

## **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

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## **Document History**

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1.0	August-22-2023	SIPREC Configuration Guide	
1.1	September-13-2023	Incorporated the changes based on comments	
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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:

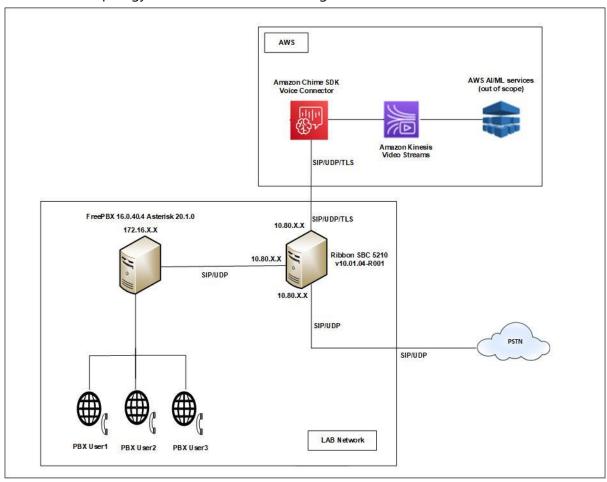


Figure 1: Network Topology

The signaling and media flow is illustrated below:

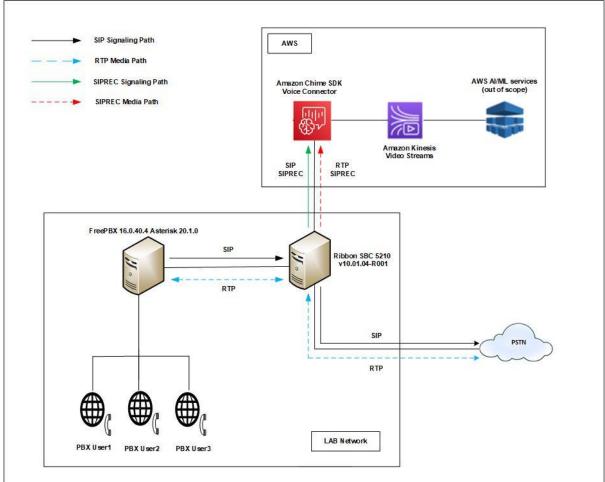


Figure 2: Signaling and Media Flow

## 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)
  - o VVX 150
  - o 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	<b>√</b>		
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.

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

Table 2 – PBX and ESBC Configuration Steps

### 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

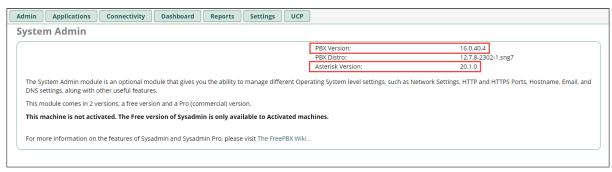


Figure 3: FreePBX Asterisk Version

#### 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

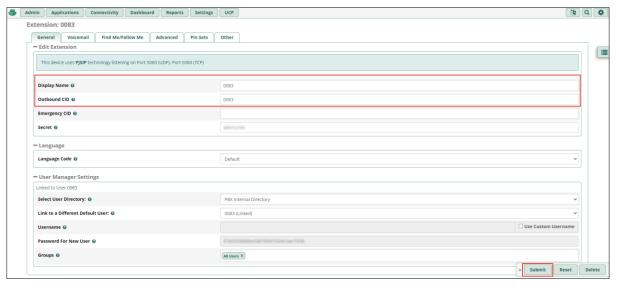


Figure 4: Asterisk Extension

• 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

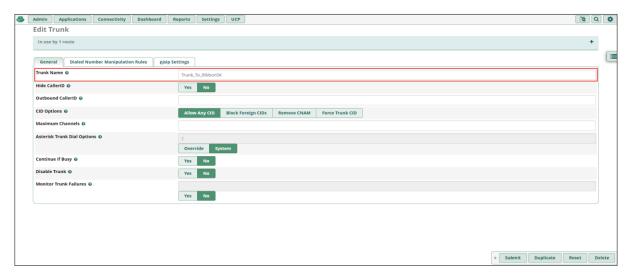


Figure 6: Asterisk Trunk

#### 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

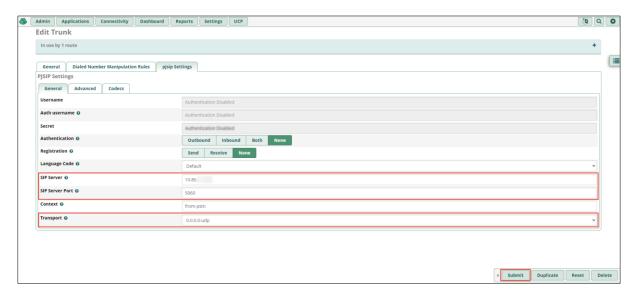


Figure 7: Asterisk Trunk Continuation

#### Navigate to Pjsip settings → Codecs

#### **Enable** Ulaw

Click Submit

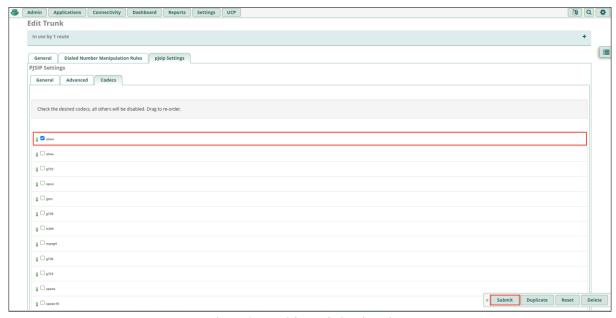


Figure 8: Asterisk Trunk Continuation

#### 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

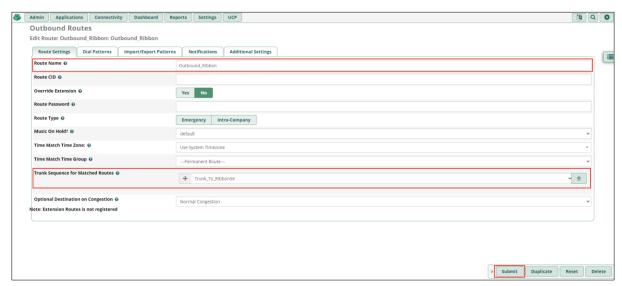


Figure 9: Asterisk Outbound Route

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

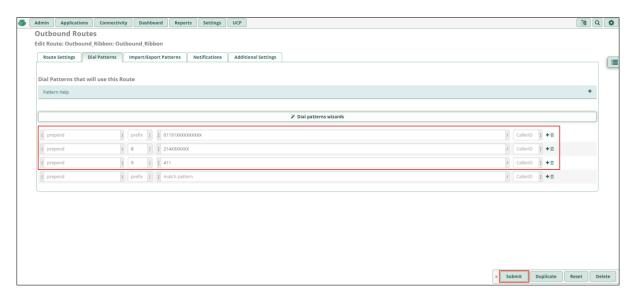


Figure 10: Asterisk Outbound Route Continuation

#### 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,

```
🧖 10.64.4.243 - PuTTY
login as: admin
a admin@10.64.4.243's password:
This system is restricted to authorized users only.
Unauthorized access or access attempts to this system or services are prohibited. All user activity is logged.
Evidence of unauthorized use collected during monitoring
may be provided to appropriate personnel for administrative, criminal or other adverse action.
 ******************
admin connected from 172.16.29.239 using ssh on ribbon5k
Your last successful login was at 2023-8-17 2:26:16
Your last successful login was from 172.16.29.239
 admin@ribbon5k> swinfo -v
 SERVER:
BIOS:
                               V02.07.00
V10.00.02-R007
EMA:
 SBC:
SBC Type:
                               V10.01.04-R001
                                isbc
 Installed host role: active Current host role: active
Build Workspace: jenkinsbuild.tx.sbx100104R0fix
Stream: //sbx/v10_01_04R0_fix
Change: 529092
CTP Change: 527452
Build Number: 15
Build Time: Wed May 24 03:51:19 CDT 2023
Build 11me: Wed May 24 03:51:19 CDT 2023
Build Host: busterdev2-tx
SBC Version: V10.01.04-R001
Required BMC Version: v03.23.00-R000
Required BIOS Version: v2.7.0
Required OS Version: 10.00.02-R007
Required Bluefin BMC Version: v03.23.00-R000
Required Bluefin BMC Version: v2.14.0
Pegmired Vellowfin BMC Version: v03.23.00-R000
Required Yellowfin BMC Version: v03.23.00-R000
Required Yellowfin BIOS Version: v1.18.0
  [ok][2023-08-17 02:28:51]
```

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 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 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 overflowNoA none
set global callRouting routingLabel PBX_RL overflowNPI none
set global callRouting routingLabel PBX_RL routePrioritizationType sequence
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 inService inService
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 numRoutesPerCall 10
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 digit Parameter<br/>Handling dm Pm Criteria Digit\_to\_Ext digit Criteria number<br/>Length 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 digit Parameter<br/>Handling dm Pm Criteria EXT\_to\_DIGIT digit Criteria number<br/>Length value  $\boldsymbol{0}$ 

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-
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

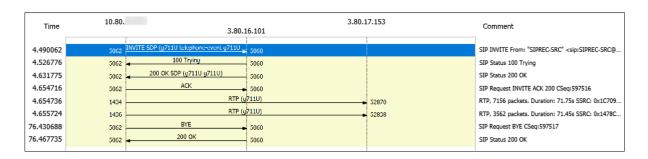
#### 4.3.18.6 Pathcheck Profile

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@gdnbXXXXXXXXXXXXX.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@gdnbXXXXXXXXXXXXX.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
  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>&guot;Parthasarathi S&guot;
<sip:2145509054@10.64.1.72&gt;;tag=1c1429872079</fromhdr>
       <tohdr>&lt;sip:0084@10.80.X.X&gt;;tag=gK00f3b9e7</tohdr>
       <callid>178250302178202381634@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="OGU5YzdIYmUtMTc1Mi0xMA==">
     <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=="</pre>
session id="OGU5YzdIYmUtMTc1Mi0xMA==">
     <associate-time>2023-08-07T13:16:35Z</associate-time>
    </participantsessionassoc>
    <participantsessionassoc participant_id="OGU5YzNiODItMTc1Mi0xMA=="</pre>
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@gdnbXXXXXXXXXXXXX.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
  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@gdnbXXXXXXXXXXXXX.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@gdnbXXXXXXXXXXXXX.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="OGU5YzdIYmUtMTc1Mi0xMA=="> <disassociate-time>2023-08-07T13:17:47Z</disassociate-time> </participant> <participant</pre> participant id="OGU5YzNiODItMTc1Mi0xMA==" session id="OGU5YzdIYmUtMTc1Mi0xMA=="> <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@gdnbXXXXXXXXXXXXX.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 <u>Call Analytics website</u>.

Test	Title	Procedure	Expected Results	Status	Comments
Case ID					
1	Inbound call from	Inbound Call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	PSTN	PSTN to PBX User	2) RTP between PSTN and PBX		
			User is captured		Recording 1: PSTN User to PBX User
			3) Inbound caller number and		
			PBX extension number are		Recording 2: PBX User to PSTN User
			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		
2	Outhound call to	Outhound call frame	user hangs up	Desced	Two cell recordings are quallable in AVA/C C2
2	Outbound call to PSTN	Outbound call from PBX user to PSTN	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	PSIN	PBX user to PSTN	2) RTP between PBX User and		Pacarding 1: DCTN Usar to DDV Usar
			PSTN is captured 3) PBX extension number and		Recording 1: PSTN User to PBX User
			outbound caller number are		Recording 2: PBX User to PSTN User
			captured in the metadata		NECOLULIS 2. FBA OSEL LO FSTIN OSEL
			(callerID capture to be tested)		
			(caller to be tested)		

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

6	Outbound call -	Outbound call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	attended call	PBX User-1 to	2) RTP between PSTN and PBX		
	transfer	PSTN. PBX User-1	User-1 is captured		Recording 1: PSTN User to PBX User-1 and PBX
		does an attended	3) RTP is not captured		User-2
		transfer to PBX	between PSTN and PBX User-1		
		User-2	during transfer		Recording 2: PBX user 1's audio to PSTN + Music on
			4) RTP between PSTN and PBX		Hold while PBX User-1 is on call with PBX User-2 +
			User-2 is captured after		PBX User-2's audio to PSTN User
			transfer		
			5) Outbound caller number		There is no re-invite from PBX during transfer.
			and PBX User-1 extension		Hence, Music on Hold is captured, and meta data is
			number are captured in the		not updated with PBX User-2's extension.
			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		
7	Inbound call -	Inbound call from	1) Call is connected	Passed	Four call recordings are available in AWS S3.
	external transfer	PSTN User-1 to PBX	2) RTP between PSTN and PBX		J. 1 . 1 . 2
		User-1, PBX User-1	User-1 is captured		Recording 1: PBX User-1's audio to PSTN User-1 +
		does an attended	3) RTP is not captured		Music on Hold while PBX User-1 is on call with
		transfer to PSTN	between PSTN User-1 and PBX		PSTN User-2 + PSTN User-2's audio with PSTN User-
		User-2	User-1 during transfer		1
		000, 2			

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

			O) The 12 control of 12		
			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 -	Inbound call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	internal	PSTN to PBX User-	2) RTP between PSTN and PBX		
	conference	1. PBX User-1	User-1 is captured		Recording 1: PSTN User to PBX User-1 and PBX
		places PSTN on	3) RTP is not captured		User-2
		hold and consults	between PSTN and PBX User-1		
		with PBX User-2.	during setup of call with PBX		Recording 2: PBX User-1's audio to PSTN + Music
		PBX User-2 is	User-2		on Hold from PBX User-1 while consulting PBX
		conferenced into	4) RTP between PSTN, PBX		User-2 for conference + PBX User-1's audio to PSTN
		the call. The call	User-1, and PBX User-2 is		User and PBX User-2 + PBX User-2's audio to PSTN
		terminates when	captured after PBX User-2 is		User and PBX User-1
		one of the last two	added to the call as an active		
		call participants	participant		There is no mid-call signaling from PBX for call
		hangs up	5) Inbound caller number and		escalation to conference. Therefore, music on hold
			PBX User-1 extension number		is recorded while escalating the call to conference
			are captured in the metadata		and meta data is not updated with PBX User-2's
			6) PBX User-2 extension		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		
11	Outbound call -	Outbound call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	internal	PBX User-1 to	2) RTP between PBX User-1		
	conference	PSTN. PBX User-1	and PSTN is captured		Recording 1: PSTN User to PBX User-1 and PBX
		places PSTN on	3) RTP is not captured		User-2
		hold and consults	between PSTN and PBX User-1		
		with PBX User-2.	during setup of call with PBX		Recording 2: PBX User-1's audio to PSTN + Music
		PBX User-2 is	User-2		on Hold from PBX User-1 while consulting PBX
		conferenced into	4) RTP between PSTN, PBX		User-2 for conference + PBX User-1's audio to PSTN
		the call. The call	User-1, and PBX User-2 is		User and PBX User-2 + PBX User-2's audio to PSTN
		terminates when	captured after PBX User-2 is		User and PBX User-1
		one of the last two	added to the call as an active		
		call participants	participant		There is no mid-call signaling from PBX for call
		hangs up	5) Outbound caller number		escalation to conference. Therefore, music on hold
			and PBX User-1 extension		is recorded while escalating the call to conference
			number are captured in the		and meta data is not updated with PBX User-2's
			metadata		extension.
			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		l l
			User-1 and User-2 hangs up		

12	Inbound call with	Inbound call from	1) Call is connected	Passed	Four call recordings are available in AWS S3.
	external	PSTN User-1 to PBX	2) RTP between PSTN and PBX		-
	conference	User-1. PBX User-1	User-1 is captured		Recording 1: PBX User-1's audio to PSTN User-1+
		places PSTN User-1	3) RTP is not captured		Music on Hold from PBX User-1 while consulting
		on hold and calls	between PSTN and PBX User-1		PSTN User-2 for conference + PSTN User-2's audio
		with PSTN User-2.	during setup of call with PSTN		and PBX User-1's audio to PSTN User-1
		PSTN User-2 is	User-2		
		conferenced into	4) RTP between PBX User-1		Recording 2: PSTN User-1's audio to PBX User-1
		the call. The call	and PSTN User-2 is captured		and PSTN User-2
		ends when one of	5) RTP between PSTN User-1,		
		the last two call	PBX User-1, and PSTN User-2		Recording 3: PSTN User-2's audio to PBX User-1
		participants hangs	is captured after PSTN User-2		and PSTN User-1
		up	is added to the call as an		
			active participant		Recording 4: PBX User-1's audio to PSTN User-2 +
			6) Inbound caller number and		PSTN User-1's audio and PBX User-1's audio to
			PBX User-1 extension number		PSTN User-2
			are captured in the metadata		
			7) PSTN User-2 caller number		There is no mid-call signaling from PBX for call
			is added to the metadata after		escalation to conference. Therefore, music on hold
			the conference starts		is recorded while escalating the call to conference.
			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		
12		0 11 1 11 1	call participants hangs up		F
13	Outbound call	Outbound call from	1) Call is connected	Passed	Four call recordings are available in AWS S3.
	with external	PBX User-1 to PSTN	2) RTP between PBX User-1		D. II. 4 DOVIN AL. II. 4 DOTANI
	conference	User-1. PBX User-1	and PSTN User-1 is captured		Recording 1: PBX User-1's audio to PSTN User-1 +
		places PSTN User-1			Music on Hold from PBX User-1 while consulting

			3) 575	T	DOTALLE OF CONTRACT CONTRACT
		on hold and calls	3) RTP is not captured		PSTN User-2 for conference + PSTN User-2's audio
		PSTN User-2. PSTN	between PSTN User-1 and PBX		and PBX User-1's audio to PSTN User-1
		User-2 is	User-1 during setup of call		
		conferenced into	with PSTN User-2		Recording 2: PSTN User-1's audio to PBX User-1
		the call. The call	4)RTP between PBX User-1		and PSTN User-2
		ends when one of	and PSTN User-2 is captured.		
		the last two call	5) RTP between PSTN User-1,		Recording 3: PSTN User-2's audio to PBX User-1
		participants hangs	PBX User-1, and PSTN User-2		and PSTN User-1
		up	is captured after PSTN User-2		
		,	is added to the call as an		Recording 4: PBX User-1's audio to PSTN User-2 +
			active participant		PSTN User-1's audio and PBX User-1's audio to
			6) Outbound caller number		PSTN User-2
			and PBX User-1 extension		
			number are captured in the		There is no mid-call signaling from PBX for call
			metadata		escalation to conference. Therefore, music on hold
			7) PSTN User-2 caller number		is recorded while escalating the call to conference.
			is added to the metadata after		is recorded write escalating the can to conference.
			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		
1.4	Inbound call -	Inbound call from	call participants hangs up	Dassad	Tive cell recordings are qualled in ANAC 52
14			1) Call is connected	Passed	Two call recordings are available in AWS S3.
	transfer to queue	PSTN to PBX User-	2) RTP between PSTN and PBX		Board Co. 4. BCTNIII BDV II 4
		1. PBX User-1	User-1 is captured		Recording 1: PSTN User to PBX User-1
		transfers the call to	3) RTP is not captured		
		call queue. PSTN	between PSTN and PBX User-1		Recording 2: PBX User-1's audio to PSTN User +
		drops the call	during transfer		Music on Hold while PBX User-1 attempts to

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User
transfer.
d meta data is
and PBX User-

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	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  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.
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

		J BCTN	2) PTD   PCTN -   2227		
		places PSTN on	2) RTP between PSTN and PBX		
		hold and calls PBX	User-1 is captured only when		Recording 2: PBX User-1's audio to PSTN User +
		User-2, who	call is not on hold		Music On Hold while PBX User-1 is on call with PBX
		answers. PBX User-	3) RTP between PBX User-1		User-2 + resumed PBX User-1's audio to PSTN User
		2 is put on hold and	and PBX User-2 is not		
		PBX User-1	captured		There is no re-invite from PBX while the call is
		resumes call with	4) Inbound caller number and		placed on HOLD. The recording is not paused and
		PSTN. This	PBX extension number are		the music on hold is captured.
		sequence may be	captured in the metadata		
		repeated multiple	5) Metadata is captured when		
		times until either	PBX User-2 is added and when		
		PSTN or PBX User-1	they are dropped from the call		
		hangs up	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		
18	Inbound call with	Inbound call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	multi-party	PSTN to PBX User-	2) RTP between PSTN and PBX		
	conference	1. PBX User-1	User-1 is captured		Recording 1: PSTN User to PBX User-1, PBX User-2,
		places PSTN on	3) RTP is not captured		and PBX User-3
		hold and consults	between PSTN and PBX User-1		and i by oser s
		with PBX User-2.	during setup of call with PBX		Recording 2: PBX User-1's audio with PSTN User +
		PBX User-2 is	User-2		Music On Hold from PBX User-1 while consulting
		conferenced into	4) RTP between PSTN, PBX		PBX User-2 for conference + PBX User-1's audio to
		the call. PBX User-1	User-1, and PBX User-2 is		PSTN User and PBX User-2 + PBX User-2's audio to
		then adds PBX	captured after PBX User-2 is		PSTN User and PBX User-1 + Music On Hold from
		User-3 to the call.	added to the call as an active		PBX User-1 while consulting PBX User-3 for
					_
		Call ends when	participant		conference + PBX User-1's audio to PBX User-2,

		either PSTN or last	5) RTP between PSTN, PBX		PBX User-3 and PSTN User + PBX User-2's audio to
		PBX User in the call	User-1, PBX User-2, and PBX		PBX User-1, PBX User-3 and PSTN User + PBX User-
		hangs up	User-3 is captured after PBX		3's audio to PBX User-1, PBX User-2 and PSTN User
		0.1	User-3 is added to the call		,
			6) Inbound caller number and		There is no mid-call signaling from PBX for call
			PBX User-1 extension number		escalation to conference. Therefore, music on hold
			are captured in the metadata		is recorded while escalating the call to conference
			7) PBX User-2 extension		and meta data is not updated with PBX User-2's
			number is added to the		and PBX User-3's extensions.
			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		
19	Outbound	PBX User-1 calls	1) Call is connected when	Passed	Two call recordings are available in AWS S3.
	conference call	PBX User-2. PBX	customer answers call from		
		User-2 calls	PBX User-2		Recording 1: PSTN User to PBX User-1 and PBX
		customer on PSTN	2) RTP between PBX User-2		User-2
		number. Call ends	and customer on PSTN is		
		when either of the	captured.		Recording 2: PBX User-2's audio to PBX User-1 and
		last two call			PSTN User + PBX User-1's audio to PBX User-2 and
					PSTN User

		participants hangs	3) RTP between PBX User-1,		
		up	PBX User-2 and customer is		Meta data information only has PBX User-2's
			captured		extension and PSTN User number.
			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		
20	Emergency calling	PBX User-1 calls	1) Call is connected	Passed	Two call recordings are available in AWS S3.
20	Liner gency canning	the 411 service	2) RTP between PBX User and	l usseu	Two can recordings are available in 7000 55.
		the 111 service	411 is captured		Recording 1: 411 User to PBX User
			3) PBX extension number and		Theodramig II 111 oder to 1 bx oder
			outbound caller number (411)		Recording 2: PBX User to 411 User
			are captured in the metadata		Theodramig 211 BX oser to 122 oser
			(caller ID capture to be tested)		Note: - This scenario is locally simulated within Lab
			4) There is one call recording		environment.
			per call leg for the duration of		C.IVII O.IIII C.IVII
			the call, with accurate start		
			and end timestamps		
			5) Streaming and recording		
			end when either PBX or 411		
			user hangs up		
21	Outbound	Outbound call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	international call	PBX User-1 to		. 45564	The same contains are available in 7440 55.
1	carriacionar can	1 27, 030, 1 10			

international PSTN	2) RTP between PBX Users and	Recording 1: International PSTN User to PBX User
number	PSTN is captured	
	3) PBX extension number and	Recording 2: PBX User to International PSTN User
	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	

## **6.2 With TLS as Transport**

Test	Title	Procedure	Expected Results	Status	Comments
Case ID					
1	Inbound call from	Inbound Call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	PSTN	PSTN to PBX User	2) RTP between PSTN and PBX		
			User is captured		Recording 1: PSTN User to PBX User
			3) Inbound caller number and		
			PBX extension number are		Recording 2: PBX User to PSTN User
			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		
2	Outbound call to	Outbound call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	PSTN	PBX user to PSTN	2) RTP between PBX User and	1 03300	Two can recordings are available in 74w3 55.
		. 5% 436. 13111	PSTN is captured		Recording 1: PSTN User to PBX User
			3) PBX extension number and		<b>3</b>
			outbound caller number are		Recording 2: PBX User to PSTN User
			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		

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

5	Inbound call -	Inbound Call from	1) Call is connected	Passed	Two call recordings are available in AWS S3
	attended call	PSTN to PBX User-	2) RTP between PSTN and PBX		
	transfer	1, PBX User-1 does	User-1 is captured		Recording 1: PSTN User to PBX User-1 and PBX
		an attended	3) RTP is not captured		User-2
		transfer to PBX	between PSTN and PBX User-1		
		User-2	during transfer		Recording 2: PBX User-1's audio to PSTN + Music
			4) RTP between PSTN and PBX		on Hold while PBX User-1 is on call with PBX User-2
			User-2 is captured after		+ PBX User-2's audio to PSTN User
			transfer		
			5) Inbound caller number and		There is no re-invite from PBX during transfer.
			PBX User-1 extension number		Hence, Music on Hold is captured, and meta data is
			are captured in the metadata		not updated with PBX User-2's extension.
			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		
6	Outbound call -	Outbound call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	attended call	PBX User-1 to	2) RTP between PSTN and PBX		
	transfer	PSTN. PBX User-1	User-1 is captured		Recording 1: PSTN User to PBX User-1 and PBX
		does an attended	3) RTP is not captured		User-2
		transfer to PBX	between PSTN and PBX User-1		
		User-2	during transfer		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) DTD		
			4) RTP between PSTN and PBX		
			User-2 is captured after		There is no re-invite from PBX during transfer.
			transfer		Hence, Music on Hold is captured, and meta data is
			5) Outbound caller number		not updated with PBX User-2's extension.
			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		
7	Inbound call -	Inbound call from	1) Call is connected	Passed	Four call recordings are available in AWS S3.
'	external transfer	PSTN User-1 to PBX	2) RTP between PSTN and PBX	l usseu	Tour can recordings are available in 7445 55.
	CACCITION CHAIRSICI	User-1, PBX User-1	User-1 is captured		Recording 1: PBX User-1's audio to PSTN User-1 +
		does an attended	3) RTP is not captured		Music on Hold while PBX User-1 is on call with
		transfer to PSTN	between PSTN User-1 and PBX		PSTN User-2 + PSTN User-2's audio with PSTN User-
		User-2	User-1 during transfer		1
		0361-2	4) RTP between PSTN User-1		
			1 -		Pacarding 2: DCTN Hear 1's audio to DDV Hear 1
			and PSTN User-2 is captured		Recording 2: PSTN User-1's audio to PBX User-1
			after transfer		and PSTN User-2
			5) Inbound caller number and		December 2: DCTN Heave 21s and a to DDY Heave 4
			PBX User-1 extension number		Recording 3: PSTN User-2's audio to PBX User-1
			are captured in metadata		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 -	Outbound call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	blind call transfer	PBX User-1 to	2) RTP between PSTN and PBX		
		PSTN. PBX User-1	User-1 is captured		Recording 1: PSTN User to PBX User-1 and PBX
		does a blind	3) RTP is not captured		User-2
		transfer to PBX	between PSTN and PBX User-1		
		User-2	during transfer		Recording 2: PBX User-1's audio to PSTN + Music
			4) RTP between PSTN and PBX		on Hold while PBX User-1 attempts to transfer +
			User-2 is captured after		PBX User-2's audio to PSTN User
			transfer		
			5) Outbound caller number		There is no re-invite from PBX during transfer.
			and PBX User-1 extension		Hence, Music on Hold is captured, and meta data is
			number are captured in the		not updated with PBX User-2's extension.
			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		

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

		hold and consults	3) RTP is not captured		
		with PBX User-2.	between PSTN and PBX User-1		Recording 2: PBX User-1's audio to PSTN + Music
		PBX User-2 is	during setup of call with PBX		on Hold from PBX User-1 while consulting PBX
		conferenced into	User-2		User-2 for conference + PBX User-1's audio to PSTN
		the call. The call	4) RTP between PSTN, PBX		User and PBX User-2 + PBX User-2's audio to PSTN
		terminates when	User-1, and PBX User-2 is		User and PBX User-1
		one of the last two	captured after PBX User-2 is		333. 4.14 . 27 333. 2
		call participants	added to the call as an active		There is no mid-call signaling from PBX for call
		hangs up	participant		escalation to conference. Therefore, music on hold
		1101185 015	5) Outbound caller number		is recorded while escalating the call to conference
			and PBX User-1 extension		and meta data is not updated with PBX User-2's
			number are captured in the		extension.
			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		
12	Inbound call with	Inbound call from	1) Call is connected	Passed	Four call recordings are available in AWS S3.
	external	PSTN User-1 to PBX	2) RTP between PSTN and PBX		
	conference	User-1. PBX User-1	User-1 is captured		Recording 1: PBX User-1's audio to PSTN User-1 +
		places PSTN User-1	3) RTP is not captured		Music on Hold from PBX User-1 while consulting
		on hold and calls	between PSTN and PBX User-1		PSTN User-2 for conference + PSTN User-2's audio
		with PSTN User-2.	during setup of call with PSTN		and PBX User-1's audio to PSTN User-1

		DCTN Lloor 2 :c	Hear 2		
		PSTN User-2 is	User-2		December 2: DCTN Heart 41: 1: 1: DDV Heart 4
		conferenced into	4) RTP between PBX User-1		Recording 2: PSTN User-1's audio to PBX User-1
		the call. The call	and PSTN User-2 is captured		and PSTN User-2
		ends when one of	5) RTP between PSTN User-1,		
		the last two call	PBX User-1, and PSTN User-2		Recording 3: PSTN User-2's audio to PBX User-1
		participants hangs	is captured after PSTN User-2		and PSTN User-1
		up	is added to the call as an		
			active participant		Recording 4: PBX User-1's audio to PSTN User-2 +
			6) Inbound caller number and		PSTN User-1's audio and PBX User-1's audio to
			PBX User-1 extension number		PSTN User-2
			are captured in the metadata		
			7) PSTN User-2 caller number		There is no mid-call signaling from PBX for call
			is added to the metadata after		escalation to conference. Therefore, music on hold
			the conference starts		is recorded while escalating the call to conference.
			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		
13	Outbound call	Outbound call from	1) Call is connected	Passed	Four call recordings are available in AWS S3.
	with external	PBX User-1 to PSTN	2) RTP between PBX User-1		, and the second
	conference	User-1. PBX User-1	and PSTN User-1 is captured		Recording 1: PBX User-1's audio to PSTN User-1 +
		places PSTN User-1	3) RTP is not captured		Music on Hold from PBX User-1 while consulting
		on hold and calls	between PSTN User-1 and PBX		PSTN User-2 for conference + PSTN User-2's audio
		PSTN User-2. PSTN	User-1 during setup of call		and PBX User-1's audio to PSTN User-1
		User-2 is	with PSTN User-2		
		conferenced into	4)RTP between PBX User-1		Recording 2: PSTN User-1's audio to PBX User-1
		the call. The call	and PSTN User-2 is captured.		and PSTN User-2
		ends when one of			
	l			I	I.

		the last two call	5) RTP between PSTN User-1,		Recording 3: PSTN User-2's audio to PBX User-1
		participants hangs	PBX User-1, and PSTN User-2		and PSTN User-1
		up	is captured after PSTN User-2		and 1 3114 GSC1 1
		ap	is added to the call as an		Recording 4: PBX User-1's audio to PSTN User-2 +
			active participant		PSTN User-1's audio and PBX User-1's audio to
			6) Outbound caller number		PSTN User-2
			and PBX User-1 extension		T STIV OSCI Z
			number are captured in the		There is no mid-call signaling from PBX for call
			metadata		escalation to conference. Therefore, music on hold
			7) PSTN User-2 caller number		is recorded while escalating the call to conference.
			is added to the metadata after		is recorded write escalating the can to conference.
			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		
14	Inbound call -	Inbound call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	transfer to queue	PSTN to PBX User-	2) RTP between PSTN and PBX		0.11.11
	1	1. PBX User-1	User-1 is captured		Recording 1: PSTN User to PBX User-1
		transfers the call to	3) RTP is not captured		3 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2
		call queue. PSTN	between PSTN and PBX User-1		Recording 2: PBX User-1's audio to PSTN User +
		drops the call	during transfer		Music on Hold while PBX User-1 attempts to
			4) RTP is captured when		transfer until extensions associated to call queue
			queue accepts call		are ringing
			5) Inbound caller number and		
			PBX User-1 extension number		There is no re-invite from PBX during transfer.
			are captured in the metadata		Hence, Music on Hold is captured, and meta data is
					not updated with call queue number.

			<ul><li>6) Queue number is captured in the metadata after call transfer</li><li>7) There is one call recording per call leg for the duration of the call</li></ul>		
			8) The timestamps for the		
			recording show accurate start		
			and end times		
			9) Streaming and recording		
4.5	1.1	tales and sall to an	end when PSTN hangs up	D I	To sell a control of the transfer AMC CO
15	Inbound call - transfer to queue	Inbound call from PSTN to PBX User-	1) Call is connected 2) RTP between PSTN and PBX	Passed	Two call recordings are available in AWS S3.
	then to agent	1. PBX User-1	User-1 is captured		Recording 1: PSTN User to PBX User-1
	then to agent	transfers the call to	3) RTP is not captured		Necolating 1. F3114 OSEI to FBX OSEI-1
		call queue. PBX	between PSTN and PBX User-1		Recording 2: PBX User-1's audio to PSTN User +
		User-2 picks up the	during transfer		Music on Hold while PBX User-1 attempts to
		call from the queue	4) RTP is captured when		transfer + PBX User-2's audio to PSTN User
			queue accepts call		
			5) RTP between PSTN and PBX		There is no re-invite from PBX during transfer.
			User-2 is captured		Hence, Music on Hold is captured, and meta data is
			6) Inbound caller number and		not updated with call queue number and PBX User-
			PBX User-1 extension number		2's extension.
			are captured in the metadata		
			7) Queue number is captured		
			in the metadata after call		
			transfer		
			8) There is one call recording		
			per call leg for the duration of		
			the call		
			9) The timestamps for the		
			recording show accurate start		
			and end times		

			10) Streaming and recording end when PSTN hangs up		
16	Inbound call with consult	Inbound call from PSTN to PBX User-	1) Call is connected 2) RTP between PSTN and PBX	Passed	Two call recordings are available in AWS S3.
		1. PBX User-1 places PSTN on	User-1 is captured only when call is not on hold		Recording 1: PSTN User to PBX User-1
		hold and calls PBX User-2, who	3) RTP between PBX User-1 and PBX User-2 is not		Recording 2: PBX User-1's audio to PSTN User + Music On Hold while PBX User-1 is on call with PBX
		answers. PBX User- 2 hangs up and PBX	captured 4) Inbound caller number and		User-2 + resumed PBX User-1's audio to PSTN User
		User-1 resumes call with PSTN	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		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-	user hangs up  1) Call is connected 2) RTP between PSTN and PBX	Passed	Two call recordings are available in AWS S3.
		1. PBX User-1 places PSTN on	User-1 is captured only when call is not on hold		Recording 1: PSTN User to PBX User-1
		hold and calls PBX User-2, who answers. PBX User- 2 is put on hold and PBX User-1	3) RTP between PBX User-1 and PBX User-2 is not captured		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	4) Inbound caller number and		There is no re-invite from PBX while the call is
		PSTN. This	PBX extension number are		
					placed on HOLD. The recording is not paused and
		sequence may be	captured in the metadata		the music on hold is captured.
		repeated multiple	5) Metadata is captured when		
		times until either	PBX User-2 is added and when		
		PSTN or PBX User-1	they are dropped from the call		
		hangs up	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		
18	Inbound call with	Inbound call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	multi-party	PSTN to PBX User-	2) RTP between PSTN and PBX		
	conference	1. PBX User-1	User-1 is captured		Recording 1: PSTN User to PBX User-1, PBX User-2,
		places PSTN on	3) RTP is not captured		and PBX User-3
		hold and consults	between PSTN and PBX User-1		
		with PBX User-2.	during setup of call with PBX		Recording 2: PBX User-1's audio with PSTN User +
		PBX User-2 is	User-2		Music On Hold from PBX User-1 while consulting
		conferenced into	4) RTP between PSTN, PBX		PBX User-2 for conference + PBX User-1's audio to
		the call. PBX User-1	User-1, and PBX User-2 is		PSTN User and PBX User-2 + PBX User-2's audio to
		then adds PBX	captured after PBX User-2 is		PSTN User and PBX User-1 + Music On Hold from
		User-3 to the call.	added to the call as an active		PBX User-1 while consulting PBX User-3 for
		Call ends when	participant		conference + PBX User-1's audio to PBX User-2,
		either PSTN or last	5) RTP between PSTN, PBX		PBX User-3 and PSTN User + PBX User-2's audio to
		PBX User in the call	User-1, PBX User-2, and PBX		PBX User-1, PBX User-3 and PSTN User + PBX User-
		hangs up	User-3 is captured after PBX		3's audio to PBX User-1, PBX User-2 and PSTN User
			User-3 is added to the call		3 3 4 4 4 1 5 1 5 N 6 3 CT 1, 1 5 N 6 3 CT 2 4 N 4 1 3 N 6 3 CT
			See. 5 is added to the can		There is no mid-call signaling from PBX for call
					escalation to conference. Therefore, music on hold
					escalation to conference. Therefore, music on hold

			6) Inbound caller number and		is recorded while escalating the call to conference
			1 ·		is recorded while escalating the call to conference
			PBX User-1 extension number		and meta data is not updated with PBX User-2's
			are captured in the metadata		and PBX User-3's extensions.
			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		
19	Outbound	PBX User-1 calls	1) Call is connected when	Passed	Two call recordings are available in AWS S3.
	conference call	PBX User-2. PBX	customer answers call from		
		User-2 calls	PBX User-2		Recording 1: PSTN User to PBX User-1 and PBX
		customer on PSTN	2) RTP between PBX User-2		User-2
		number. Call ends	and customer on PSTN is		
		when either of the	captured.		Recording 2: PBX User-2's audio to PBX User-1 and
		last two call	3) RTP between PBX User-1,		PSTN User + PBX User-1's audio to PBX User-2 and
		participants hangs	PBX User-2 and customer is		PSTN User
		up	captured		
		'	4) PBX User-1, PBX User-2, and		Meta data information only has PBX User-2's
			customer called number are		extension and PSTN User number.
			captured in the metadata		
	L	L	tapta. ca in the included		

			C) Thomasia and sell researching		
			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	1) Call is connected	Passed	Two call recordings are available in AWS S3.
		the 411 service	2) RTP between PBX User and		
			411 is captured		Recording 1: 411 User to PBX User
			3) PBX extension number and		
			outbound caller number (411)		Recording 2: PBX User to 411 User
			are captured in the metadata		
			(caller ID capture to be tested)		Note: - This scenario is locally simulated within Lab
			4) There is one call recording		environment.
			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		
21	Outbound	Outbound call from	1) Call is connected	Passed	Two call recordings are available in AWS S3.
	international call	PBX User-1 to	2) RTP between PBX Users and		
		international PSTN	PSTN is captured		Recording 1: International PSTN User to PBX User
		number	3) PBX extension number and		-
			outbound caller number are		Recording 2: PBX User to International PSTN User
			captured in the metadata		-
			(caller ID capture to be tested)		
	1	l	1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1		

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	