



Amazon Chime SDK Voice Connector

SIPREC Configuration Guide

**FreePBX 16.0.40, Asterisk 20.1.0 and
Oracle 4600 Enterprise Session Border
Controller (Oracle 4600 ESBC)**

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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 **Oracle 4600 Enterprise Session Border Controller** (Oracle 4600 ESBC) 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 for SIPREC reference configuration is illustrated below and is representative of **FreePBX Asterisk** and **Oracle 4600 ESBC**.

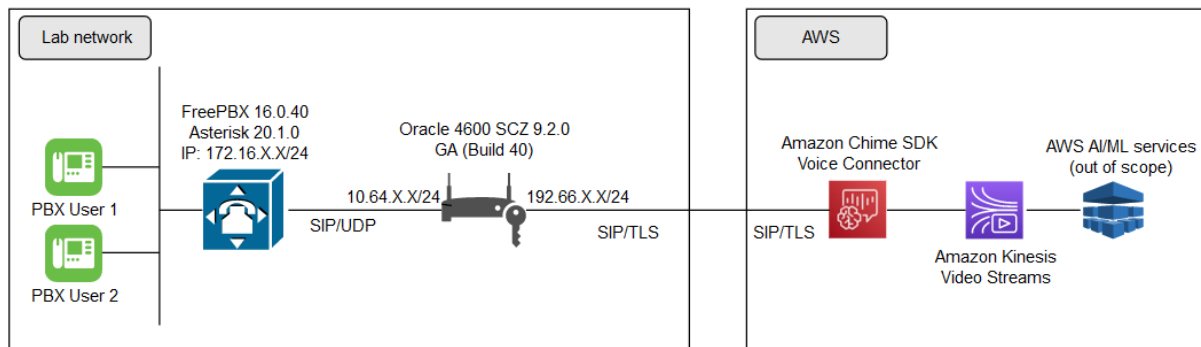


Figure 1: Network Topology

2.1 Hardware Components

- VMWare server running ESXi 7.0 or later used for the following virtual machine
 - FreePBX Asterisk
- Oracle 4600 ESBC
- Polycom IP Phone(s)
 - VVX150
 - VVX250

2.2 Software Requirements

- FreePBX 16.0.40 Asterisk 20.1.0
- Oracle 4600 ESBC SCZ9.2.0 GA (Build 40)

3 Features

3.1 Features Supported

The following PBX features were tested:

- Inbound Call
- Outbound Call
- Call Hold
- Attended Transfer
- Unattended Transfer

3.2 Features Not Supported

- None

3.3 Features Not Tested

- None

3.4 Caveats and Limitations

- None

4 Configuration

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

4.1 Configuration Checklist

This section presents an overview of the steps that are required to configure FreePBX Asterisk and Oracle 4600 ESBC for SIPREC using SIP Trunking with Amazon Chime SDK Voice Connector.

Table 1 – PBX and ESBC Configuration Steps

Steps	Description	Reference
Step 1	FreePBX Asterisk Configuration	Section 4.2
Step 2	Oracle 4600 ESBC Configuration	Section 4.3
Step 3	Amazon Chime Voice Connector Configuration	Amazon Chime Voice Connector

4.2 FreePBX Asterisk Configuration

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

4.2.1 FreePBX Asterisk Version

4.2.1.1 FreePBX Version

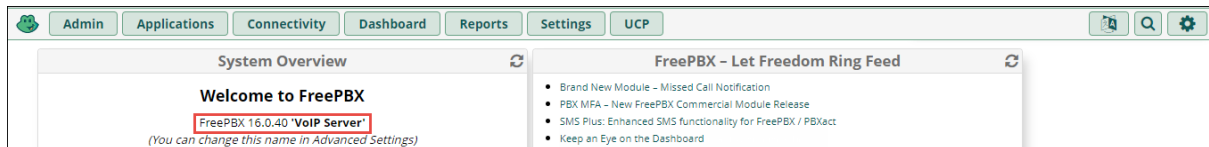


Figure 2: FreePBX Version

4.2.1.2 Asterisk Version

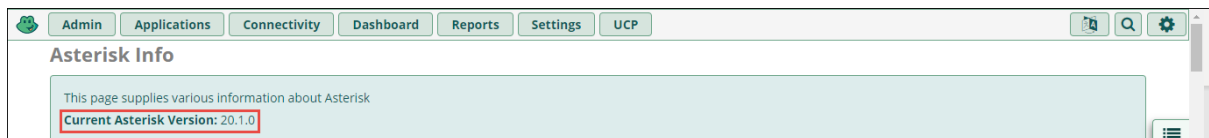


Figure 3: Asterisk Version

4.2.2 Trunk

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

Trunk Name: Enter a name for the Trunk

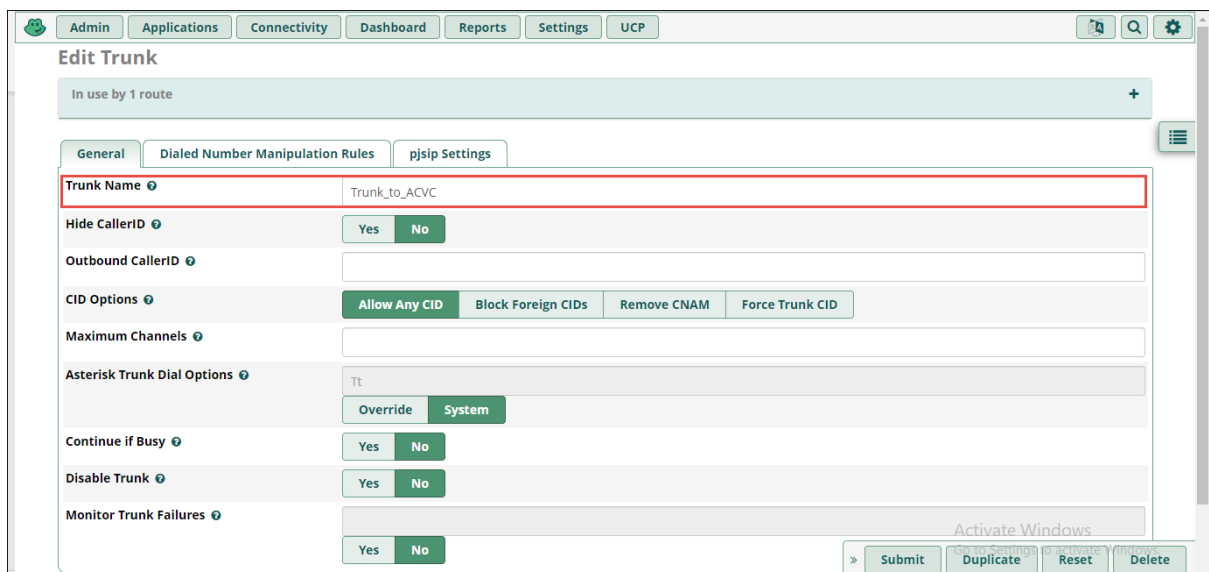


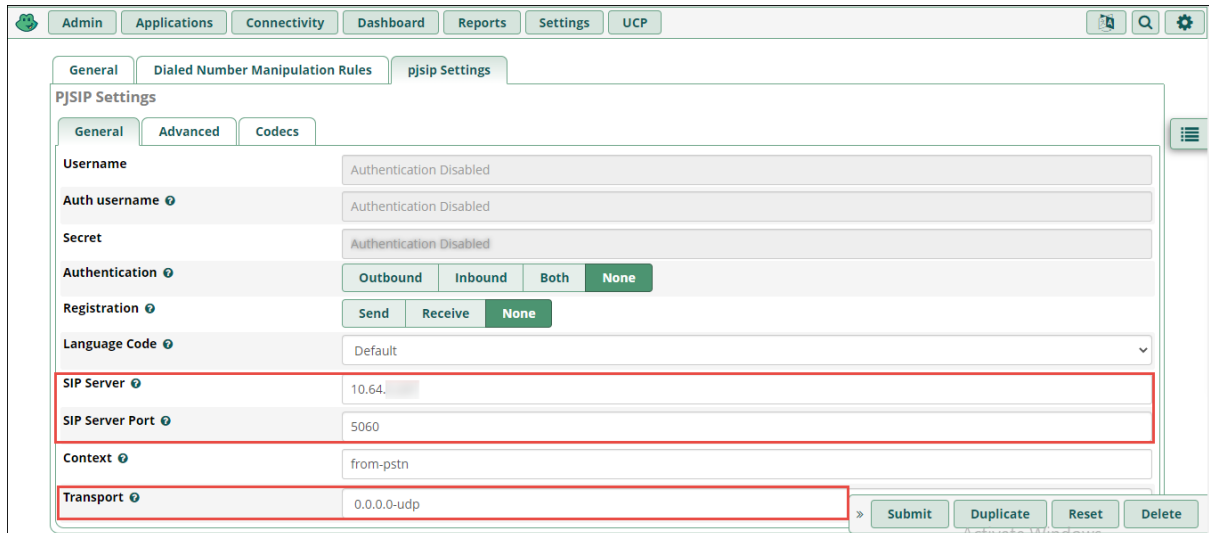
Figure 4: Asterisk Trunk

Navigate to **Pjsip settings** → **General**

SIP Server: 10.X.X.X (IP of Oracle ESBC's Network Interface towards the FreePBX Asterisk)

SIP Server Port: 5060

Transport: 0.0.0.0-udp



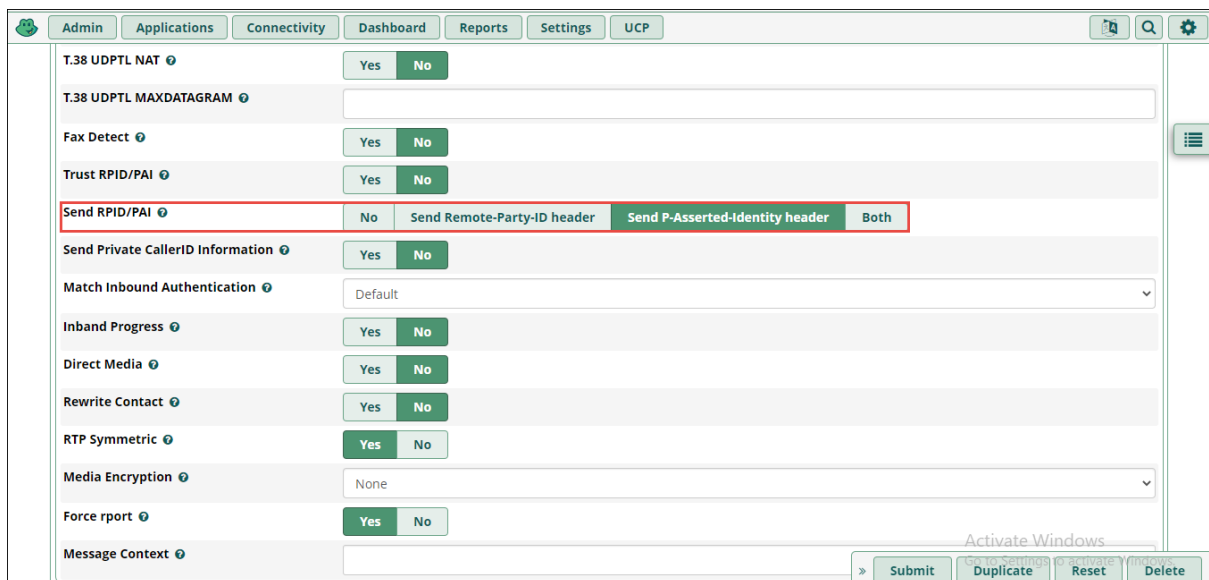
The screenshot shows the Asterisk administration interface for PJSIP Settings. The 'General' tab is selected. The 'SIP Server' field is set to '10.64.', 'SIP Server Port' is '5060', and 'Transport' is '0.0.0.0-udp'. These three fields are highlighted with a red border. Other fields include Username, Auth username, Secret, Authentication (set to 'None'), Registration (set to 'None'), Language Code (set to 'Default'), and Context (set to 'from-pstn').

Field	Value
Username	Authentication Disabled
Auth username	Authentication Disabled
Secret	Authentication Disabled
Authentication	Outbound Inbound Both None
Registration	Send Receive None
Language Code	Default
SIP Server	10.64.
SIP Server Port	5060
Context	from-pstn
Transport	0.0.0.0-udp

Figure 5: Asterisk Trunk Continuation

Navigate to **Pjsip settings** → **Advanced**

Send RPID/PAI: Select **Send P-Asserted-Identity header**



The screenshot shows the Asterisk administration interface for PJSIP Settings - Advanced tab. The 'Send RPID/PAI' field is highlighted with a red border, and the 'Send P-Asserted-Identity header' option is selected. Other fields include T.38 UDPTL NAT (Yes/No), T.38 UDPTL MAXDATAGRAM, Fax Detect (Yes/No), Trust RPID/PAI (Yes/No), Send Private CallerID Information (Yes/No), Match Inbound Authentication (Default), Inband Progress (Yes/No), Direct Media (Yes/No), Rewrite Contact (Yes/No), RTP Symmetric (Yes/No), Media Encryption (None), Force rport (Yes/No), and Message Context.

Field	Value
T.38 UDPTL NAT	Yes No
T.38 UDPTL MAXDATAGRAM	
Fax Detect	Yes No
Trust RPID/PAI	Yes No
Send RPID/PAI	No Send Remote-Party-ID header Send P-Asserted-Identity header Both
Send Private CallerID Information	Yes No
Match Inbound Authentication	Default
Inband Progress	Yes No
Direct Media	Yes No
Rewrite Contact	Yes No
RTP Symmetric	Yes No
Media Encryption	None
Force rport	Yes No
Message Context	

Figure 6: Asterisk Trunk Continuation

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

Enable Ulaw

Click Submit

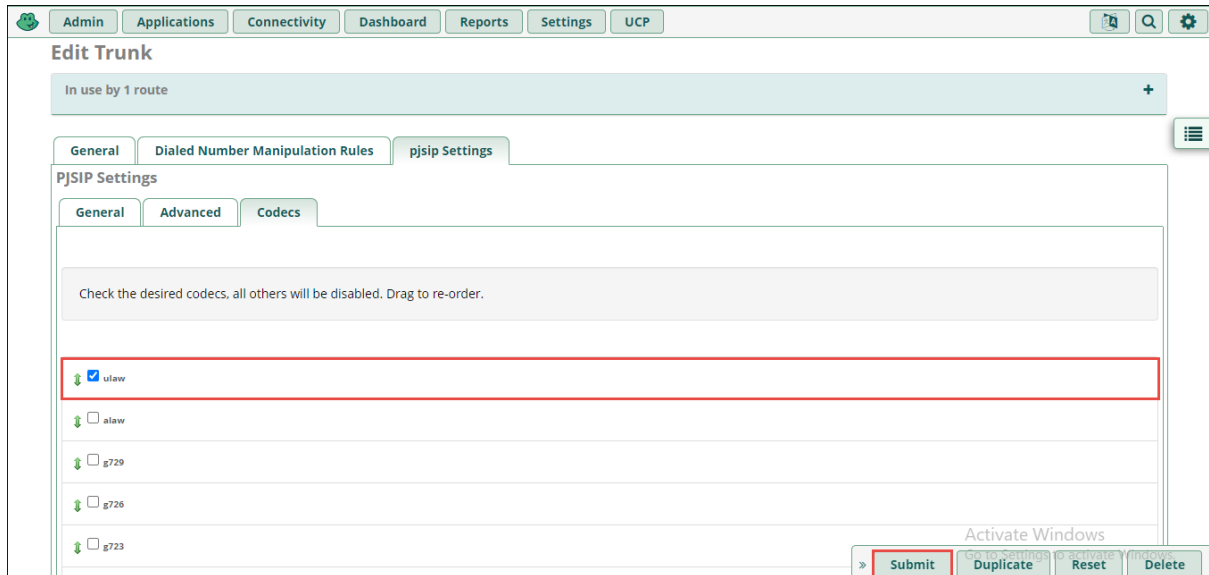


Figure 7: Asterisk Trunk Continuation

4.2.3 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

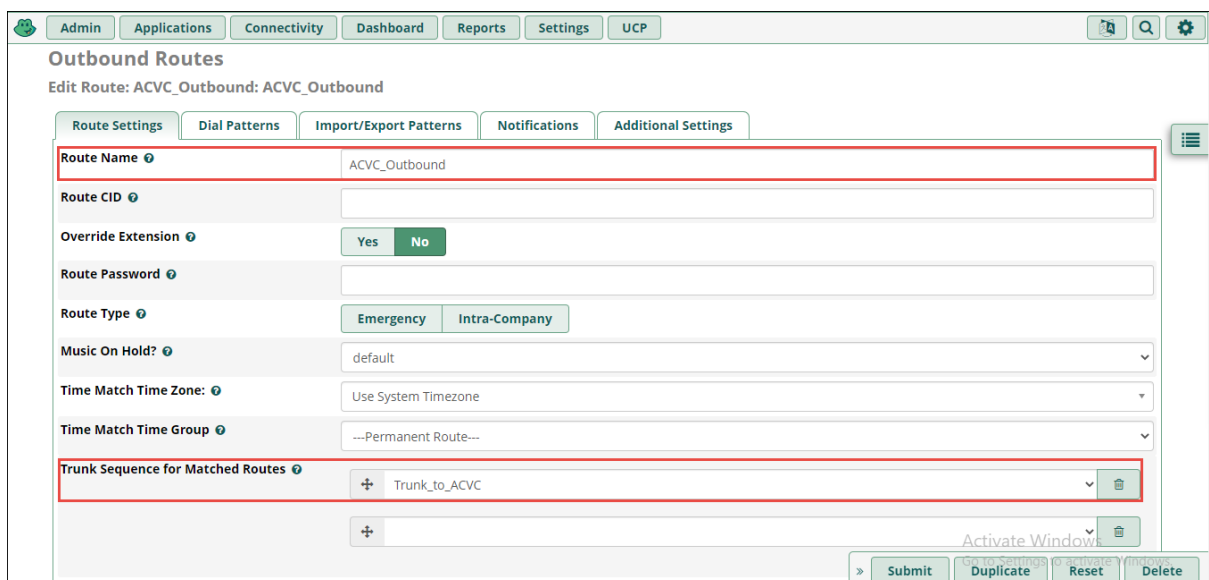


Figure 8: Asterisk Outbound Route

Navigate to **Dial Patterns**

Match Pattern: 214XXXXXXX

Click Submit

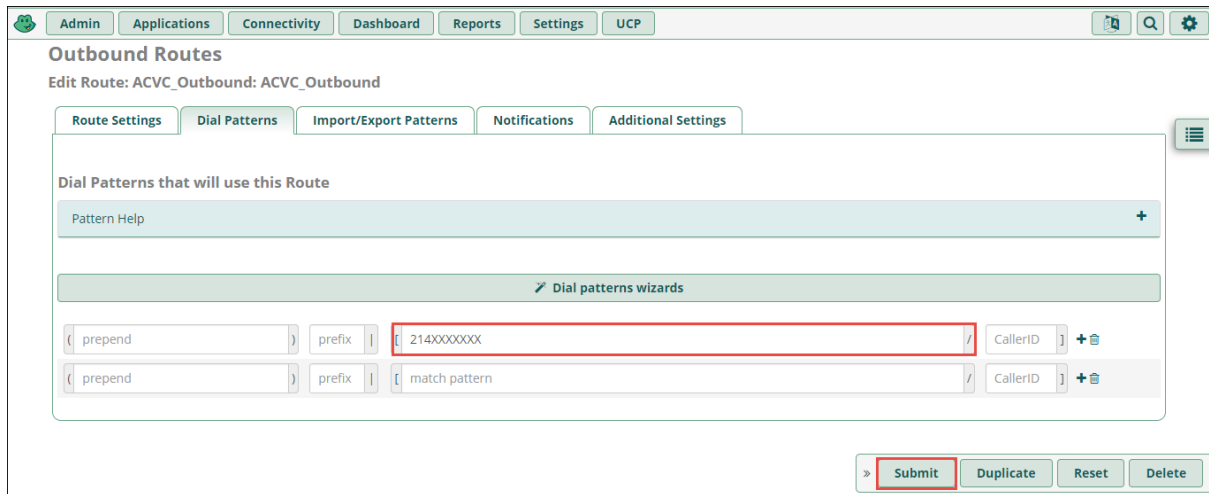


Figure 9: Asterisk Outbound Route Continuation

4.2.4 Inbound Routes

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

DID Number: +191XXXXXXX

Set Destination: Select Extensions/91XXXXXXX

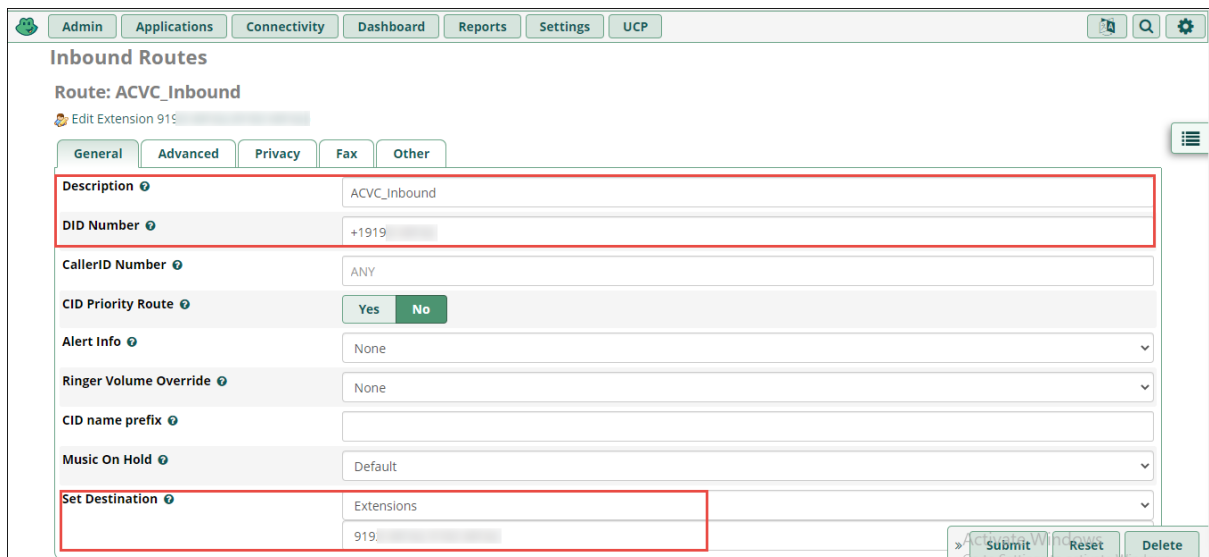


Figure 10: Asterisk Inbound Route

Create another Inbound Route with

DID Number: +132XXXXXXX

Set Destination: Select Extensions/32XXXXXXX

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

DID	CID	Description	Destination	Actions
+1325	Any	ACVC2_inbound	Extensions: 325;	
+1919	Any	ACVC1_inbound	Extensions: 919;	

Figure 11: Inbound Routes List

4.2.5 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

Add PJSIP Extension 919

General | Voicemail | Find Me/Follow Me | Advanced | Pin Sets | Other

— Add Extension

This device uses PJSIP technology listening on Port 5060 (UDP)

User Extension 919

Display Name

Outbound CID 919

Emergency CID

Secret

— Language

Language Code English

— User Manager Settings

Select User Directory: PBX Internal Directory

Submit Reset

Figure 12: Asterisk Extension

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

	Extension	Name	CW	DND	FM/FM	CF	CFB	CFU	Type	Actions
<input type="checkbox"/>	325	325	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	pjsip	
<input type="checkbox"/>	919	919	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	virtual	

Figure 13: Asterisk Extensions List

4.3 Oracle 4600 ESBC Configuration

4.3.1 Oracle 4600 ESBC Version

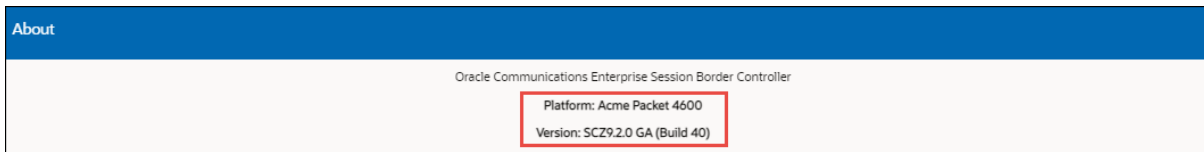


Figure 14: Oracle 4600 ESBC Version

4.3.2 Physical Interface

The following Physical Interfaces are created in Oracle 4600 ESBC

Navigate to **Configuration** → **System** → **Phy-interface**

- 1) s0p0 - Interface towards Amazon Chime SDK Voice Connector to fork the RTP for recording
- 2) s0p1 – Interface towards FreePBX Asterisk

4.3.2.1 Interface towards Amazon Chime SDK Voice Connector

- **Name:** Enter a Name for the interface
- **Operation Type:** Media

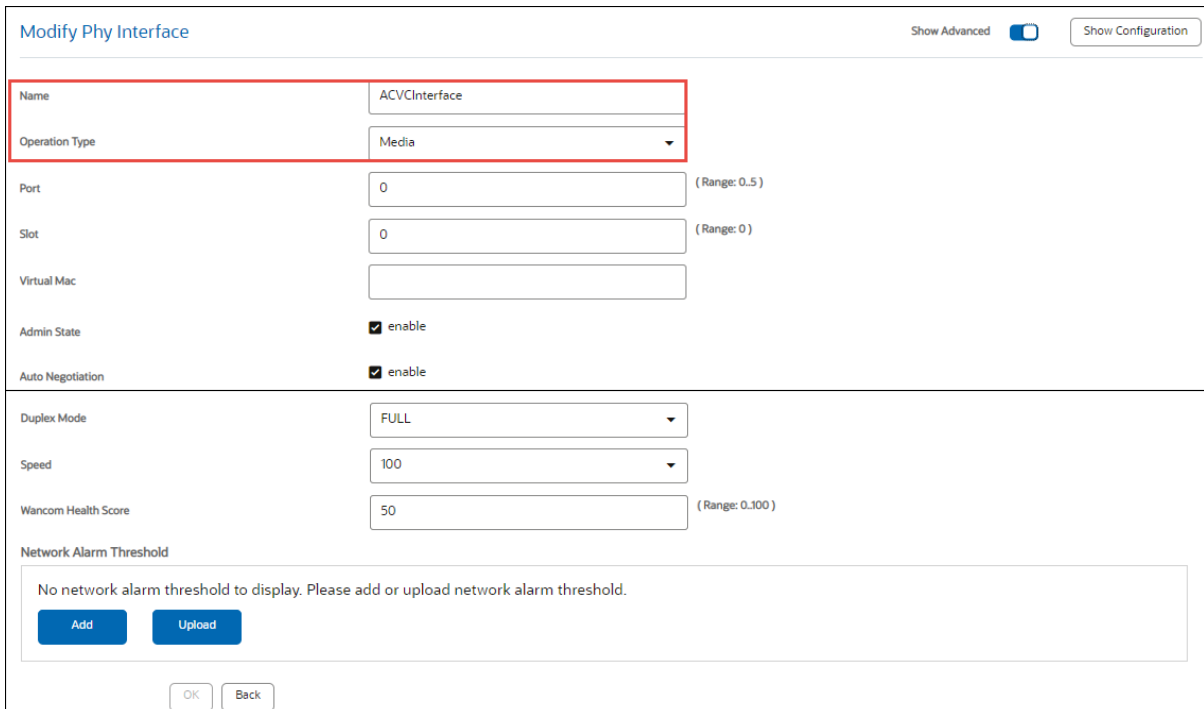


Figure 15: Oracle 4600 Physical Interface Amazon Chime SDK Voice Connector

4.3.2.2 Interface towards FreePBX Asterisk

The screenshot shows the 'Modify Phy Interface' configuration page. At the top right, there are 'Show Advanced' (with a toggle switch) and 'Show Configuration' buttons. The main configuration area is divided into several sections:

- Name:** AsteriskInterface
- Operation Type:** Media (dropdown menu)
- Port:** 1 (Range: 0.5)
- Slot:** 0 (Range: 0)
- Virtual Mac:** (empty text box)
- Admin State:** enable
- Auto Negotiation:** enable
- Duplex Mode:** FULL (dropdown menu)
- Speed:** 100 (dropdown menu)
- Wancom Health Score:** 50 (Range: 0.100)

Below these fields is a 'Network Alarm Threshold' section with a message: 'No network alarm threshold to display. Please add or upload network alarm threshold.' and two buttons: 'Add' and 'Upload'.

At the bottom of the form are 'OK' and 'Back' buttons.

Figure 16: Oracle 4600 Physical Interface Asterisk

4.3.3 Network Interface

The following Network Interfaces are created in Oracle 4600 ESBC

Navigate to **Configuration** → **System** → **Network-interface**

- 1) s0p0 - Assign an IP address to the Interface towards Amazon Chime SDK Voice Connector to fork the RTP for recording
- 2) s0p1 – Assign an IP address to the Interface towards FreePBX Asterisk

4.3.3.1 Network Interface towards Amazon Chime SDK Voice Connector

Name: Select the Physical Interface created for Amazon Chime SDK Voice Connector

IP Address: Enter the IP Address for the Network Interface

Network: Enter the Netmask for the IP Address

Gateway: Enter the Gateway IP Address

DNS IP Primary: Enter the DNS IP

DNS Domain: Enter the Outbound Hostname from Amazon Chime SDK Voice Connector

The screenshot shows the 'Modify Network Interface' configuration page. The interface name is 'ACVInterface'. The IP Address is '192.65.255.1', the Netmask is '255.255.255.0', and the Gateway is '192.65.255.1'. The 'Gw Heartbeat' section is expanded, showing 'State' as 'enable', 'Heartbeat' as '0', 'Retry Count' as '0', 'Retry Timeout' as '1', and 'Health Score' as '0'. The 'DNS IP Primary' is '8.8.8.8'. There are also fields for 'DNS IP Backup1' and 'DNS IP Backup2'.

Name	ACVInterface
Sub Port Id	0 (Range: 0..4095)
Description	Towards ACVC
Hostname	
IP Address	192.65.255.1
Pri Utility Addr	
Sec Utility Addr	
Netmask	255.255.255.0
Gateway	192.65.255.1
Gw Heartbeat	
State	<input checked="" type="checkbox"/> enable
Heartbeat	0 (Range: 0..65535)
Retry Count	0 (Range: 0..65535)
Retry Timeout	1 (Range: 1..65535)
Health Score	0 (Range: 0..100)
DNS IP Primary	8.8.8.8
DNS IP Backup1	
DNS IP Backup2	

Figure 17: Oracle 4600 Network Interface Amazon Chime SDK Voice Connector

DNS Domain	fa2e5s2	
DNS Timeout	11	(Range: 1.999999999)
DNS Max Ttl	86400	(Range: 30..2073600)
Signalling Mtu	0	(Range: 0.576..4096)
HIP IP List		
ICMP Address		
SSH Address		
Tunnel Config	No tunnel config to display. Please add. <input type="button" value="Add"/>	
		Activate Windows Go to Settings to activate Windows.
<input type="button" value="OK"/> <input type="button" value="Back"/>		

Figure 18: Oracle 4600 Network Interface Amazon Chime SDK Voice Connector Continuation

4.3.3.2 Network Interface towards FreePBX Asterisk

Name: Select the Physical Interface created for FreePBX Asterisk

IP Address: Enter the IP Address for the Network Interface

Network: Enter the Netmask for the IP Address

Gateway: Enter the Gateway IP Address

DNS IP Primary: Enter the DNS IP Address

DNS Domain: Enter the DNS Domain

ICMP address: Enter the IP address of the Network Interface

Modify Network Interface		Show Advanced <input checked="" type="checkbox"/> <input type="button" value="Show Configuration"/>
Name	AsteriskInterface	
Sub Port Id	0	(Range: 0..4095)
Description	Towards Asterisk	
Hostname		
IP Address	10.64	
Pri Utility Addr		
Sec Utility Addr		
Netmask	255.25	

Figure 19: Oracle 4600 Network Interface Asterisk

Gateway	10.64
▼ Gw Heartbeat	
State	<input type="checkbox"/> enable
Heartbeat	<input type="text" value="0"/> (Range: 0..65535)
Retry Count	<input type="text" value="0"/> (Range: 0..65535)
Retry Timeout	<input type="text" value="1"/> (Range: 1..65535)
Health Score	<input type="text" value="0"/> (Range: 0..100)
DNS IP Primary	10.87
DNS IP Backup1	<input type="text"/>
DNS IP Backup2	<input type="text"/>
DNS Domain	.com
DNS Timeout	<input type="text" value="11"/> (Range: 1..999999999)
DNS Max Ttl	<input type="text" value="86400"/> (Range: 30..2073600)
Signaling Mtu	<input type="text" value="0"/> (Range: 0..576..4096)
HIP IP List	<input type="text"/>
ICMP Address	10.64
SSH Address	<input type="text"/>
Tunnel Config	
No tunnel config to display. Please add.	
<input type="button" value="Add"/>	
<div style="text-align: right;"> Activate Windows Go to Settings to activate Windows. </div>	
<input type="button" value="OK"/> <input type="button" value="Back"/>	

Figure 20: Oracle 4600 Network Interface Asterisk Continuation

4.3.4 Media Configuration

4.3.4.1 Codec Policy

Navigate to **Configuration** → **Media-Manager** → **Codec-Policy**

1. **Name:** Enter the Name for Codec Policy
2. **Allow Codecs:** Select the Codec and DTMF methods (PCMU, telephone-event)

The screenshot shows the 'Modify Codec Policy' configuration page. The 'Name' field is highlighted with a red box and contains 'G711U'. The 'Allow Codecs' field is also highlighted with a red box and contains 'PCMU' and 'telephone-event'. Other fields include 'Add Codecs On Egress', 'Order Codecs', 'Packetization Time' (20), 'Force Pttime' (checkbox), 'Secure Dtmf Cancellation' (checkbox), 'Dtmf In Audio' (dropdown menu), 'Tone Detection', 'Tone Detect Renegotiate Timer' (500), 'Reverse Fax Tone Detection Reininvite' (checkbox), 'Fax Single M Line' (dropdown menu), and 'EvrC Tty Baudot Transcode' (checkbox). The page also features 'Show Advanced' and 'Show Configuration' buttons, and an 'Activate Windows' watermark.

Figure 21: Oracle 4600 Codec Policy

4.3.4.2 Realm Configuration

Navigate to **Configuration → Media-Manager → Realm-Config** and add two realm configs for

1. Amazon Chime SDK Voice Connector Recording
2. FreePBX Asterisk

Identifier: Enter the identifier name

Description: Enter the description

Network Interface: Select the network interface created for Amazon Chime SDK Voice Connector

Codec Policy: Select the codec policy created

Out Manipulation id: Select the manipulation id (Refer [Section 4.3.5.2](#) for SIP Manipulation Configuration)

Media Sec Policy: Select the media policy

4.3.4.2.1 Realm for Amazon Chime SDK Voice Connector

Modify Realm Config Show Advanced Show Configuration

Identifier	ACVCRecdServer
Description	ACVCRecdServer
Addr Prefix	0.0.0.0
Network Interfaces	ACVCInterface:0.4 x
Media Realm List	
Mm In Realm	<input checked="" type="checkbox"/> enable
Mm In Network	<input checked="" type="checkbox"/> enable
Mm Same Ip	<input checked="" type="checkbox"/> enable
QoS Enable	<input type="checkbox"/> enable
Max Bandwidth	0 (Range: 0-999999999)
Max Priority Bandwidth	0 (Range: 0-999999999)
Parent Realm	
DNS Realm	
Media Policy	
Nsep Media Policy	
Media Sec Policy	SRTP
RTCP Mux	<input type="checkbox"/> enable

Figure 22: Oracle 4600 Realm Amazon Chime SDK Voice Connector

Ice Profile	<input type="text"/>	▼
Teams Fqdn	<input type="text"/>	
Teams Fqdn In Uri	<input type="checkbox"/>	enable
SDP Inactive Only	<input type="checkbox"/>	enable
DTLS Srtp Profile	<input type="text"/>	▼
Srtp Msm Passthrough	<input type="checkbox"/>	enable
Class Profile	<input type="text"/>	▼
In Session Translations		
No in session translation list to display. Please add.		
<input type="button" value="Add"/>		
No out session translation list to display. Please add.		
<input type="button" value="Add"/>		
In ManipulationId	<input type="text"/>	▼
Out ManipulationId	AmazonManipulation	▼
Average Rate Limit	<input type="text" value="0"/>	(Range: 0..4294967295)
Access Control Trust Level	none	▼
Invalid Signal Threshold	<input type="text" value="0"/>	(Range: 0..4294967295)
Maximum Signal Threshold	<input type="text" value="0"/>	(Range: 0..4294967295)
Untrusted Signal Threshold	<input type="text" value="0"/>	(Range: 0..4294967295)
Nat Trust Threshold	<input type="text" value="0"/>	(Range: 0..65535)
Max Endpoints Per Nat	<input type="text" value="0"/>	(Range: 0..65535)
Nat Invalid Message Threshold	<input type="text" value="0"/>	(Range: 0..65535)
Wait Time For Invalid Register		
	<input type="text" value="0"/>	(Range: 0..4..300)
Deny Period	<input type="text" value="30"/>	(Range: 0..4294967295)
Session Max Life Limit	<input type="text" value="0"/>	
Untrust Cac Failure Threshold	<input type="text" value="0"/>	(Range: 0..4294967295)
Subscription Id Type	END_USER_NONE	▼
Trunk Context	<input type="text"/>	
Early Media Allow	<input type="text"/>	
Enforcement Profile	<input type="text"/>	▼
Additional Prefixes	<input type="text"/>	
Restricted Latching	none	▼
Options	<input type="text"/>	
SPL Options	<input type="text"/>	
Delay Media Update	<input type="checkbox"/>	enable
Refer Call Transfer	enabled	▼
Hold Refer Reinwrite	<input checked="" type="checkbox"/>	enable
Refer Notify Provisional	none	▼
Dyn Refer Term	<input type="checkbox"/>	enable
Codec Policy	G711U	▼
Codec ManIP In Realm	<input type="checkbox"/>	enable
Codec ManIP In Network	<input checked="" type="checkbox"/>	enable
RTCP Policy	<input type="text"/>	▼
Constraint Name	<input type="text"/>	▼

Figure 23: Oracle 4600 Realm Amazon Chime SDK Voice Connector Continuation

Session Recording Server	<input type="text"/>	
Session Recording Required	<input type="checkbox"/> enable	
SIP Profile	<input type="text"/>	
Flow Time Limit	<input type="text" value="-1"/>	(Range: -1.2147483547)
Initial Guard Timer	<input type="text" value="-1"/>	(Range: -1.2147483547)
Subsq Guard Timer	<input type="text" value="-1"/>	(Range: -1.2147483547)
TCP Flow Time Limit	<input type="text" value="-1"/>	(Range: -1.2147483547)
TCP Initial Guard Timer	<input type="text" value="-1"/>	(Range: -1.2147483547)
TCP Subsq Guard Timer	<input type="text" value="-1"/>	(Range: -1.2147483547)
SIP Isup Profile	<input type="text"/>	
QoS Constraint	<input type="text"/>	
TCP Media Profile	<input type="text"/>	
Monitoring Filters	<input type="text"/>	
Node Functionality	<input type="text"/>	
Default Location String	<input type="text"/>	
Alt Family Realm	<input type="text"/>	
Pref Addr Type	<input type="text" value="none"/>	
Sm Icsi Match For Invite	<input type="text"/>	
Sm Icsi Match For Message	<input type="text"/>	
Ringback Trigger	<input type="text" value="none"/>	
Ringback File	<input type="text"/>	
Merge Early Dialogs	<input type="checkbox"/> enable	
User Site	<input type="text"/>	
Srvcc Trfo	<input type="text"/>	
Feature Trfo	<input type="text"/>	
Auth Attribute	<div style="border: 1px solid #ccc; padding: 5px;"> <p>No auth attributes to display. Please add.</p> <p><input type="button" value="Add"/></p> </div>	
Fqdn Hostname	<input type="text"/>	
Fqdn Hostname In Header	<input type="text"/>	
P Asserted Identity	<input type="text"/>	
P Asserted Identity For	<input type="text"/>	
Steering Pool Threshold	<input type="text" value="0"/>	(Range: 0.100)
Steering Pool Lower Threshold	<input type="text" value="70"/>	(Range: 1.95)
Steering Pool Alarm Monitoring Time	<input type="text" value="15"/>	(Range: 5.600)
<input type="button" value="OK"/> <input type="button" value="Back"/>		<p>Activate Windows Go to Settings to activate Windows.</p>

Figure 24: Oracle 4600 Realm Amazon Chime SDK Voice Connector Continuation

4.3.4.2.2 Realm for FreePBX Asterisk

Modify Realm Config Show Advanced Show Configuration

Identifier	Asterisk
Description	Asterisk
Addr Prefix	0.0.0.0
Network Interfaces	AsteriskInterface:0.4 x
Media Realm List	
Mm In Realm	<input checked="" type="checkbox"/> enable
Mm In Network	<input checked="" type="checkbox"/> enable
Mm Same Ip	<input checked="" type="checkbox"/> enable
QoS Enable	<input type="checkbox"/> enable
Max Bandwidth	0 (Range: 0.999999999)
Max Priority Bandwidth	0 (Range: 0.999999999)
Parent Realm	
DNS Realm	
Media Policy	
Nsep Media Policy	
Media Sec Policy	RTP
RTCP Mux	<input type="checkbox"/> enable
Ice Profile	
Teams Fqdn	
Teams Fqdn In Uri	<input type="checkbox"/> enable
SDP Inactive Only	<input type="checkbox"/> enable
DTLS Srtp Profile	
Srtp Msm Passthrough	<input type="checkbox"/> enable
Class Profile	
In Session Translations	No in session translation list to display. Please add. <input type="button" value="Add"/>
Out Session Translations	No out session translation list to display. Please add. <input type="button" value="Add"/>

Figure 25: Oracle 4600 Realm Asterisk

In Manipulationid	<input type="text"/>	
Out Manipulationid	AsteriskManipulation	
Average Rate Limit	<input type="text" value="0"/>	(Range: 0..4294967295)
Access Control Trust Level	none	
Invalid Signal Threshold	<input type="text" value="0"/>	(Range: 0..4294967295)
Maximum Signal Threshold	<input type="text" value="0"/>	(Range: 0..4294967295)
Untrusted Signal Threshold	<input type="text" value="0"/>	(Range: 0..4294967295)
Nat Trust Threshold	<input type="text" value="0"/>	(Range: 0..65535)
Max Endpoints Per Nat	<input type="text" value="0"/>	(Range: 0..65535)
Nat Invalid Message Threshold	<input type="text" value="0"/>	(Range: 0..65535)
Activate Windows		
Deny Period	<input type="text" value="30"/>	(Range: 0..4294967295)
Session Max Life Limit	<input type="text" value="0"/>	
Untrust Cac Failure Threshold	<input type="text" value="0"/>	(Range: 0..4294967295)
Subscription Id Type	END_USER_NONE	
Trunk Context	<input type="text"/>	
Early Media Allow	<input type="text"/>	
Enforcement Profile	<input type="text"/>	
Additional Prefixes	<input type="text"/>	
Restricted Latching	none	
Options	<input type="text"/>	
Activate Windows		
SPL Options	<input type="text"/>	
Delay Media Update	<input type="checkbox"/> enable	
Refer Call Transfer	enabled	
Hold Refer Reinvite	<input checked="" type="checkbox"/> enable	
Refer Notify Provisional	none	
Dyn Refer Term	<input type="checkbox"/> enable	
Codec Policy	G711U	
Codec ManIP In Realm	<input type="checkbox"/> enable	
Codec ManIP In Network	<input checked="" type="checkbox"/> enable	
RTCP Policy	<input type="text"/>	
Constraint Name	<input type="text"/>	

Figure 26: Oracle 4600 Realm Asterisk Continuation

Session Recording Server: Select the Session Recording Server created for SIPREC call recording to Amazon Chime SDK Voice Connector (Refer to [Section 4.3.5.5](#) for Session Recording Server Configuration)

Session Recording Server		SRSACVC x
Session Recording Required	<input type="checkbox"/>	enable
SIP Profile	<input type="text"/>	
Flow Time Limit	<input type="text" value="-1"/>	(Range: -1.2147483647)
Initial Guard Timer	<input type="text" value="-1"/>	(Range: -1.2147483647)
Subsq Guard Timer	<input type="text" value="-1"/>	(Range: -1.2147483647)
TCP Flow Time Limit	<input type="text" value="-1"/>	(Range: -1.2147483647)
TCP Initial Guard Timer	<input type="text" value="-1"/>	(Range: -1.2147483647)
TCP Subsq Guard Timer	<input type="text" value="-1"/>	(Range: -1.2147483647)
SIP Isup Profile	<input type="text"/>	
QoS Constraint	<input type="text"/>	
TCP Media Profile	<input type="text"/>	
Monitoring Filters	<input type="text"/>	
Node Functionality	<input type="text"/>	
Default Location String	<input type="text"/>	
Alt Family Realm	<input type="text"/>	
Pref Addr Type	none	
Sm Icsi Match For Invite	<input type="text"/>	
Sm Icsi Match For Message	<input type="text"/>	
Ringback Trigger	none	
Ringback File	<input type="text"/>	
Merge Early Dialogs	<input type="checkbox"/>	enable
User Site	<input type="text"/>	
Srvcc Trfo	<input type="text"/>	
Feature Trfo	<input type="text"/>	
Auth Attribute	<div style="border: 1px solid #ccc; padding: 5px;"> <p>No auth attributes to display. Please add.</p> <p><input type="button" value="Add"/></p> </div>	
Fqdn Hostname	<input type="text"/>	
Fqdn Hostname In Header	<input type="text"/>	
P Asserted Identity	<input type="text"/>	
P Asserted Identity For	<input type="text"/>	
Steering Pool Threshold	<input type="text" value="0"/>	(Range: 0.100)
Steering Pool Lower Threshold	<input type="text" value="70"/>	(Range: 1.95)
Steering Pool Alarm Monitoring Time	<input type="text" value="15"/>	(Range: 5.600)
<input type="button" value="OK"/> <input type="button" value="Back"/>		<p>Activate Windows Go to Settings to activate Windows.</p>

Figure 27: Oracle 4600 Realm Asterisk Continuation

4.3.4.3 Steering Pool

Navigate to **Configuration → Media-Manager → Steering-Pool** and add two Steering Pool for

1. Amazon Chime SDK Voice Connector Recording
2. FreePBX Asterisk

IP Address: Enter the IP Address

Realm Id: Select the Realm Id

Network Interface: Select the Network Interface

4.3.4.3.1 Steering Pool for Amazon Chime SDK Voice Connector

Modify Steering Pool		Show Configuration
IP Address	192.65	
Start Port	20000	(Range: 0..65535)
End Port	39999	(Range: 0..65535)
Realm ID	ACVCRecdServer	▼
Network Interface	ACVCInterface:0.4	▼
Port Allocation Strategy	mixed	▼

Figure 28: Oracle 4600 Steering Pool Amazon Chime SDK Voice Connector

4.3.4.3.2 Steering Pool for FreePBX Asterisk

Modify Steering Pool		Show Configuration
IP Address	10.64	
Start Port	20000	(Range: 0..65535)
End Port	39999	(Range: 0..65535)
Realm ID	Asterisk	▼
Network Interface	AsteriskInterface:0.4	▼
Port Allocation Strategy	mixed	▼

Figure 29: Oracle 4600 Steering Pool Asterisk

4.3.5 SIP Configuration

4.3.5.1 Media Profile

Navigate to **Configuration** → **Session-Router** → **Media-Profile**

- Media Profile is created to add Codec and DTMF events when Early offer is forced from Oracle 4600 ESBC.
- Two Media Profiles are created.
 - Media Profile for Codec PCMU
 - Media Profile for DTMF- telephone-event

Name: Enter the name for the media profile

Subname: Enter the subname for the media profile

Media Type: Audio

Payload Type: 0,101

Transport: RTP/AVP

4.3.5.1.1 Media Profile for Codec PCMU

The screenshot shows the 'Modify Media Profile' configuration page. The form is titled 'Modify Media Profile' and has a 'Show Configuration' button. The form contains the following fields:

Name	PCMU
Subname	64K
Media Type	audio
Payload Type	0,101
Transport	RTP/AVP
Clock Rate	0 (Range: 0.4294967295)
Req Bandwidth	0 (Range: 0.999999999)
Frames Per Packet	0 (Range: 0.256)
Parameters	
As Bandwidth	0 (Range: 0.4294967295)

At the bottom of the form, there are 'OK' and 'Back' buttons. A watermark for 'Activate Windows' is visible in the bottom right corner.

Figure 30: Oracle 4600 Media Profile PCMU

4.3.5.1.2 Media Profile for DTMF-telephone-event

Modify Media Profile Show Advanced Show Configuration

Name	telephone-event
Subname	telephone-event
Media Type	audio
Payload Type	101
Transport	RTP/AVP
Clock Rate	0 (Range: 0.4294967295)
Req Bandwidth	0 (Range: 0.999999999)
Frames Per Packet	0 (Range: 0.256)
Parameters	
As Bandwidth	0 (Range: 0.4294967295)

OK Back

Activate Windows
Go to Settings to activate Windows.

Figure 31: Oracle 4600 Media Profile Telephone Event

4.3.5.2 SIP Manipulation

- Navigate to **Configuration** → **Session-Router** → **SIP-Manipulation**

4.3.5.2.1 SIP Manipulations for Amazon Chime SDK Voice Connector Trunk

The following are the SIP manipulations used to modify the headers and the SDP attributes based on the interaction between Oracle AP4600 and Amazon Chime SDK Voice Connector.

sip-manipulation

name	AmazonManipulation
description	AmazonManipulation

header-rule (This manipulation is to replace the IP address in the From header of the requests like Invite & Options with the IP address of ESBC's Network interface towards the Amazon Chime SDK Voice Connector)

name	fromhost
header-name	from
action	manipulate
msg-type	request
methods	INVITE,OPTIONS
element-rule	
name	fromhost
type	uri-host
action	replace
match-val-type	any
new-value	\$LOCAL_IP

header-rule (This manipulation is to replace the IP address in the To header in the Invite request with the FQDN of the Outbound Hostname from Amazon Chime SDK Voice Connector)

name	ToHost
header-name	to
action	manipulate
msg-type	request
methods	INVITE
element-rule	
name	Tohost
type	uri-host
action	replace
match-val-type	ip
new-value	<Outbound_HostName from Amazon Chime SDK Voice Connector>

header-rule (This manipulation is to replace the IP address in the P-Asserted-Identity with the FQDN of the Outbound Hostname from Amazon Chime SDK Voice Connector)

name	pai
header-name	P-Asserted-Identity
action	manipulate
msg-type	request
element-rule	
name	pai
type	uri-host
action	replace
match-val-type	ip
new-value	<Outbound_HostName from Amazon Chime SDK Voice Connector>

header-rule (This manipulation is to append +1 to the DID in the P-asserted identity)

name	paiuser
header-name	P-Asserted-Identity
action	manipulate
comparison-type	pattern-rule
methods	INVITE
element-rule	
name	paiheaderuser
type	uri-user
action	replace
comparison-type	pattern-rule
match-value	(^919.*)
new-value	" +1" + \$ORIGINAL

header-rule (This manipulation is to replace the IP address in the P-asserted identity header with IP address of the ESBC's network interface towards Amazon Chime SDK Voice Connector)

name	paireponse
header-name	P-Asserted-Identity
action	manipulate
msg-type	reply
element-rule	
name	pai
type	uri-host
action	replace
match-val-type	ip
new-value	\$LOCAL_IP

header-rule (This manipulation is to replace the owner name in the SDP with the name "OracleACME")

name	modsdpowner
header-name	Content-type
action	manipulate
element-rule	
name	changeowner
parameter-name	application/sdp
type	mime
action	find-replace-all
match-value	(.*)
new-value	OracleACME

header-rule (This manipulation is to replace the User agent in the request with the ESBC's name and version)

name	modifyuseragent
header-name	User-Agent
action	manipulate
comparison-type	pattern-rule
msg-type	request
methods	ACK, BYE, INVITE, PRACK, UPDATE
element-rule	
name	modua
type	header-value
action	replace
comparison-type	pattern-rule
match-value	^FPBX(.*)
new-value	OracleE\-ESBC/SCZ920

header-rule (This manipulation is to append +1 to the DID in the From header of the Invite request)

name	fromuser
header-name	from
action	manipulate
comparison-type	pattern-rule
msg-type	request
methods	INVITE
element-rule	
name	fromuser1
type	uri-user
action	replace
comparison-type	pattern-rule
match-value	(^919.*)
new-value	" + 1" + \$ORIGINAL
element-rule	
name	fromuser2

type	uri-user
action	replace
comparison-type	pattern-rule
match-value	(^325.*)
new-value	"+1"+\$ORIGINAL

header-rule (This manipulation is to append +1 to the DID in the To header of the Invite request)

name	ToUser
header-name	To
action	manipulate
comparison-type	pattern-rule
msg-type	request
methods	INVITE
element-rule	
name	Touser1
type	uri-user
action	replace
comparison-type	pattern-rule
match-value	(^214.*)
new-value	"+1"+\$ORIGINAL

header-rule (This manipulation is to append +1 to the DID in the request uri)

name	RequestURI
header-name	request-uri
action	manipulate
comparison-type	pattern-rule
msg-type	request
element-rule	
name	RequestURI1
type	uri-user
action	replace
comparison-type	pattern-rule
match-value	(^214.*)
new-value	"+1"+\$ORIGINAL

4.3.5.2.2 SIP Manipulations for Asterisk Trunk

The following are the SIP manipulations used to modify the headers and the SDP attributes based on the interaction between Asterisk and Oracle AP4600.

sip-manipulation

name AsteriskManipulation
description AsteriskManipulation

header-rule (This manipulation is to replace the IP address in the From Header with IP address of the ESBC's IP address of the network interface towards FreePBX Asterisk)

name	fromhost
header-name	From
action	manipulate
msg-type	request
element-rule	
name	fromhost
type	uri-host
action	replace
new-value	\$LOCAL_IP

header-rule(This manipulation is to replace the IP address in the From Header with IP address of the ESBC's IP address of the FreePBX Asterisk)

name	tohost
header-name	To
action	manipulate
msg-type	request
element-rule	
name	tohost
type	uri-host
action	replace
new-value	\$REMOTE_IP

header-rule (This manipulation is to replace SDP owner name with "OracleACME")

name	modsdpowner
header-name	Content-type
action	manipulate
element-rule	
name	modsdpowner
parameter-name	application/sdp
type	mime
action	find-replace-all
match-value	Sonus_UAC
new-value	OracleACME

header-rule (This manipulation is to replace Useragent name with "Oracle")

name	moduseragent
header-name	User-Agent
action	manipulate
comparison-type	pattern-rule
element-rule	
name	moduseragent
type	header-value
action	replace
comparison-type	pattern-rule
match-value	^VineProx
new-value	Oracle

4.3.5.3 SIP Interface

Navigate to **Configuration** → **Session-Router** → **SIP-Interface** and add two realm SIP Interfaces for

1. Amazon Chime SDK Voice Connector Recording
2. FreePBX Asterisk

Realm Id: Select the Realm Id

Description: Enter the Description

SIP Ports: Enter the Address, Port Number and select the Transport Protocol

TLS Profile: Select the TLS Profile in the SIP Ports for the Transport Protocol TLS.

4.3.5.3.1 SIP Interface – Amazon Chime SDK Voice Connector

The screenshot displays the 'Modify SIP Interface' configuration page. At the top right, there are 'Show Advanced' and 'Show Configuration' buttons. The 'State' section has a checked 'enable' checkbox. The 'Realm ID' dropdown is set to 'ACVRecdServer', and the 'Description' text field also contains 'ACVRecdServer'. Below this is the 'SIP Ports' section, which includes a table with the following data:

Select	Action	Address	Port	Transport Protocol	TLS Profile	Allow Anonymous	Multi Home Addr
<input type="checkbox"/>	⋮	192.65	5061	TLS	Amazon_TLS_Profile	all	

Below the table, there are several configuration fields with their respective ranges:

- Initial Inv Trans Expire: 0 (Range: 0.2147475)
- Session Max Life Limit: 0
- Proxy Mode: (Dropdown menu)
- Redirect Action: (Dropdown menu)
- Nat Traversal: none (Dropdown menu)
- Nat Interval: 30 (Range: 1.999999999)
- TCP Nat Interval: 90 (Range: 0.999999999)
- Registration Caching: enable
- Min Reg Expire: 300 (Range: 0.999999999)
- Registration Interval: 3600 (Range: 1.999999999)

An 'Activate Windows' watermark is visible in the bottom right corner of the screenshot.

Figure 32: Oracle 4600 SIP Interface Amazon Chime SDK Voice Connector

Route To Registrar	<input type="checkbox"/> enable	
Secured Network	<input type="checkbox"/> enable	
Uri Fqdn Domain	<input type="text"/>	
Options	<input type="text"/>	
SPL Options	<input type="text"/>	
Trust Mode	all	
Max Nat Interval	3600	(Range: 0.999999999)
Stop Recurse	401,407	
Port Map Start	0	(Range: 0,025..65535)
Port Map End	0	(Range: 0,025..65535)
In Manipulationid	<input type="text"/>	
Out Manipulationid	<input type="text"/>	
SIP Atcf Feature	<input type="checkbox"/> enable	
Rfc2833 Payload	101	(Range: 96..127)
Rfc2833 Mode	transparent	
Response Map	<input type="text"/>	
Local Response Map	<input type="text"/>	
Sec Agree Feature	<input type="checkbox"/> enable	
Enforcement Profile	<input type="text"/>	
TCP Keepalive	none	

Figure 33: Oracle 4600 SIP Interface Amazon Chime SDK Voice Connector Continuation

Add SDP Invite: Invite (To enable early offer from Oracle 4600 ESBC)

Add SDP Profiles: Choose Media Profiles PCMU and Telephone-events (Refer [Section 4.3.5.1](#) for media profile configuration)

Add SDP Invite	invite
Add SDP In Msg	<input type="text"/>
P Early Media Header	disabled
P Early Media Direction	<input type="text"/>
Add SDP Profiles	<input type="text" value="PCMU:64K x"/> <input type="text" value="TELEPHONE-EVENT:TELEPHONE-EVENT x"/>
Add SDP Profiles In Msg	<input type="text"/>
SIP Profile	<input type="text"/>
SIP Isup Profile	<input type="text"/>
TCP Conn Dereg	0 (Range: 0.999999999)

Figure 34: Oracle 4600 SIP Interface Amazon Chime SDK Voice Connector Continuation

Kpm2833 Iwf On Hairpin	<input type="checkbox"/> enable	
Msrp Delay Egress Bye	<input type="checkbox"/> enable	
Send 380 Response		<input type="text"/>
Pscf Restoration		<input type="text"/>
Session Timer Profile		<input type="text" value=""/>
Session Recording Server		<input type="text"/>
Session Recording Required	<input type="checkbox"/> enable	
Service Tag		<input type="text"/>
Reg Cache Route	<input type="checkbox"/> enable	
Diversion Info Mapping Mode		<input type="text" value="none"/>
Activate Windows		
Atcf Icsi Match		<input type="text"/>
SIP Recursion Policy		<input type="text" value=""/>
Asymmetric Preconditions	<input type="checkbox"/> enable	
Asymmetric Preconditions Mode		<input type="text" value="send-with-nodelay"/>
Sm Icsi Match For Invite		<input type="text"/>
Sm Icsi Match For Message		<input type="text"/>
S8hr Profile		<input type="text" value=""/>
Ringback Trigger		<input type="text" value="none"/>
Ringback File		<input type="text"/>
Fax Continue Session		<input type="text" value="none"/>
NpII Profile		<input type="text" value=""/>
Hist To Div For Cause 380		<input type="text" value="inherit"/>
User Agent		<input type="text"/>
Allow Diff2833 Clock Rate Mode		<input type="text" value="disabled"/>
Activate Windows Go to Settings to activate Windows.		
<input type="button" value="OK"/> <input type="button" value="Back"/>		

Figure 35: Oracle 4600 SIP Interface Amazon Chime SDK Voice Connector Continuation

4.3.5.3.2 SIP Interface FreePBX Asterisk

Modify SIP Interface
Show Advanced
Show Configuration

State enable

Realm ID	Asterisk
Description	Asterisk

SIP Ports

Select	Action	Address	Port	Transport Protocol	TLS Profile	Allow Anonymous	Multi Home Addr
<input type="checkbox"/>	⋮	10.64	5060	UDP		all	

Displaying 1 - 1 of 1

Initial Inv Trans Expire	<input type="text" value="0"/>	(Range: 0.2147473)
Session Max Life Limit	<input type="text" value="0"/>	
Proxy Mode	<input type="text"/>	
Redirect Action	<input type="text"/>	
Nat Traversal	<input type="text" value="none"/>	
Nat Interval	<input type="text" value="30"/>	(Range: 1.999999999)
TCP Nat Interval	<input type="text" value="90"/>	(Range: 0.999999999)
Registration Caching	<input type="checkbox"/> enable	
Min Reg Expire	<input type="text" value="300"/>	(Range: 0.999999999)
Registration Interval	<input type="text" value="3600"/>	(Range: 1.999999999)

Route To Registrar	<input type="checkbox"/> enable	
Secured Network	<input type="checkbox"/> enable	
Uri Fqdn Domain	<input type="text"/>	
Options	<input type="text"/>	
SPL Options	<input type="text"/>	
Trust Mode	<input type="text" value="all"/>	
Max Nat Interval	<input type="text" value="3600"/>	(Range: 0.999999999)
Stop Recurse	<input type="text" value="401,407"/>	
Port Map Start	<input type="text" value="0"/>	(Range: 0,025..65535)
Port Map End	<input type="text" value="0"/>	(Range: 0,025..65535)

Figure 36: Oracle 4600 SIP Interface Asterisk

In Manipulationid	<input type="text"/>	
Out Manipulationid	<input type="text"/>	
SIP Atcf Feature	<input type="checkbox"/> enable	
Rfc2833 Payload	<input type="text" value="101"/>	(Range: 96..127)
Rfc2833 Mode	<input type="text" value="transparent"/>	
Response Map	<input type="text"/>	
Local Response Map	<input type="text"/>	
Sec Agree Feature	<input type="checkbox"/> enable	
Enforcement Profile	<input type="text"/>	
TCP Keepalive	<input type="text" value="none"/>	
Activate Windows		
Add SDP Invite	<input type="text" value="disabled"/>	
Add SDP In Msg	<input type="text"/>	
P Early Media Header	<input type="text" value="disabled"/>	
P Early Media Direction	<input type="text"/>	
Add SDP Profiles	<input type="text"/>	
Add SDP Profiles In Msg	<input type="text"/>	
SIP Profile	<input type="text"/>	
SIP Isup Profile	<input type="text"/>	
TCP Conn Dereg	<input type="text" value="0"/>	(Range: 0..999999999)
Kpml Interworking	<input type="checkbox"/> enable	
Activate Windows		
Kpml Interworking	<input type="checkbox"/> enable	
Kpml2833 Iwf On Hairpin	<input type="checkbox"/> enable	
Msrp Delay Egress Bye	<input type="checkbox"/> enable	
Send 380 Response	<input type="text"/>	
Pcsf Restoration	<input type="text"/>	
Session Timer Profile	<input type="text"/>	
Session Recording Server	<input type="text"/>	
Session Recording Required	<input type="checkbox"/> enable	
Service Tag	<input type="text"/>	
Reg Cache Route	<input type="checkbox"/> enable	

Figure 37: Oracle 4600 SIP Interface Asterisk Continuation

Diversion Info Mapping Mode	none
Atcf Icsi Match	
SIP Recursion Policy	
Asymmetric Preconditions	<input type="checkbox"/> enable
Asymmetric Preconditions Mode	send-with-nodelay
Sm Icsi Match For Invite	
Sm Icsi Match For Message	
S8hr Profile	
Ringback Trigger	none
Ringback File	
Fax Continue Session	none
NpII Profile	
Hist To Div For Cause 380	inherit
User Agent	
Allow Diff2833 Clock Rate Mode	disabled

OK Back

Activate Windows
Go to Settings to activate Windows.

Figure 38: Oracle 4600 SIP Interface Asterisk Continuation

4.3.5.4 Session-Agent

Navigate to **Configuration** → **Session-Router** → **Session-Agent** and add two Session-Agents

1. Amazon Chime SDK Voice Connector Recording
2. FreePBX Asterisk

Hostname: Enter the hostname

IP Address: Enter the IP address

Port: Enter the Port Number

APP Protocol: SIP

Transport Method: Select the Transport Method

Realm Id: Select the Realm Id

Codec Policy: Select the Codec Policy

4.3.5.4.1 Session-Agent-Amazon Chime SDK Voice Connector

Modify Session Agent Show Advanced Show Configuration

Hostname	fa2e5s3tr
IP Address	
Port	5061 (Range: 0,1025..65535)
State	<input checked="" type="checkbox"/> enable
App Protocol	SIP
App Type	
Transport Method	StaticTLS
Realm ID	ACVCRcdServer
Egress Realm ID	
Description	

Match Identifier

No match identifier to display. Please add.

Associated Agents

Constraints enable

Max Sessions	0 (Range: 0..999999999)
Max Inbound Sessions	0 (Range: 0..999999999)
Max Outbound Sessions	0 (Range: 0..999999999)
Max Burst Rate	0 (Range: 0..999999999)
Max Inbound Burst Rate	0 (Range: 0..999999999)
Max Outbound Burst Rate	0 (Range: 0..999999999)

Figure 39: Oracle 4600 Session Agent Amazon Chime SDK Voice Connector

Max Sustain Rate	<input type="text" value="0"/>	(Range: 0.999999999)
Max Inbound Sustain Rate	<input type="text" value="0"/>	(Range: 0.999999999)
Max Outbound Sustain Rate	<input type="text" value="0"/>	(Range: 0.999999999)
Min Asr	<input type="text" value="0"/>	(Range: 0.100)
Cac Trap Threshold	<input type="text" value="0"/>	(Range: 0.99)
Session Max Life Limit	<input type="text" value="0"/>	
Time To Resume	<input type="text" value="0"/>	(Range: 0.999999999)
In Service Period	<input type="text" value="0"/>	(Range: 0.999999999)
Burst Rate Window	<input type="text" value="0"/>	(Range: 0.999999999)
Sustain Rate Window	<input type="text" value="0"/>	(Range: 0.999999999)
Proxy Mode	<input type="text"/>	
Redirect Action	<input type="text"/>	
Loose Routing	<input checked="" type="checkbox"/> enable	
Response Map	<input type="text"/>	
Ping Method	<input type="text" value="OPTIONS"/>	
Ping Interval	<input type="text" value="60"/>	(Range: 0.999999999)
Ping Send Mode	<input type="text" value="keep-alive"/>	
Ping All Addresses	<input type="checkbox"/> enable	
Ping In Service Response Codes	<input type="text"/>	
Load Balance DNS Query	<input type="text" value="hunt"/>	
Options	<input type="text"/>	
SPL Options	<input type="text"/>	
Media Profiles	<input type="text"/>	
In Session Translations	<div style="border: 1px solid #ccc; padding: 5px;"> <p>No in session translation list to display. Please add.</p> <p><input type="button" value="Add"/></p> </div>	
Out Session Translations	<div style="border: 1px solid #ccc; padding: 5px;"> <p>No out session translation list to display. Please add.</p> <p><input type="button" value="Add"/></p> </div>	
Trust Me	<input type="checkbox"/> enable	
Stop Recurse	<input type="text"/>	
Local Response Map	<input type="text"/>	
Ping Response	<input type="checkbox"/> enable	

Figure 40: Oracle 4600 Session Agent Amazon Chime SDK Voice Connector Continuation

In Manipulationid	<input type="text"/>	
Out Manipulationid	<input type="text"/>	
Manipulation String	<input type="text"/>	
Manipulation Pattern	<input type="text"/>	
Trunk Group	<input type="text"/>	
Max Register Sustain Rate	<input type="text" value="0"/>	(Range: 0.999999999)
Invalidate Registrations	<input type="checkbox"/> enable	
Rfc2833 Mode	<input type="text" value="none"/>	
Rfc2833 Payload	<input type="text" value="0"/>	(Range: 0,96.327)
Codec Policy	<input type="text" value="G711U"/>	
Refer Call Transfer	<input type="text" value="disabled"/>	
Refer Notify Provisional	<input type="text" value="none"/>	
Reuse Connections	<input type="text" value="NONE"/>	
TCP Keepalive	<input type="text" value="none"/>	
TCP Reconn Interval	<input type="text" value="0"/>	(Range: 0,2_300)
Max Register Burst Rate	<input type="text" value="0"/>	(Range: 0.999999999)
Rate Constraints	<div style="border: 1px solid #ccc; padding: 5px;"> <p>No rate constraints to display. Please add.</p> <p><input type="button" value="Add"/></p> </div>	
SIP Profile	<input type="text"/>	
SIP Isup Profile	<input type="text"/>	
Kpml Interworking	<input type="text" value="inherit"/>	
Kpml2833 Iwf On Hairpin	<input type="text" value="inherit"/>	
Precedence	<input type="text" value="0"/>	(Range: 0.4294967295)
Monitoring Filters	<input type="text"/>	
Auth Attribute	<div style="border: 1px solid #ccc; padding: 5px;"> <p>No auth attributes to display. Please add.</p> <p><input type="button" value="Add"/></p> </div>	
Session Recording Server	<input type="text"/>	
Session Recording Required	<input type="checkbox"/> enable	
Hold Refer Reinvite	<input checked="" type="checkbox"/> enable	
Send TCP Fin	<input type="checkbox"/> enable	
SIP Recursion Policy	<input type="text"/>	
Sm Icsi Match For Invite	<input type="text"/>	
Sm Icsi Match For Message	<input type="text"/>	
Ringback Trigger	<input type="text" value="none"/>	
Ringback File	<input type="text"/>	
Fax Servers	<input type="text"/>	
Trigger Oos Alarm	<input type="checkbox"/> enable	

Activate Windows
Go to Settings to activate Windows.

Figure 41: Oracle 4600 Session Agent Amazon Chime SDK Voice Connector Continuation

4.3.5.4.2 Session-Agent-FreePBX Asterisk

Modify Session Agent
Show Advanced
Show Configuration

Hostname	<input type="text" value="172.16.██"/>	
IP Address	<input type="text" value="172.16.██"/>	
Port	<input type="text" value="5060"/>	(Range: 0,1025..65535)

State enable

App Protocol

App Type

Transport Method	<input type="text" value="UDP"/>	
Realm ID	<input type="text" value="Asterisk"/>	

Egress Realm ID

Description

Match Identifier

No match identifier to display. Please add.

Associated Agents

Constraints enable

Max Sessions	<input type="text" value="0"/>	(Range: 0..999999999)
Max Inbound Sessions	<input type="text" value="0"/>	(Range: 0..999999999)
Max Outbound Sessions	<input type="text" value="0"/>	(Range: 0..999999999)
Max Burst Rate	<input type="text" value="0"/>	(Range: 0..999999999)
Max Inbound Burst Rate	<input type="text" value="0"/>	(Range: 0..999999999)
Max Outbound Burst Rate	<input type="text" value="0"/>	(Range: 0..999999999)
Max Sustain Rate	<input type="text" value="0"/>	(Range: 0..999999999)

Max Inbound Sustain Rate	<input type="text" value="0"/>	(Range: 0..999999999)
Max Outbound Sustain Rate	<input type="text" value="0"/>	(Range: 0..999999999)
Min Asr	<input type="text" value="0"/>	(Range: 0..100)
Cac Trap Threshold	<input type="text" value="0"/>	(Range: 0..99)
Session Max Life Limit	<input type="text" value="0"/>	
Time To Resume	<input type="text" value="0"/>	(Range: 0..999999999)
In Service Period	<input type="text" value="0"/>	(Range: 0..999999999)
Burst Rate Window	<input type="text" value="0"/>	(Range: 0..999999999)
Sustain Rate Window	<input type="text" value="0"/>	(Range: 0..999999999)

Proxy Mode

Redirect Action

Figure 42: Oracle 4600 Session Agent Asterisk

Loose Routing	<input checked="" type="checkbox"/> enable
Response Map	<input type="text" value=""/>
Ping Method	<input type="text" value="OPTIONS"/>
Ping Interval	<input type="text" value="60"/> (Range: 0.999999999)
Ping Send Mode	<input type="text" value="keep-alive"/>
Ping All Addresses	<input type="checkbox"/> enable
Ping In Service Response Codes	<input type="text" value=""/>
Load Balance DNS Query	<input type="text" value="hunt"/>
Options	<input type="text" value=""/>
SPL Options	<input type="text" value=""/>
Media Profiles	<input type="text" value=""/>

In Session Translations

No in session translation list to display. Please add.

Out Session Translations

No out session translation list to display. Please add.

Trust Me	<input type="checkbox"/> enable
Stop Recurse	<input type="text" value=""/>
Local Response Map	<input type="text" value=""/>
Ping Response	<input type="checkbox"/> enable
In Manipulationid	<input type="text" value=""/>
Out Manipulationid	<input type="text" value=""/>

Manipulation String	<input type="text" value=""/>
Manipulation Pattern	<input type="text" value=""/>
Trunk Group	<input type="text" value=""/>
Max Register Sustain Rate	<input type="text" value="0"/> (Range: 0.999999999)
Invalidate Registrations	<input type="checkbox"/> enable
Rfc2833 Mode	<input type="text" value="none"/>
Rfc2833 Payload	<input type="text" value="0"/> (Range: 0.96.127)
Codec Policy	<input type="text" value="G711U"/>
Refer Call Transfer	<input type="text" value="disabled"/>
Refer Notify Provisional	<input type="text" value="none"/>
Reuse Connections	<input type="text" value="NONE"/>

Figure 43: Oracle 4600 Session Agent Asterisk Continuation

TCP Keepalive	<input type="text" value="none"/>	
TCP Reconn Interval	<input type="text" value="0"/>	(Range: 0.2-300)
Max Register Burst Rate	<input type="text" value="0"/>	(Range: 0.999999999)
Rate Constraints		
No rate constraints to display. Please add.		
<input type="button" value="Add"/>		
SIP Profile	<input type="text"/>	
SIP Isup Profile	<input type="text"/>	
Kpml Interworking	<input type="text" value="inherit"/>	
Kpml2833 lwf On Hairpin	<input type="text" value="inherit"/>	
Precedence	<input type="text" value="0"/>	(Range: 0.4294967295)
Monitoring Filters		
<input type="text"/>		
Auth Attribute		
No auth attributes to display. Please add.		
<input type="button" value="Add"/>		
Session Recording Server	<input type="text"/>	
Session Recording Required	<input type="checkbox"/> enable	
Hold Refer Reinvite	<input checked="" type="checkbox"/> enable	
Send TCP Fin	<input type="checkbox"/> enable	
SIP Recursion Policy	<input type="text"/>	
Sm Icsi Match For Invite	<input type="text"/>	
Sm Icsi Match For Message		
<input type="text"/>		
Ringback Trigger	<input type="text" value="none"/>	
Ringback File	<input type="text"/>	
Fax Servers	<input type="text"/>	
Trigger Oos Alarm	<input type="checkbox"/> enable	
<input type="button" value="OK"/> <input type="button" value="Back"/>		Activate Windows Go to Settings to activate Windows.

Figure 44: Oracle 4600 Session Agent Asterisk Continuation

4.3.5.5 Session Recording Server

Navigate to **Configuration** → **Session-Router** → **Session-Recording-Server**

- Add a Session Recording Server pointing to Amazon Chime SDK Voice Connector SDK

Name: Enter the name for the Session Recording Server

Description: Enter the description for the Session Recording Server

Realm: Choose the Realm created for Amazon Chime SDK Voice Connector

Mode: Selective

Destination: Select the destination

Port: 5061

Transport Method: Static TLS

Modify Session Recording Server

Show Advanced Show Configuration

Name	SRSACVC
Description	SRSACVC
Realm	ACVCRecdServer
Mode	selective
Destination	fa2e5s3
Port	5061 (Range: 1024..65535)
Transport Method	StaticTLS

Force Parity	<input type="checkbox"/> enable
Ping Method	
Ping Interval	0 (Range: 0..4294967295)
Refresh Interval	0 (Range: 0..60)

OK Back

Activate Windows
Go to Settings to activate Windows.

Figure 45: Oracle 4600 Session Recording Server

4.3.5.6 Local Policy

Navigate to **Configuration** → **Session-Router** → **Local-Policy** and add two local Policies for

1. FreePBX Asterisk
2. Amazon Chime SDK Voice Connector Recording

From Address: *

To Address: *

Source Realm: Select the Realm

Next Hop: Select the Next Hop

4.3.5.6.1 Local Policy -Amazon Chime SDK Voice Connector

Modify Local Policy Show Advanced Show Configuration

From Address

To Address

Source Realm

Description

State enable

Parallel Forking enable

Policy Priority

Policy Attributes

Select	Action	Next Hop	Realm	Action	Terminate Recursion	Cost	State	App Protocol	Lookup	Next Key	Auth User Lookup
<input type="checkbox"/>	:	172.16.1.1	Asterisk	none	disabled	0	enabled		single		

Figure 46: Oracle 4600 Local Policy Amazon Chime SDK Voice Connector

4.3.5.6.2 Local Policy -FreePBX Asterisk

Modify Local Policy
Show Advanced
Show Configuration

From Address

To Address

Source Realm

Description

State enable

Parallel Forking enable

Policy Priority

Policy Attributes

Select	Action	Next Hop	Realm	Action	Terminate Recursion	Cost	State	App Protocol	Lookup	Next Key	Auth User Lookup
<input type="checkbox"/>	:	fa2e5s3	ACVCRecdSer...	none	disabled	0	enabled		single		

Figure 47: Oracle 4600 Local Policy Asterisk

4.3.6 Security Configuration

4.3.6.1 Certificate Record

Navigate to **Configuration** → **Security** → **Certificate Record**

Name: Enter the name for the Certificate Record

Common Name: Enter the common name for the Certificate Record

- Generate the CSR for the Oracle ESBC and upload the CSR signed by the CA to the certificate record created for the ESBC.

4.3.6.1.1 Certificate Record for Oracle ESBC

Modify Certificate Record Show Advanced Show Configuration

Name	Oracle_4600
Country	US
State	TX
Locality	Plano
Organization	
Unit	
Common Name	192.65
Key Size	2048
Alternate Name	
Trusted	<input checked="" type="checkbox"/> enable
Key Usage List	digitalSignature x keyEncipherment x
Extended Key Usage List	serverAuth x
Key Algor	rsa
Digest Algor	sha256
Ecdsa Key Size	p256
Cert Status Profile List	
Options	

OK Back

Activate Windows
Go to Settings to activate Windows.

Figure 48: Oracle 4600 Certificate Record

4.3.6.1.2 Certificate Record for Amazon Chime SDK Voice Connector

- Import the Amazon's Root Certificate to the Certificate Record.
- Amazon Chime Voice Connector Root Certificate can be downloaded from Amazon Chime SDK Voice Connector account.

The screenshot displays the 'Modify Certificate Record' interface. At the top right, there are 'Show Advanced' (checked) and 'Show Configuration' buttons. The main form contains the following fields:

Name	AmazonCert
Country	US
State	MA
Locality	Burlington
Organization	Engineering
Unit	
Common Name	Amazon Root CA 1
Key Size	2048
Alternate Name	

Below the main form, there is a 'Trusted' checkbox which is checked and labeled 'enable'. To the right of this section is a watermark that says 'Activate Windows'. A sidebar on the left contains a navigation menu with the following items: realm-config, steering-pool, security (with a dropdown arrow), authentication-profile, certificate-record (highlighted), tls-global, tls-profile, session-router (with a right arrow), and system (with a right arrow). At the bottom left of the sidebar is a 'Show All' toggle switch. At the bottom center are 'OK' and 'Back' buttons. The right side of the sidebar contains configuration options for Key Usage List (digitalSignature x, keyEncipherment x), Extended Key Usage List (serverAuth x, clientAuth x), Key Algor (rsa), Digest Algor (sha256), Ecdsa Key Size (p256), Cert Status Profile List, and Options.

Figure 49: Oracle 4600 Certificate Record Amazon Chime SDK Voice Connector

4.3.6.2 TLS Profile

Navigate to **Configuration** → **Security** → **TLS Profile**

- **Name:** Enter the name for the TLS profile
- **End Entity Certificate:** Select the Certificate Record Created for Oracle ESBC
- **Trusted Ca Certificate:** Select the Certificate Record Created Amazon Chime SDK Voice Connector
- **Cipher List:** Select the list of Ciphers to be used
- **TLS Version:** tlsv12

The screenshot displays the 'Modify TLS Profile' configuration page for Oracle 4600. The page is organized into three main sections. The top section, titled 'Modify TLS Profile', shows the profile name as 'Amazon_TLS_Profile', the end entity certificate as 'Oracle_4600', and the trusted CA certificate as 'AmazonCert'. The cipher list includes: TLS_DHE_RSA_WITH_AES_128_CBC_SHA256, TLS_DHE_RSA_WITH_AES_128_GCM_SHA256, TLS_RSA_WITH_AES_256_GCM_SHA384, TLS_RSA_WITH_AES_256_CBC_SHA256, TLS_RSA_WITH_AES_128_GCM_SHA256, TLS_RSA_WITH_AES_128_CBC_SHA256, TLS_RSA_WITH_AES_128_CBC_SHA, and TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384. The middle section shows a cipher list with: TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA384, TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256, TLS_ECDHE_RSA_WITH_AES_128_CBC_SHA256, TLS_ECDHE_ECDSA_WITH_AES_256_GCM_SHA384, TLS_ECDHE_ECDSA_WITH_AES_128_GCM_SHA256, TLS_DHE_RSA_WITH_AES_256_CBC_SHA256, and TLS_DHE_RSA_WITH_AES_256_GCM_SHA384. The 'Verify Depth' is set to 10, and 'Mutual Authenticate' is disabled. The bottom section shows the 'TLS Version' set to 'tlsv12', 'Options' as an empty field, and several checkboxes for 'Cert Status Check', 'Cert Status Profile List', 'Ignore Dead Responder', and 'Allow Self Signed Cert', all of which are disabled. The page includes 'Show Advanced' and 'Show Configuration' buttons, and an 'Activate Windows' watermark.

Figure 50: Oracle 4600 TLS Profile