattern>
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria digitMatch
operation equals
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria
natureOfAddress value 950
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria
natureOfAddress operation ignore
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria
numberingPlanIndicator value data
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria
numberingPlanIndicator operation ignore
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria numberLength
value 0
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria numberLength
operation ignore
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria
presentationMatch value none
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria
presentationMatch operation ignore
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria
screeningMatch value none
set profiles digitParameterHandling dmPmCriteria EXT_to_DIGIT digitCriteria
screeningMatch operation ignore
commit
```

Below subRule is set to add first 6 digits of the number in "From" header when applied to EXT_to_DIGIT criteria.

```
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 criteria EXT_to_DIGIT
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 ruleType digit
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation numberType callingNumber
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation numberParameterManipulation natureOfAddress none
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation numberParameterManipulation numberingPlanIndicator none
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation numberParameterManipulation numberLength noInput
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation numberParameterManipulation presentation none
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation numberParameterManipulation screening none
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation numberParameterManipulation includeInEgress none
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation digitStringManipulation startDigitPosition 0
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation digitStringManipulation numberOfDigits 0
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation digitStringManipulation replacement type constant
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation digitStringManipulation replacement digitString callingNumber
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation digitStringManipulation replacement startDigitPosition 0
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation digitStringManipulation replacement numberOfDigits 10
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation digitStringManipulation replacement value <Prefix that needs to be added for this manipulation>
set profiles digitParameterHandling dmPmRule Rule_Digit_to_Ext subRule 1 digitManipulation digitStringManipulation action none
commit
```

Further, above rule is configured at Trunk Group level using below commands:

```
set addressContext default zone PBX sipTrunkGroup PBX_TG policy digitParameterHandling ingressDmPmRule Rule_Digit_to_Ext
set addressContext default zone PSTN sipTrunkGroup PSTN_TG policy digitParameterHandling ingressDmPmRule Rule_Digit_to_Ext
commit
```

4.3.14 Message manipulation

1. To modify Request URI header in requests sent to Amazon Chime SDK Voice Connector (Mandatory)

```
set profiles signaling sipAdaptorProfile AC_Req_Uri state enabled
set profiles signaling sipAdaptorProfile AC_Req_Uri advancedSMM enabled
set profiles signaling sipAdaptorProfile AC_Req_Uri profileType messageManipulation
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 applyMatchHeader one
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 criterion 1 type message
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 criterion 1 message
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 criterion 1 message messageTypes
request
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 criterion 1 message methodTypes
[ cancel invite ]
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 criterion 2 type header
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 criterion 2 header
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 criterion 2 header name request-
line
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 criterion 2 header condition exist
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 criterion 2 header hdrInstance all
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 action 1 type token
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 action 1 operation modify
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 action 1 from
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 action 1 from type value
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 action 1 from value <FQDN of
Amazon Chime SDK Voice Connector>
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 action 1 to
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 action 1 to type token
set profiles signaling sipAdaptorProfile AC_Req_Uri rule 1 action 1 to tokenValue
urihostname
commit
```

Above sipAdaptorProfile is then mapped to AWS trunk using below command,

```
set addressContext default zone AWS sipTrunkGroup AWS_TG signaling
messageManipulation outputAdapterProfile AC_Req_Uri
commit
```

2. To strip first 6 digits in meta-data information that is sent to Amazon Chime SDK Voice Connector (Mandatory)

Configuration of SIP Trunks is done in such a way that the SIPREC invite is triggered from the Ingress of PSTN Trunk hence the following manipulation is required to modify 10-digit extension to 4-digit extension in the meta data information that is being sent in the INVITE. Manipulations are subjective and may vary based on the Enterprise customers requirement in sending the Meta data

```
set profiles signaling sipAdaptorProfile Digitmanip state enabled
set profiles signaling sipAdaptorProfile Digitmanip advancedSMM enabled
set profiles signaling sipAdaptorProfile Digitmanip rule 1 applyMatchHeader one
```

```
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 1 type message
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 1 message
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 1 message messageTypes
requestAll
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 2 type header
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 2 header
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 2 header name request-line
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 2 header condition exist
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 2 header hdrInstance all
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 3 type token
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 3 token
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 3 token condition exist
set profiles signaling sipAdaptorProfile Digitmanip rule 1 criterion 3 token tokenType
uriusername
set profiles signaling sipAdaptorProfile Digitmanip rule 1 action 1 type token
set profiles signaling sipAdaptorProfile Digitmanip rule 1 action 1 operation regdel
set profiles signaling sipAdaptorProfile Digitmanip rule 1 action 1 to
set profiles signaling sipAdaptorProfile Digitmanip rule 1 action 1 to type token
set profiles signaling sipAdaptorProfile Digitmanip rule 1 action 1 to tokenValue uriusername
set profiles signaling sipAdaptorProfile Digitmanip rule 1 action 1 regexp
set profiles signaling sipAdaptorProfile Digitmanip rule 1 action 1 regexp string <Prefix that
needs to be stripped>
set profiles signaling sipAdaptorProfile Digitmanip rule 1 action 1 regexp matchInstance all
set profiles signaling sipAdaptorProfile Digitmanip rule 2 applyMatchHeader one
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 1 type message
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 1 message
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 1 message messageTypes
requestAll
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 2 type header
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 2 header
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 2 header name To
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 2 header condition exist
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 2 header hdrInstance all
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 3 type token
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 3 token
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 3 token condition exist
set profiles signaling sipAdaptorProfile Digitmanip rule 2 criterion 3 token tokenType
uriusername
set profiles signaling sipAdaptorProfile Digitmanip rule 2 action 1 type token
set profiles signaling sipAdaptorProfile Digitmanip rule 2 action 1 operation regdel
set profiles signaling sipAdaptorProfile Digitmanip rule 2 action 1 to
set profiles signaling sipAdaptorProfile Digitmanip rule 2 action 1 to type token
set profiles signaling sipAdaptorProfile Digitmanip rule 2 action 1 to tokenValue uriusername
set profiles signaling sipAdaptorProfile Digitmanip rule 2 action 1 regexp
set profiles signaling sipAdaptorProfile Digitmanip rule 2 action 1 regexp string <Prefix that
needs to be stripped>
set profiles signaling sipAdaptorProfile Digitmanip rule 2 action 1 regexp matchInstance all
commit
```

Above sipAdaptorProfile is then mapped to PSTN trunk using below command,

```
set addressContext default zone PSTN sipTrunkGroup PSTN_TG signaling
messageManipulation inputAdapterProfile Digitmanip
commit
```

3. **To remove phone-context parameter as Asterisk FreePBX doesn't accept request with this parameter (Mandatory)**

```
set profiles signaling sipAdaptorProfile removephonecontext state enabled
set profiles signaling sipAdaptorProfile removephonecontext advancedSMM enabled
set profiles signaling sipAdaptorProfile removephonecontext rule 1 applyMatchHeader all
set profiles signaling sipAdaptorProfile removephonecontext rule 1 criterion 1 type message
set profiles signaling sipAdaptorProfile removephonecontext rule 1 criterion 1 message
set profiles signaling sipAdaptorProfile removephonecontext rule 1 criterion 1 message
messageTypes requestAll
set profiles signaling sipAdaptorProfile removephonecontext rule 1 criterion 2 type header
set profiles signaling sipAdaptorProfile removephonecontext rule 1 criterion 2 header
set profiles signaling sipAdaptorProfile removephonecontext rule 1 criterion 2 header name
request-line
set profiles signaling sipAdaptorProfile removephonecontext rule 1 criterion 2 header
condition exist
set profiles signaling sipAdaptorProfile removephonecontext rule 1 criterion 2 header
hdrInstance all
set profiles signaling sipAdaptorProfile removephonecontext rule 1 action 1 type parameter
set profiles signaling sipAdaptorProfile removephonecontext rule 1 action 1 operation delete
set profiles signaling sipAdaptorProfile removephonecontext rule 1 action 1 paramType
userinfo
set profiles signaling sipAdaptorProfile removephonecontext rule 1 action 1 to
set profiles signaling sipAdaptorProfile removephonecontext rule 1 action 1 to type parameter
set profiles signaling sipAdaptorProfile removephonecontext rule 1 action 1 to value phone-
context
commit
```

Above sipAdaptorProfile is then mapped to PBX trunk using below command,

```
set addressContext default zone PBX sipTrunkGroup PBX_TG signaling
messageManipulation outputAdapterProfile removephonecontext
commit
```

4.3.15 SRS Group Profile

```
set global servers srsGroupProfile SRSGROUP description ""
set global servers srsGroupProfile SRSGROUP loadDistribution sequence
set global servers srsGroupProfile SRSGROUP numSimultaneousStream 1
set global servers srsGroupProfile SRSGROUP srsGroupData 0 transport udp
set global servers srsGroupProfile SRSGROUP srsGroupData 0 ipAddress ""
set global servers srsGroupProfile SRSGROUP srsGroupData 0 fqdn < FQDN of Amazon
Chime SDK Voice Connector>
set global servers srsGroupProfile SRSGROUP srsGroupData 0 fqdnPort 0
set global servers srsGroupProfile SRSGROUP srsGroupData 0 ipTGId AWS_TG
set global servers srsGroupProfile SRSGROUP srsGroupData 0 srtp disable
commit
```

4.3.16 SRS Group Cluster

```
set global servers srsGroupCluster SRSGRP1 srsGroupClusterData 0 srsGroupId SRSGROUP
commit
```

4.3.17 Call recording criteria

```
set global servers callRecordingCriteria AWS srsGroupClusterId SRSGRP1
set global servers callRecordingCriteria AWS nextHopIP 0.0.0.0
set global servers callRecordingCriteria AWS previousHopIP 0.0.0.0
set global servers callRecordingCriteria AWS recordingType allLegs
set global servers callRecordingCriteria AWS recorderType SIPRec
set global servers callRecordingCriteria AWS criteriaState enable
commit
```

4.3.18 SIPREC using TLS as Transport

4.3.18.1 Import Public CA Root Certificate

The uploaded Trust certificate was provided by Amazon and the certificate can be downloaded from the Amazon Chime SDK Voice Connector console.

```
set system security pki certificate AWS_VC state enabled
set system security pki certificate AWS_VC fileName AWS.der
set system security pki certificate AWS_VC type remote
commit
```

4.3.18.2 TLS Profile

```
set profiles security tlsProfile AWS_PROF appAuthTimer 5
set profiles security tlsProfile AWS_PROF handshakeTimer 5
set profiles security tlsProfile AWS_PROF sessionResumpTimer 3600
set profiles security tlsProfile AWS_PROF cipherSuite1 rsa-with-aes-128-cbc-sha
set profiles security tlsProfile AWS_PROF cipherSuite2 rsa-with-aes-128-cbc-sha-256
set profiles security tlsProfile AWS_PROF cipherSuite3 rsa-with-aes-256-cbc-sha-256
set profiles security tlsProfile AWS_PROF allowedRoles clientandserver
set profiles security tlsProfile AWS_PROF authClient false
set profiles security tlsProfile AWS_PROF clientCertName AWS_VC
set profiles security tlsProfile AWS_PROF serverCertName AWS_VC
set profiles security tlsProfile AWS_PROF acceptableCertValidationErrors none
set profiles security tlsProfile AWS_PROF v1_0 enabled
set profiles security tlsProfile AWS_PROF v1_1 enabled
set profiles security tlsProfile AWS_PROF v1_2 enabled
set profiles security tlsProfile AWS_PROF suppressEmptyFragments disabled
set profiles security tlsProfile AWS_PROF peerNameVerify disabled
set profiles security tlsProfile AWS_PROF hashType sha1
commit
```

4.3.18.3 SipSignaling port

set addressContext default zone AWS sipSigPort 3 tlsProfileName AWS_PROF
set addressContext default zone AWS sipSigPort 3 transportProtocolsAllowed sip-tls-tcp
commit

4.3.18.4 SIP Trunk

set addressContext default zone AWS sipTrunkGroup AWS_TG signaling transportPreference preference1 tls-tcp
commit

4.3.18.5 Signaling profile

set profiles signaling ipSignalingProfile AWS_IPSP egressIpAttributes transport type1 tlsOverTcp
commit

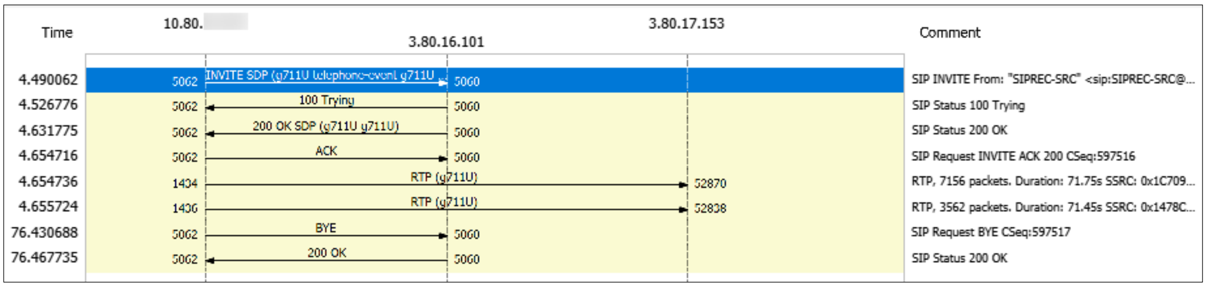
4.3.18.6 Pathcheck Profile

set profiles services pathCheckProfile AWS transportPreference preference1 tls-tcp
commit

4.3.18.7 SRS group profile

set global servers srsGroupProfile SRSGROUP srsGroupData 0 transport tls
set global servers srsGroupProfile SRSGROUP srsGroupData 0 ipTGId AWS_TG
set global servers srsGroupProfile SRSGROUP srsGroupData 0 srtp enable
commit

5 Sample SIPREC trace between SBC and Amazon Chime SDK Voice Connector with meta-data information



INVITE sip:SIPREC-SRS@gdnbXXXXXXXXXXXX.voiceconnector.chime.aws:5060 SIP/2.0
Via: SIP/2.0/UDP 10.80.X.X:5062;branch=z9hG4bK00Bf400997037fdcec7
From: "SIPREC-SRC" <sip:SIPREC-SRC@10.80.X.X>;tag=gK0073bbf5
To: "SIPREC-SRS" <sip:SIPREC-SRS@gdnbXXXXXXXXXXXX.voiceconnector.chime.aws>

```
   Call-ID: 25_16777283_96419925@10.80.X.X
   CSeq: 597516 INVITE
   Max-Forwards: 70
   Allow:
INVITE,ACK,CANCEL,BYE,REGISTER,REFER,INFO,SUBSCRIBE,NOTIFY,PRACK,UPDATE,OPTIONS,MESS
AGE,PUBLISH
   Accept: application/sdp, application/rs-metadata-request, application/rs-metadata
   Contact: "SIPREC-SRC" <sip:SIPREC-SRC@10.80.X.X:5062>;+sip.src
   Require: siprec
   Supported: timer,100rel
   Session-Expires: 1800
   Min-SE: 90
   Content-Length:  3208
   Content-Type: multipart/mixed;boundary=sonus-content-delim
   MIME-Version: 1.0

   --sonus-content-delim
   Content-Disposition: session; handling=required
   Content-Length:   411
   Content-Type: application/sdp

   v=0
   o=Sonus_UAC 664747 964569 IN IP4 10.80.X.X
   s=SIP Media Capabilities
   t=0 0
   m=audio 1436 RTP/AVP 0 101
   c=IN IP4 10.80.X.X
   a=label:1
   a=rtpmap:0 PCMU/8000
   a=rtpmap:101 telephone-event/8000
   a=fmtp:101 0-15
   a=sendonly
   a=maxptime:10
   m=audio 1434 RTP/AVP 0 101
   c=IN IP4 10.80.X.X
   a=label:2
   a=rtpmap:0 PCMU/8000
   a=rtpmap:101 telephone-event/8000
   a=fmtp:101 0-15
   a=sendonly
   a=maxptime:10

   --sonus-content-delim
   Content-Disposition: recording-session; handling=required
   Content-Length:  2490
   Content-Type: application/rs-metadata+xml

   <?xml version="1.0" encoding="UTF-8"?>
   <recording xmlns='urn:ietf:params:xml:ns:recording'>
      <datamode>complete</datamode>
      <group group_id="OGU5YzNiODAtMTc1Mi0xMA==">
```

```
      <associate-time>2023-08-07T13:16:35Z</associate-time>
      <callData xmlns='urn:ietf:params:xml:ns:callData'>
        <fromhdr>&quot;Parthasarathi S&quot;
&lt;sip:2145509054@10.64.1.72&gt;;tag=1c1429872079</fromhdr>
        <tohdr>&lt;sip:0084@10.80.X.X&gt;;tag=gK00f3b9e7</tohdr>
        <callid>1782503021782023816 34@10.80.X.X</callid>
        <gcid>25</gcid>
      </callData>
    </group>
    <session session_id="OGU5YzdlYmUtMTc1Mi0xMA==">
      <group-ref>OGU5YzNiODAtMTc1Mi0xMA==</group-ref>
      <start-time>2023-08-07T13:16:35Z</start-time>
    </session>
    <participant participant_id="OGU5YzNiODEtMTc1Mi0xMA==">
      <nameID aor="2145509054@10.64.1.72">
        <name xml:lang="en">Parthasarathi S</name>
      </nameID>
    </participant>
    <participant participant_id="OGU5YzNiODItMTc1Mi0xMA==">
      <nameID aor="0084@10.80.X.X">
        <name xml:lang="en"> </name>
      </nameID>
    </participant>
    <stream stream_id="OGU5YzNiODQtMTc1Mi0xMA=="
session_id="OGU5YzdlYmUtMTc1Mi0xMA==">
      <label>1</label>
      <associate-time>2023-08-07T13:16:35Z</associate-time>
    </stream>
    <stream stream_id="OGU5YzNiODUtMTc1Mi0xMA=="
session_id="OGU5YzdlYmUtMTc1Mi0xMA==">
      <label>2</label>
      <associate-time>2023-08-07T13:16:35Z</associate-time>
    </stream>
    <sessionrecordingassoc session_id="OGU5YzdlYmUtMTc1Mi0xMA==">
      <associate-time>2023-08-07T13:16:35Z</associate-time>
    </sessionrecordingassoc>
    <participantsessionassoc participant_id="OGU5YzNiODEtMTc1Mi0xMA=="
session_id="OGU5YzdlYmUtMTc1Mi0xMA==">
      <associate-time>2023-08-07T13:16:35Z</associate-time>
    </participantsessionassoc>
    <participantsessionassoc participant_id="OGU5YzNiODItMTc1Mi0xMA=="
session_id="OGU5YzdlYmUtMTc1Mi0xMA==">
      <associate-time>2023-08-07T13:16:35Z</associate-time>
    </participantsessionassoc>
    <participantstreamassoc participant_id="OGU5YzNiODEtMTc1Mi0xMA==">
      <send>OGU5YzNiODUtMTc1Mi0xMA==</send>
      <recv>OGU5YzNiODQtMTc1Mi0xMA==</recv>
    </participantstreamassoc>
    <participantstreamassoc participant_id="OGU5YzNiODItMTc1Mi0xMA==">
      <send>OGU5YzNiODQtMTc1Mi0xMA==</send>
      <recv>OGU5YzNiODUtMTc1Mi0xMA==</recv>
```

```
    </participantstreamassoc>
  </recording>

  --sonus-content-delim--

  SIP/2.0 100 Trying
  Via: SIP/2.0/UDP
10.80.X.X:5062;branch=z9hG4bK00Bf400997037fdcec7;rport=5062;received=199.182.124.60
  From: "SIPREC-SRC" <sip:SIPREC-SRC@10.80.X.X>;tag=gK0073bbf5
  To: "SIPREC-SRS" <sip:SIPREC-SRS@gdnbXXXXXXXXXXXX.voiceconnector.chime.aws>
  Call-ID: 25_16777283_96419925@10.80.X.X
  CSeq: 597516 INVITE
  Content-Length: 0

  SIP/2.0 200 OK
  Via: SIP/2.0/UDP
10.80.X.X:5062;rport=5062;received=199.182.124.60;branch=z9hG4bK00Bf400997037fdcec7
  Record-Route: <sip:3.80.16.101;lr;ftag=gK0073bbf5;did=9241.5713;nat=yes>
  From: "SIPREC-SRC" <sip:SIPREC-SRC@10.80.X.X>;tag=gK0073bbf5
  To: "SIPREC-SRS" <sip:SIPREC-
SRS@gdnbXXXXXXXXXXXX.voiceconnector.chime.aws>;tag=DD0SZgDv987Kp
  Call-ID: 25_16777283_96419925@10.80.X.X
  CSeq: 597516 INVITE
  Contact: <sip:10.0.39.27:5060>
  Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, INFO, REGISTER
  Content-Type: application/sdp
  Content-Length: 267
  X-Vine-ID: b65cac75-fac0-4a7b-a65f-346d25a286fb

  v=0
  o=- 1691414198423 1691414198423 IN IP4 3.80.17.153
  s=session
  c=IN IP4 3.80.17.153
  t=0 0
  m=audio 52838 RTP/AVP 0
  a=rtpmap:0 PCMU/8000
  a=recvonly
  a=rtcp:52839
  a=ptime:20
  m=audio 52870 RTP/AVP 0
  a=rtpmap:0 PCMU/8000
  a=recvonly
  a=rtcp:52871
  a=ptime:20

  ACK sip:10.0.39.27:5060 SIP/2.0
  Via: SIP/2.0/UDP 10.80.X.X:5062;branch=z9hG4bK00Bf4016940278b44f6
  From: "SIPREC-SRC" <sip:SIPREC-SRC@10.80.X.X>;tag=gK0073bbf5
  To: "SIPREC-SRS" <sip:SIPREC-
SRS@gdnbXXXXXXXXXXXX.voiceconnector.chime.aws>;tag=DD0SZgDv987Kp
  Call-ID: 25_16777283_96419925@10.80.X.X
```

```
   CSeq: 597516 ACK
   Max-Forwards: 70
   Route: <sip:3.80.16.101;lr;ftag=gK0073bbf5;did=9241.5713;nat=yes>
   Content-Length: 0

   BYE sip:10.0.39.27:5060 SIP/2.0
   Via: SIP/2.0/UDP 10.80.X.X:5062;branch=z9hG4bK00Bf41ea511278b44f6
   From: "SIPREC-SRC" <sip:SIPREC-SRC@10.80.X.X>;tag=gK0073bbf5
   To: "SIPREC-SRS" <sip:SIPREC-
SRS@gdnbXXXXXXXXXXXX.voiceconnector.chime.aws>;tag=DD0SZgDv987Kp
   Call-ID: 25_16777283_96419925@10.80.X.X
   CSeq: 597517 BYE
   Max-Forwards: 70
   Route: <sip:3.80.16.101;lr;ftag=gK0073bbf5;did=9241.5713;nat=yes>
   Content-Length:   651
   Content-Disposition: recording-session
   Content-Type: application/rs-metadata+xml

   <?xml version="1.0" encoding="UTF-8"?>
       <recording xmlns='urn:ietf:params:xml:ns:recording'>
       <datamode>Partial</datamode>
       <session session_id="OGU5YzdlYmUtMTc1Mi0xMA==">
       </session>
       <participant
       participant_id="OGU5YzNiODEtMTc1Mi0xMA=="
       session_id="OGU5YzdlYmUtMTc1Mi0xMA==">
       <disassociate-time>2023-08-07T13:17:47Z</disassociate-time>
       </participant>
       <participant
       participant_id="OGU5YzNiODItMTc1Mi0xMA=="
       session_id="OGU5YzdlYmUtMTc1Mi0xMA==">
       <disassociate-time>2023-08-07T13:17:47Z</disassociate-time>
       </participant>
       </recording>

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP
10.80.X.X:5062;rport=5062;received=199.182.124.60;branch=z9hG4bK00Bf41ea511278b44f6
   From: "SIPREC-SRC" <sip:SIPREC-SRC@10.80.X.X>;tag=gK0073bbf5
   To: "SIPREC-SRS" <sip:SIPREC-
SRS@gdnbXXXXXXXXXXXX.voiceconnector.chime.aws>;tag=DD0SZgDv987Kp
   Call-ID: 25_16777283_96419925@10.80.X.X
   CSeq: 597517 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 3) PBX extension number and outbound caller number are captured in the metadata (callerID capture to be tested)	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User Recording 2: PBX User to 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		
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 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 while PBX User-1 is on call with PBX User-2 + PBX User-2's audio to PSTN User

		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 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.

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	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

			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		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 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 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 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.

			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		
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	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.

			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		
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	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.

			up or last participant from PBX User-1 and User-2 hangs up		
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	1) Call is connected 2) RTP between PBX User-1 and PSTN User-1 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

		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	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		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	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

			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		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 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 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) 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.
17	Inbound call with extended consult	Inbound call from PSTN to PBX User-1. PBX User-1	1) Call is connected	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to 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	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		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 the call. PBX User-1 then adds PBX User-3 to the call. Call ends when	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	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,

		either PSTN or last PBX User in the call hangs up	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-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	1) Call is connected when customer answers call from PBX User-2 2) RTP between PBX User-2 and customer on PSTN is captured.	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

		participants hangs up	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		Meta data information only has PBX User-2's extension and PSTN User number.
20	Emergency calling	PBX User-1 calls the 411 service	1) Call is connected 2) RTP between PBX User and 411 is captured 3) PBX extension number and outbound caller number (411) 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 411 user hangs up	Passed	Two call recordings are available in AWS S3. Recording 1: 411 User to PBX User Recording 2: PBX User to 411 User Note: - This scenario is locally simulated within Lab environment.
21	Outbound international call	Outbound call from PBX User-1 to	1) Call is connected	Passed	Two call recordings are available in AWS S3.

		international PSTN number	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 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 1: International PSTN User to PBX User Recording 2: PBX User to International PSTN User

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 5) Streaming and recording end when either PBX or PSTN 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

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 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 User or PSTN hangs up	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 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-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 + PBX User-2'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	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 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.
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	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

			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		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	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

			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		Recording 4: PBX User-1's audio to PSTN user 2 + PSTN 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 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	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.

			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		
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	1) Call is connected 2) RTP between PBX User-1 and PSTN is captured	Passed	Two call recordings are available in AWS S3. Recording 1: PSTN User to PBX User-1 and PBX User-2

		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	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		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.	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	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

		PSTN User-2 is conferenced into the call. The call ends when one of the last two call participants hangs up	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		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	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

		the last two call 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 5) Inbound caller number and PBX User-1 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 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.

			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		
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 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	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 + 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.

			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) 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.
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	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	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

		resumes call with PSTN. This sequence may be repeated multiple times until either PSTN or PBX User-1 hangs up	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		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 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	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

			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		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	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.

			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		
20	Emergency calling	PBX User-1 calls the 411 service	1) Call is connected 2) RTP between PBX User and 411 is captured 3) PBX extension number and outbound caller number (411) 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 411 user hangs up	Passed	Two call recordings are available in AWS S3. Recording 1: 411 User to PBX User Recording 2: PBX User to 411 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		