



Amazon Chime SDK Voice Connector

SIP Trunking Configuration Guide

**FreePBX 16.0.40, Asterisk 20.1.0 and
Oracle Acme Packet 4600 Enterprise
Session Border Controller (Oracle
AP4600 ESBC)**

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1 Audience

This document is intended for technical staff and Value-Added Resellers (VARs) with installation and operational responsibilities. This configuration guide provides steps for configuring SIP trunks using **FreePBX (Asterisk)** and **Oracle Acme Packet 4600 Enterprise Session Border Controller** (Oracle AP4600 ESBC) to connect to **Amazon Chime SDK Voice Connector** for inbound and/or outbound telephony capabilities.

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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 SIP trunk reference configuration is illustrated below and is representative of **FreePBX Asterisk** and **Oracle Acme Packet 4600 ESBC** configuration.

IP PBX-2 is used as a secondary PBX in the topology to perform call failover and call distribution

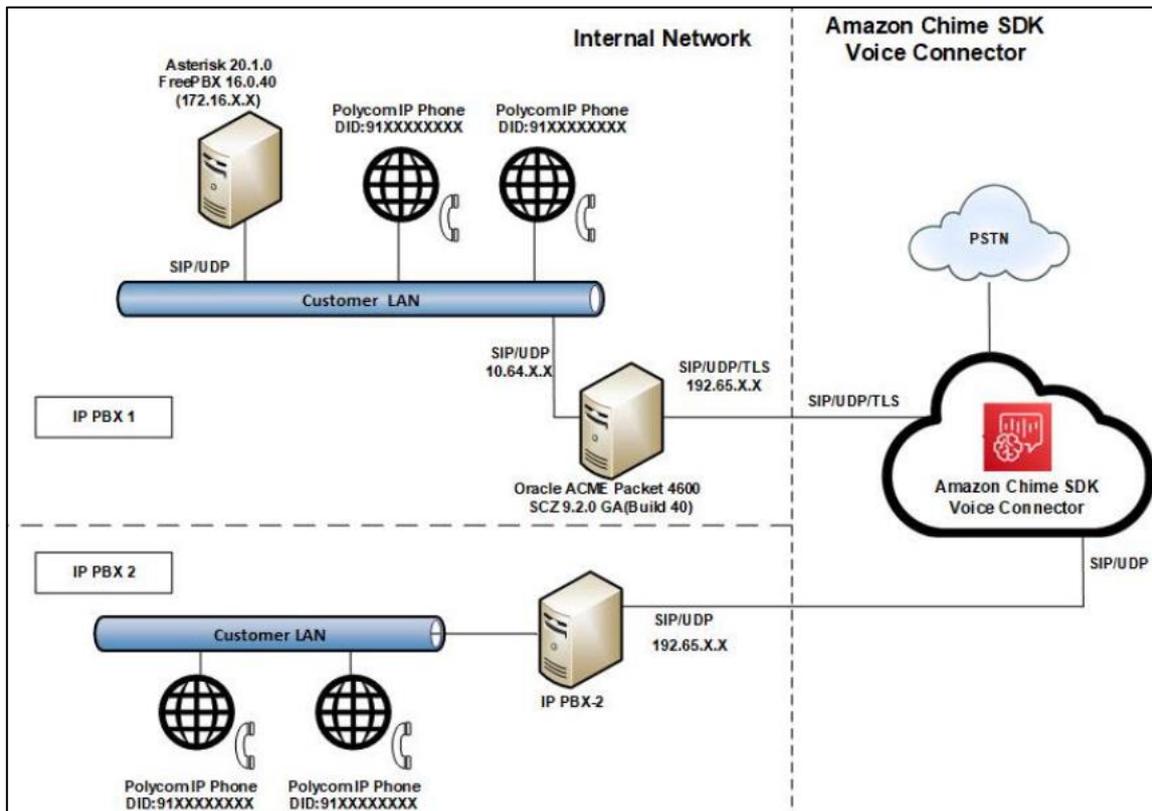


Figure 1: Network Topology

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 Acme Packet 4600 ESBC SCZ9.2.0 GA (Build 40)

3 Features

3.1 Features Supported

- Calls to and from non Toll Free number
- Calls to Toll Free number
- Calls to Premium Telephone number
- Calling Party Number Presentation
- Calling Party Number Restriction
- Inbound Calls to an IVR
- International Calls
- Call Authentication
- Anonymous call
- Secured inbound and outbound calls with media encryption
- DTMF-RFC 2833
- Long duration calls
- Calls to conference scheduled by Amazon Chime user
- Calls to Amazon Chime Business number
- Call Distribution
- Call Failover

3.2 Features Not Supported

- The following are not supported by Amazon Chime SDK Voice Connector,
 - Keep Alive – Double CRLF
 - Keep Alive – SIP OPTIONS

3.3 Features Not Tested

- None

3.4 Caveats and Limitations

- Amazon Chime SDK Voice connector:
 - does not support SIP NOTIFY or SIP INFO for DTMF
 - does not send SIP session refresher for long duration calls
- When the WAN link is down and a call is in progress, the PSTN call leg is not disconnected automatically after a period of inactivity. The call has to be cleared manually.

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 Acme Packet 4600 ESBC** for SIP Trunking with **Amazon Chime SDK Voice Connector**.

Table 1 – PBX and ESBC Configuration Steps

Steps	Description	Reference
Step 1	FreePBX Asterisk Configuration	Section 5
Step 2	Oracle 4600 ESBC Configuration	Section 6
Step3	Amazon Chime Voice Connector Configuration	Amazon Chime Voice Connector

4.2 IP Address Worksheet

The specific values listed in the table below and subsequent sections are used in the lab configuration described in this document and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

Table 2 – IP Addresses

Component	Lab Value
Oracle Acme Packet 4600	
LAN IP Address	10.64.5.247
LAN Subnet Mask	255.255.0.0
FreePBX Asterisk	
IP Address	172.16.29.56
Subnet Mask	255.255.255.0

5 FreePBX Asterisk Configuration

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

5.1 FreePBX Asterisk Version

5.1.1 FreePBX Version



Figure 2: FreePBX Version

5.1.2 Asterisk Version

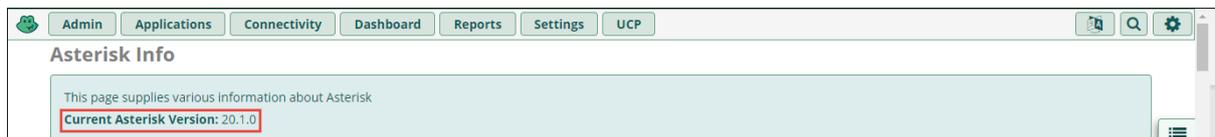


Figure 3: Asterisk Version

5.2 SIP Trunk

Navigate to **Settings** → **Asterisk SIP Settings** → **SIP Settings (Chan_pjsip)**

UDP-0.0.0.0-All: Yes

Port to Listen on: 5060

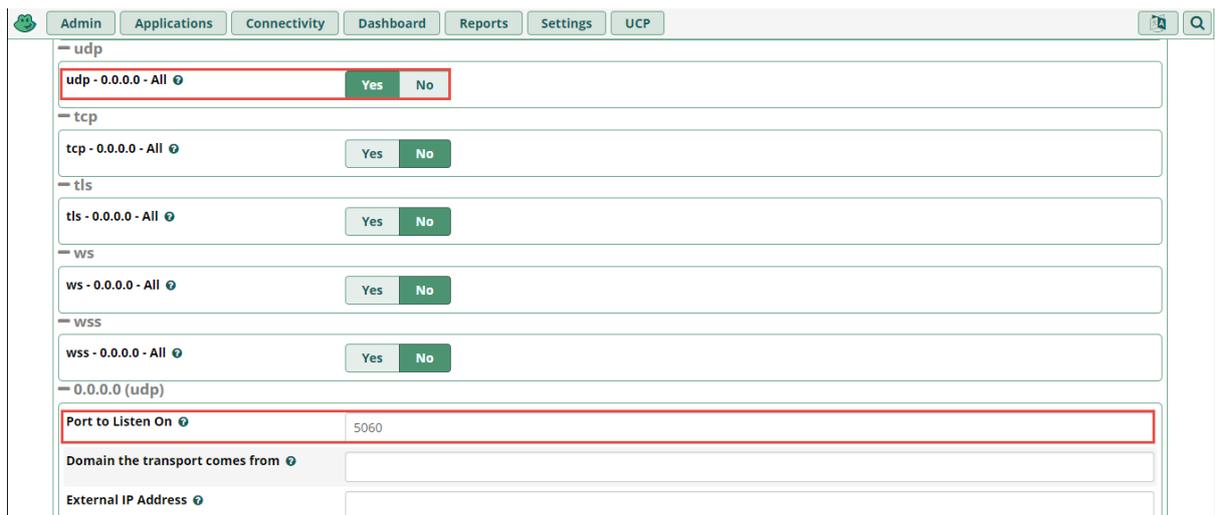
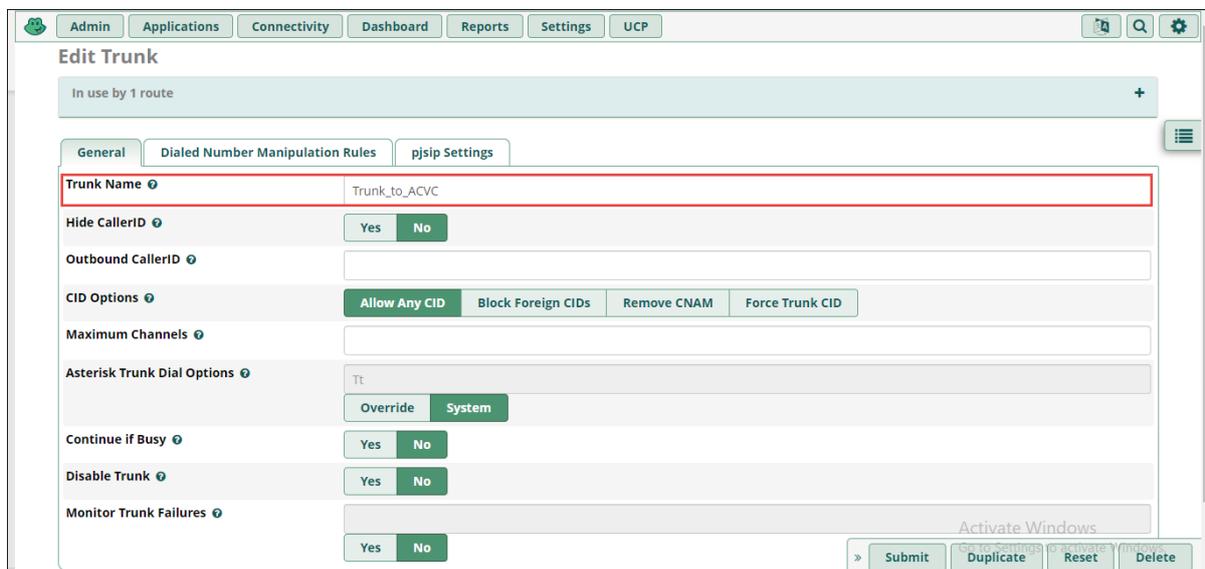


Figure 4: Asterisk UDP Port

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

Trunk Name: Enter a name for the Trunk



The screenshot shows the 'Edit Trunk' configuration page in Asterisk. The 'Trunk Name' field is highlighted with a red box and contains the text 'Trunk_to_ACVC'. Other fields include 'Hide CallerID' (Yes/No), 'Outbound CallerID', 'CID Options' (Allow Any CID, Block Foreign CIDs, Remove CNAM, Force Trunk CID), 'Maximum Channels', 'Asterisk Trunk Dial Options' (Tt, Override/System), 'Continue if Busy' (Yes/No), 'Disable Trunk' (Yes/No), and 'Monitor Trunk Failures' (Yes/No). The page has a navigation bar at the top with 'Admin', 'Applications', 'Connectivity', 'Dashboard', 'Reports', 'Settings', and 'UCP'. At the bottom right, there are buttons for 'Submit', 'Duplicate', 'Reset', and 'Delete'.

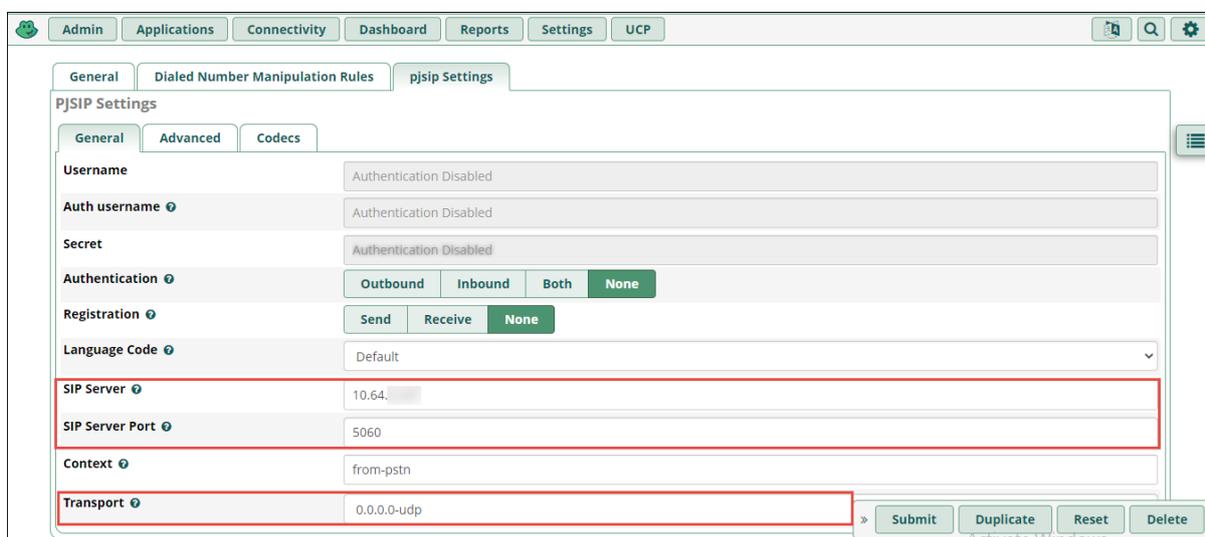
Figure 5: Asterisk SIP Trunk

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

SIP Server: 10.64.XX.XX (Oracle ESBC Network Interface IP address towards the FreePBX Asterisk)

SIP Server Port: 5060

Transport: 0.0.0.0-udp



The screenshot shows the 'PJSIP Settings' configuration page in Asterisk. The 'General' tab is selected. Fields include '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.XX.XX), 'SIP Server Port' (5060), 'Context' (from-pstn), and 'Transport' (0.0.0.0-udp). The 'SIP Server', 'SIP Server Port', and 'Transport' fields are highlighted with red boxes. The page has a navigation bar at the top with 'Admin', 'Applications', 'Connectivity', 'Dashboard', 'Reports', 'Settings', and 'UCP'. At the bottom right, there are buttons for 'Submit', 'Duplicate', 'Reset', and 'Delete'.

Figure 6: Asterisk SIP Trunk Continuation

Navigate to **Pjsip settings** → **Advanced**
Send RPID/PAI: Select **Send P-Asserted-Identity header**

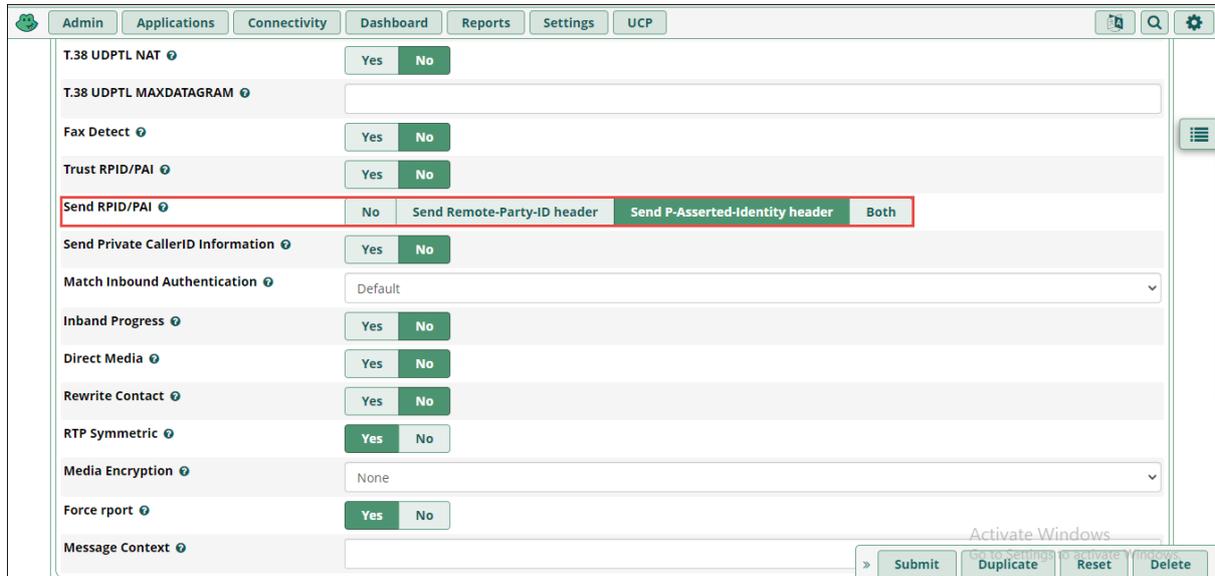


Figure 7: Asterisk SIP Trunk Continuation

Navigate to **Pjsip settings** → **Codecs**
Enable Ulaw
Click **Submit**

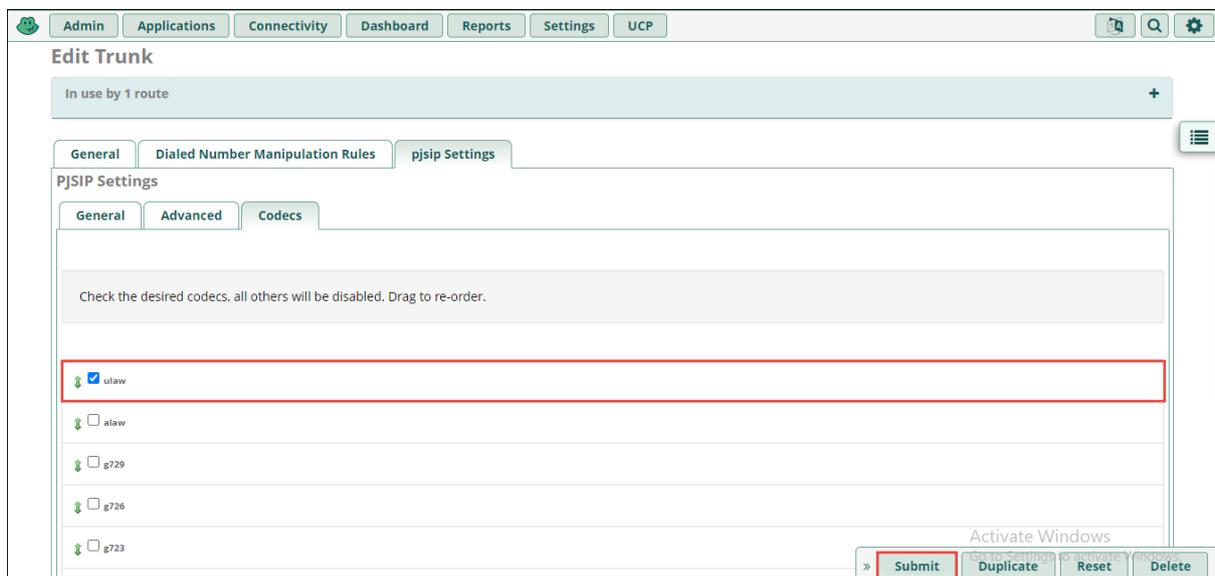


Figure 8: Asterisk SIP Trunk Continuation

5.3 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

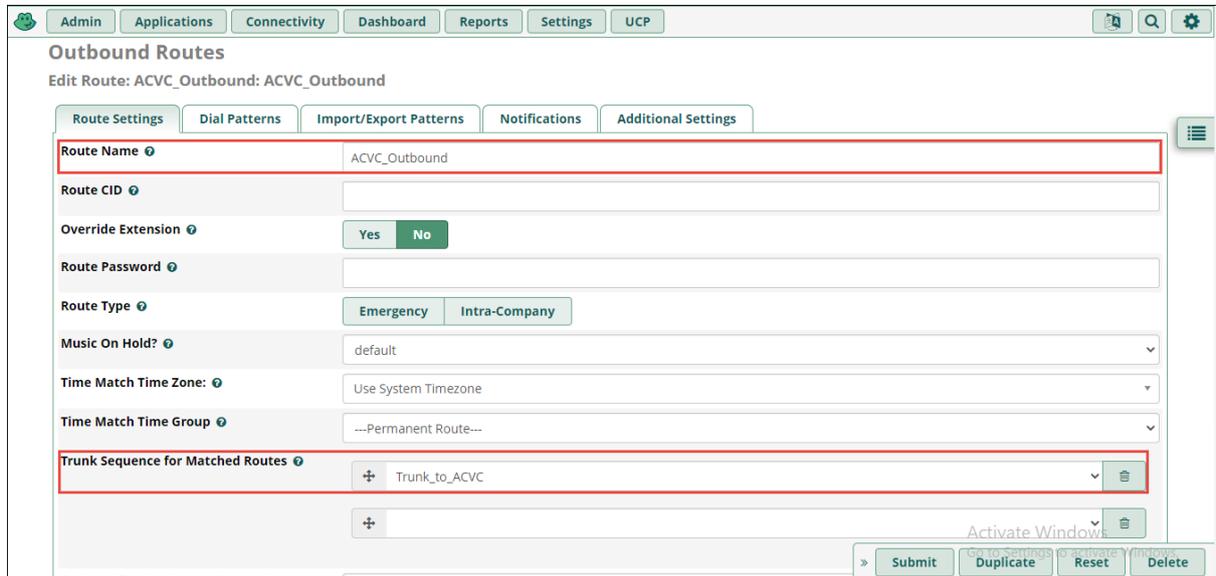


Figure 9: Asterisk Outbound Route

Navigate to **Dial Patterns**

Match Pattern: Enter the dial patterns

Prepend: +1 or + (It is required to convert the DID in the 'To' header and 'Request uri' into E.164 format or it can also be done in the SBC via manipulation)

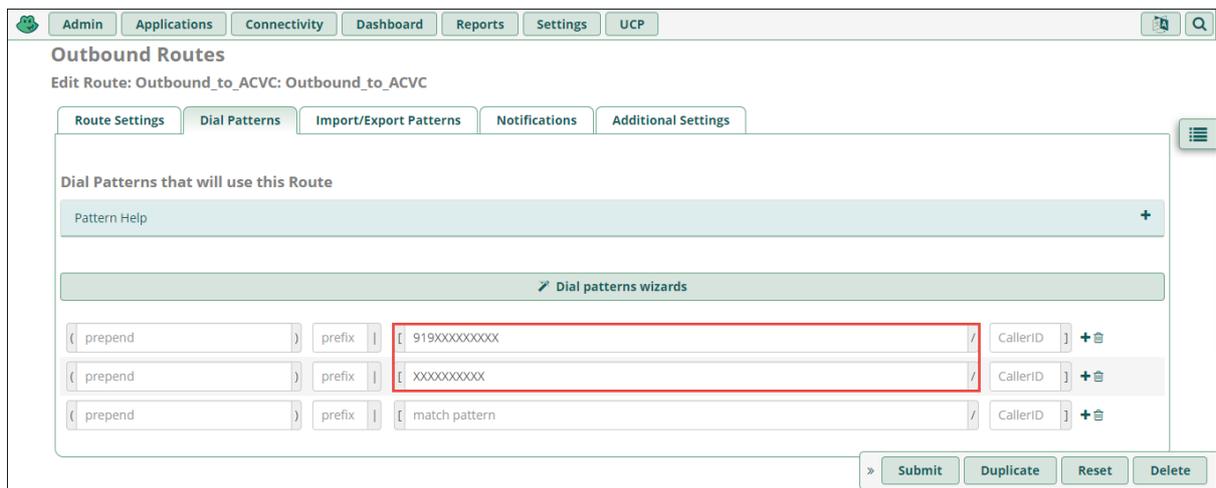


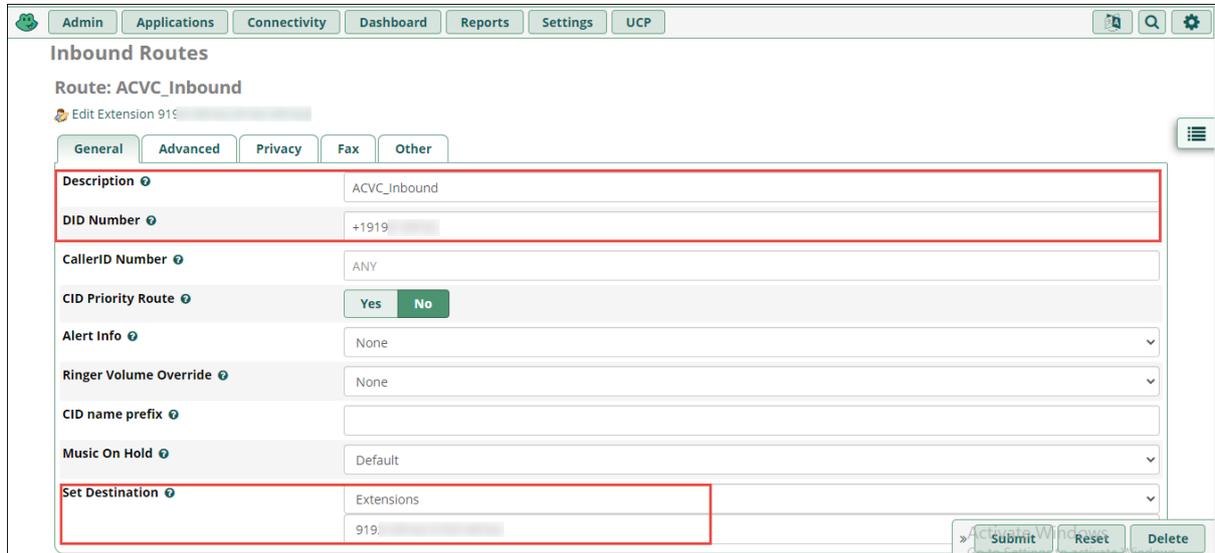
Figure 10: Asterisk Outbound Route Continuation

5.4 Inbound Route

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

DID Number: +191XXXXXXXX

Set Destination: Select Extensions/91XXXXXXXX



The screenshot shows the Asterisk Inbound Route configuration interface. The page title is "Inbound Routes" and the specific route is "Route: ACVC_Inbound". The "General" tab is selected, showing various configuration fields. Two red boxes highlight the "Description" field (containing "ACVC_Inbound") and the "Set Destination" field (containing "Extensions" and "919:"). The "Set Destination" field is a dropdown menu with "Extensions" selected and a text input field below it containing "919:". At the bottom right, there are "Submit", "Reset", and "Delete" buttons.

Field	Value
Description	ACVC_Inbound
DID Number	+1919
CallerID Number	ANY
CID Priority Route	Yes No
Alert Info	None
Ringer Volume Override	None
CID name prefix	
Music On Hold	Default
Set Destination	Extensions 919:

Figure 11: Asterisk Inbound Route

5.5 Extension

Navigate to **Application → Extensions → Add New SIP (Chan_Pjsip) Extension**

- **User Extension:** 919XXXXXXX
- **Display Name:** 919XXXXXXX
- **Outbound CID:** +1919XXXXXXX (It is required to convert the DID in the 'From' header into E.164 format or it can also be done in SBC via manipulation)

The screenshot shows the 'Add PJSIP Extension' configuration page in a web interface. The page has a navigation bar at the top with tabs for Admin, Applications, Connectivity, Dashboard, Reports, Settings, and UCP. Below the navigation bar, there are sub-tabs for General, Voicemail, Find Me/Follow Me, Advanced, Pin Sets, and Other. The main content area is titled 'Add PJSIP Extension 919' and contains several sections:

- Add Extension:** A section with a note: 'This device uses PJSIP technology listening on Port 5060 (UDP), Port 5060 (TCP)'. It contains input fields for 'User Extension' (919), 'Display Name' (919), 'Outbound CID', 'Emergency CID', and 'Secret'.
- Language:** A section with a 'Language Code' dropdown menu set to 'Default'.
- User Manager Settings:** A section with a 'Select User Directory' dropdown menu set to 'PBX Internal Directory'. It also includes 'Submit' and 'Reset' buttons.
- User Management:** A section with a 'Link to a Different Default User' dropdown menu set to '919', a 'Username' input field, a 'Password For New User' input field, and a 'Groups' dropdown menu set to 'All Users'. It also includes 'Submit', 'Reset', and 'Delete' buttons.

Figure 12: Asterisk Extension

6 Oracle 4600 ESBC Configuration

6.1 Oracle 4600 ESBC Version

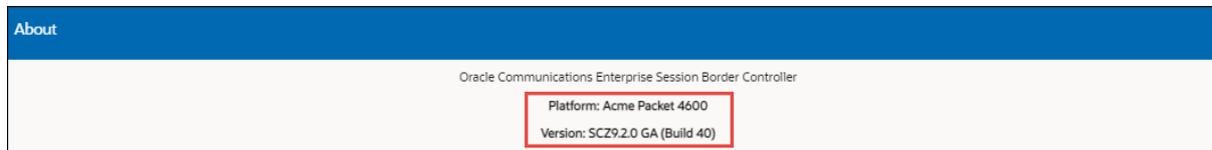


Figure 13: Oracle 4600 ESBC Version

6.2 Physical Interface

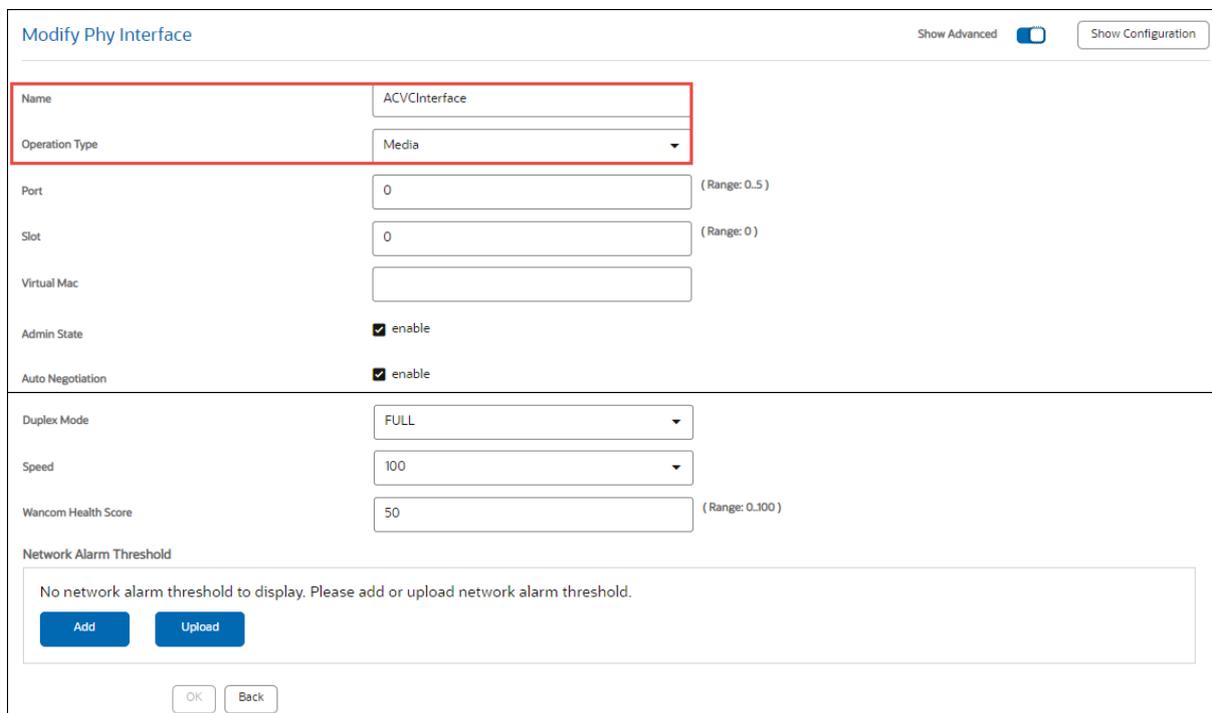
The following Physical Interfaces are created in Oracle 4600 ESBC

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

- 1) s0p0 - Interface towards Amazon Chime SDK Voice Connector
- 2) s0p1 – Interface towards FreePBX Asterisk

6.2.1 Physical Interface towards Amazon Chime SDK Voice Connector

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

The image shows a screenshot of the 'Modify Phy Interface' configuration page. The page title is 'Modify Phy Interface' and it includes a 'Show Advanced' toggle and a 'Show Configuration' button. The configuration fields are as follows:

- Name:** ACVCInterface (highlighted with a red box)
- Operation Type:** Media (highlighted with a red box)
- Port:** 0 (Range: 0.5)
- Slot:** 0 (Range: 0)
- Virtual Mac:** (empty field)
- Admin State:** enable
- Auto Negotiation:** enable
- Duplex Mode:** FULL
- Speed:** 100
- Wancom Health Score:** 50 (Range: 0.100)
- Network Alarm Threshold:** No network alarm threshold to display. Please add or upload network alarm threshold. (Buttons: Add, Upload)

At the bottom, there are 'OK' and 'Back' buttons.

Figure 14: Oracle 4600 Physical Interface Amazon Chime SDK Voice Connector

6.2.2 Physical Interface towards FreePBX Asterisk

Modify Phy Interface Show Advanced Show Configuration

Name	AsteriskInterface
Operation Type	Media
Port	1 (Range: 0.5)
Slot	0 (Range: 0)
Virtual Mac	
Admin State	<input checked="" type="checkbox"/> enable
Auto Negotiation	<input checked="" type="checkbox"/> enable
Duplex Mode	FULL
Speed	100
Wancom Health Score	50 (Range: 0.100)

Network Alarm Threshold

No network alarm threshold to display. Please add or upload network alarm threshold.

Figure 15: Oracle 4600 Physical Interface Asterisk

6.3 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
- 2) s0p1 – Assign an IP address to the Interface towards FreePBX Asterisk

6.3.1 Network Interface towards Amazon Chime SDK Voice Connector

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

IP Address: 192.65.XX.XX

Network: 255.255.XXX.XXX

Gateway: 192.65.XX.XX

DNS IP Primary: 8.8.8.8

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

HIP IP List: 192.65.XX.XX

ICMP Address: 192.65.XX.XX

The screenshot shows the 'Modify Network Interface' configuration page. The interface name is 'ACVCInterface'. The sub-port ID is '0'. The description is 'Towards ACVC'. The IP address is '192.65.XX.XX' and the netmask is '255.255.XXX.XXX'. The page includes a 'Show Advanced' toggle and a 'Show Configuration' button.

Modify Network Interface	
Name	ACVCInterface
Sub Port Id	0 (Range: 0..4095)
Description	Towards ACVC
Hostname	
IP Address	192.65.XX.XX
Pri Utility Addr	
Sec Utility Addr	
Netmask	255.255.XXX.XXX

Figure 16: Oracle 4600 Network Interface Amazon Chime SDK Voice Connector

Gateway

enable
 State

Heartbeat (Range: 0..65535)

Retry Count (Range: 0..65535)

Retry Timeout (Range: 1..65535)

Health Score (Range: 0..100)

DNS IP Primary

DNS IP Backup1

DNS IP Backup2

Activate Windows

Modify Network Interface Show Advanced Show Configuration

DNS Domain

DNS Timeout (Range: 1..999999999)

DNS Max TTL (Range: 30..2073600)

Signalling Mtu (Range: 0..4096)

HIP IP List

ICMP Address

SSH Address

Tunnel Config

No tunnel config to display. Please add.

Activate Windows
Go to Settings to activate Windows.

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

6.3.2 Network Interface towards FreePBX Asterisk

Name: Select the Physical Interface created for FreePBX Asterisk

IP Address: 10.64.XX.XX

Network: 255.255.XX.XX

Gateway: 10.64.XX.XX

DNS IP Primary: 10.87.XX.XX

DNS Domain: Enter the DNS Domain

HIP IP List: 10.64.XX.XX

ICMP address: 10.64.XX.XX

The screenshot displays the 'Modify Network Interface' configuration page for an Oracle 4600 network interface. The interface is titled 'AsteriskInterface'. Key configuration fields are highlighted with red boxes:

- Name:** AsteriskInterface
- Sub Port Id:** 0 (Range: 0-4095)
- Description:** Towards Asterisk
- IP Address:** 10.64
- Netmask:** 255.25
- Gateway:** 10.64
- DNS IP Primary:** 10.87

Other visible fields include Hostname, Pri Utility Addr, Sec Utility Addr, Gw Heartbeat (with sub-fields for State, Heartbeat, Retry Count, Retry Timeout, and Health Score), DNS IP Backup1, and DNS IP Backup2. The 'Show Advanced' toggle is turned on, and a 'Show Configuration' button is present in the top right corner.

Figure 18: Oracle 4600 Network Interface Asterisk

DNS Domain	<input type="text" value="...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" value="10.64"/>	
ICMP Address	<input type="text" value="10.64"/>	
SSH Address	<input type="text"/>	
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 19: Oracle 4600 Network Interface Asterisk Continuation

6.4 Codec Policy

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

1. **Name:** Enter a 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 the value 'G711U'. The 'Allow Codecs' field is also highlighted with a red box and contains two tags: 'PCMU' and 'telephone-event'. Below these fields are several other configuration options:

- Add Codecs On Egress: [Empty text box]
- Order Codecs: [Empty text box]
- Packetization Time: [20]
- Force Ptime: enable
- Secure Dtmf Cancellation: enable
- Dtmf In Audio: [disabled]
- Tone Detection: [Empty text box]
- Tone Detect Renegotiate Timer: [500] (Range: 50..32000)
- Reverse Fax Tone Detection Reininvite: enable
- Fax Single M Line: [disabled]
- EvrC Tty Baudot Transcode: enable

At the bottom left, there are 'OK' and 'Back' buttons. At the bottom right, there is a watermark: 'Activate Windows Go to Settings to activate Windows.'

Figure 20: Oracle 4600 Codec Policy

6.5 Realm Configuration

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

1. Amazon Chime SDK Voice Connector
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 to [Section 6.9](#) for SIP Manipulation Configuration)

Media Sec Policy: Select the media policy

6.5.1 Realm for Amazon Chime SDK Voice Connector

The screenshot shows the 'Modify Realm Config' interface. At the top right, there are 'Show Advanced' and 'Show Configuration' buttons. The form contains the following fields and values:

- Identifier: ACVCRecdServer
- Description: ACVCRecdServer
- Addr Prefix: 0.0.0.0
- Network Interfaces: ACVCInterface:0.4 x
- Media Realm List: (empty)
- Mm In Realm: enable
- Mm In Network: enable
- Mm Same Ip: enable
- QoS Enable: enable
- Max Bandwidth: 0 (Range: 0.999999999)
- Max Priority Bandwidth: 0 (Range: 0.999999999)
- Parent Realm: (dropdown)
- DNS Realm: (dropdown)
- Media Policy: (dropdown)
- Nsep Media Policy: (dropdown)
- Media Sec Policy: RTP (dropdown, highlighted with a red box)
- RTPCP Mux: enable

Figure 21: 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 22: 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	Activate Windows
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 style="text-align: center;"><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"/>		Activate Windows Go to Settings to activate Windows.

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

6.5.2 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	<input type="text" value="0"/> (Range: 0.999999999)
Max Priority Bandwidth	<input type="text" value="0"/> (Range: 0.999999999)
Parent Realm	<input type="text"/>
DNS Realm	<input type="text"/>
Media Policy	<input type="text"/>
Nsep Media Policy	<input type="text"/>
Media Sec Policy	RTP
RTCP Mux	<input type="checkbox"/> enable
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"/>
Out Session Translations	No out session translation list to display. Please add. <input type="button" value="Add"/>

Figure 24: 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 25: Oracle 4600 Realm Asterisk 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: -1.2147485647)
Initial Guard Timer	<input type="text" value="-1"/> (Range: -1.2147485647)
Subsq Guard Timer	<input type="text" value="-1"/> (Range: -1.2147485647)
TCP Flow Time Limit	<input type="text" value="-1"/> (Range: -1.2147485647)
TCP Initial Guard Timer	<input type="text" value="-1"/> (Range: -1.2147485647)
TCP Subsq Guard Timer	<input type="text" value="-1"/> (Range: -1.2147485647)
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)
<div style="float: right;"> <p>Activate Windows Go to Settings to activate Windows.</p> </div> <div style="clear: both;"></div>	
<input type="button" value="OK"/> <input type="button" value="Back"/>	

Figure 26: Oracle 4600 Realm Asterisk Continuation

6.6 Steering Pool

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

1. Amazon Chime SDK Voice Connector
2. FreePBX Asterisk

IP Address: Enter the IP Address

Realm Id: Select the Realm Id

Network Interface: Select the Network Interface

6.6.1 Steering Pool for Amazon Chime SDK Voice Connector

The screenshot shows the 'Modify Steering Pool' configuration interface. The fields are as follows:

Field	Value
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 27: Oracle 4600 Steering Pool Amazon Chime SDK Voice Connector

6.6.2 Steering Pool for FreePBX Asterisk

The screenshot shows the 'Modify Steering Pool' configuration interface. The fields are as follows:

Field	Value
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 28: Oracle 4600 Steering Pool Asterisk

6.7 Media Profile

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

- Media Profile is created to add Codec and DTMF events when an 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 a name for the media profile

Subname: Enter a subname for the media profile

Media Type: Audio

Payload Type: 0,101

Transport: RTP/AVP

6.7.1 Media Profile for Codec PCMU

The screenshot displays the 'Modify Media Profile' configuration interface. The top right corner features 'Show Advanced' (checked) and 'Show Configuration' buttons. The main configuration area includes 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: 0.9999999999)
Frames Per Packet	0 (Range: 0.256)
Parameters	
As Bandwidth	0 (Range: 0.4294967295)

At the bottom left, there are 'OK' and 'Back' buttons. At the bottom right, there is an 'Activate Windows' watermark with the text 'Go to Settings to activate Windows.'

Figure 29: Oracle 4600 Media Profile PCMU

6.7.2 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)

Activate Windows
Go to Settings to activate Windows.

Figure 30: Oracle 4600 Media Profile Telephone Event

6.8 Media Sec Policy for RTP

- Associate **media-sec-policy: RTP** with the Realm config of FreePBX Asterisk

Name: RTP

Inbound(Mode): RTP

Outbound(Mode): RTP

Modify Media Sec Policy Show Advanced Show Configuration

Name	RTP
Pass Through	<input type="checkbox"/> enable
Options	<input type="text"/>
▼ Inbound	
Profile	<input type="text"/>
Mode	rtp
Protocol	none
Hide Egress Media Update	<input type="checkbox"/> enable
▼ Outbound	
Profile	<input type="text"/>
Mode	rtp
Protocol	none

OK Back

Activate Windows
Go to Settings to activate Windows.

Figure 31: Oracle 4600 Media SEC Policy for RTP

6.9 Session Timer Profile

Name: Enter a name for Timer Profile

Session Expires: 1800

Min Se: 90

Request Refresher: uac

Response Refresher: uas

- Associate the Session Timer Profile to the SIP interface for the Amazon Chime SDK Voice Connector and FreePBX Asterisk

The screenshot shows the 'Modify Session Timer Profile' configuration page. The page title is 'Modify Session Timer Profile' and it includes 'Show Advanced' and 'Show Configuration' buttons. The configuration fields are as follows:

Name	ACVC_SessionTimer_Profile	
Session Expires	1800	(Range: 64.999999999)
Min Se	90	(Range: 64.999999999)
Force Reinvite	<input type="checkbox"/> enable	
Request Refresher	uac	
Response Refresher	uas	

Figure 32: Oracle 4600 Session Timer Profile

6.10 SIP Manipulation

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

6.10.1 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. Tags – **Optional** & **Mandatory** are used to indicate the SIP manipulations that are Mandatory and Optional in an SBC configuration.

sip-manipulation

name	AmazonManipulation
description	AmazonManipulation

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

header-rule

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	<Outbound_HostName from Amazon Chime SDK Voice Connector>

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

header-rule

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>

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 (Optional)

header-rule

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>

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

header-rule

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

This manipulation is to replace the owner name in the SDP with "OracleACME" (Optional)

header-rule

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

This manipulation is to replace the User agent in the request with the name and version of ESBC (Optional)

header-rule	
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

This manipulation is to replace the IP address in the Contact header with the Oracle ESBC IP address (Optional)

header-rule	
name	mod contact
header-name	Contact
action	manipulate
comparison-type	pattern-rule
element-rule	
name	modcontact
type	header-value
action	replace
comparison-type	pattern-rule
match-value	(.*)
new-value	<sip:192.65.X.X:5060>

This manipulation is to append +1 to the DID in the From header of the Invite request. It is to convert the DID in the From header into E.164 format. This manipulation is not required if the "Outbound CID" is configured as +1919XXXXXXX in the Extensions of the Asterisk FreePBX (Mandatory)

header-rule	
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

This manipulation is to append +1 to the DID in the To header of the Invite request. It is to convert the DID in the To header into E.164 format. This manipulation is not required if the Dial Pattern of the Outbound Route is configured with prepend as +1 or + in the Asterisk FreePBX (Mandatory)

header-rule

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

element-rule

name	Touser800
type	uri-user
action	replace
comparison-type	pattern-rule
match-value	(^800.*)
new-value	" +1" + \$ORIGINAL

element-rule

name	TouserINTL
type	uri-user
action	replace
comparison-type	pattern-rule
match-value	(^9199.*)
new-value	" + " + \$ORIGINAL

This manipulation is to append +1 or + to the DID in the request uri. It is to convert the DID in the Request uri into E.164 format. This manipulation is not required if the Dial Pattern of the Outbound Route is configured with prepend as +1 or + in the Asterisk FreePBX (Mandatory)

header-rule

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

element-rule

name	RequestURI800
type	uri-user
action	replace
comparison-type	pattern-rule
match-value	(^800.*)
new-value	" +1" + \$ORIGINAL

element-rule

name	RequestURI1
type	uri-user
action	replace
comparison-type	pattern-rule
match-value	(^9199.*)
new-value	" + " + \$ORIGINAL

6.10.2 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

This manipulation is to replace the IP address in the From Header with the Oracle ESBC IP address (Optional)

header-rule

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

This manipulation is to replace the IP address in the To Header with the IP address of FreePBX Asterisk (Optional)

header-rule

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

This manipulation is to replace the SDP owner name with "OracleACME" (Optional)

header-rule

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

This manipulation is to replace the Useragent name with "Oracle" (Optional)

header-rule

name

header-name

action

comparison-type

element-rule

name

type

action

comparison-type

match-value

new-value

moduseragent

User-Agent

manipulate

pattern-rule

moduseragent

header-value

replace

pattern-rule

^VineProx

Oracle

6.11 SIP Interface

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

1. Amazon Chime SDK Voice Connector
2. FreePBX Asterisk

Realm Id: Select the Realm Id

Description: Enter the Description

SIP Ports

- **Address:** Enter the IP address
- **Port:** 5060
- **Transport Protocol:** UDP

Session Timer Profile: Select the session timer profile

6.11.1 SIP Interface for Amazon Chime SDK Voice Connector

The screenshot shows the 'Modify SIP Interface' configuration page. At the top right, there are 'Show Advanced' (checked) and 'Show Configuration' buttons. The 'State' is set to 'enable'. The 'Realm ID' dropdown is set to 'ACVCrecdServer' and the 'Description' text field contains 'ACVCrecdServer'. Below this is a table for 'SIP Ports' with one entry highlighted in red:

Select	Action	Address	Port	Transport Protocol	TLS Profile	Allow Anonymous	Multi Home Addr
<input type="checkbox"/>		192.65	5060	UDP		all	

Below the table, there are several configuration fields:

- Initial Inv Trans Expire: 0 (Range: 0.2147475)
- Session Max Life Limit: 0
- Proxy Mode: (dropdown)
- Redirect Action: (dropdown)
- Nat Traversal: none (dropdown)
- 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 33: 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 34: 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 to [Section 6.7](#) 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	PCMU:-64K x 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 35: Oracle 4600 SIP Interface Amazon Chime SDK Voice Connector Continuation

Kpm12833 lwf 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	ACVC_SessionTimer_Profile ▾
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	none ▾
Atcf Icsi Match	<input type="text"/>
SIP Recursion Policy	<input type="text"/>
Asymmetric Preconditions	<input type="checkbox"/> enable
Asymmetric Preconditions Mode	send-with-nodelay ▾
Sm Icsi Match For Invite	<input type="text"/>
Sm Icsi Match For Message	<input type="text"/>
Sbhr Profile	<input type="text"/>
Ringback Trigger	none ▾
Ringback File	<input type="text"/>
Fax Continue Session	none ▾
Npfl Profile	<input type="text"/>
Hist To Div For Cause 380	inherit ▾
User Agent	<input type="text"/>
Allow Diff2833 Clock Rate Mode	disabled ▾

OK Back

Activate Windows
Go to Settings to activate Windows.

Figure 36: Oracle 4600 SIP Interface Amazon Chime SDK Voice Connector Continuation

6.11.2 SIP Interface for 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..2147475)
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: 0,1025..65535)
Port Map End	<input type="text" value="0"/>	(Range: 0,1025..65535)

Figure 37: 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		
Kpml2833 Iwf On Hairpin	<input type="checkbox"/> enable	
Msrp Delay Egress Bye	<input type="checkbox"/> enable	
Send 380 Response	<input type="text"/>	
Pescf Restoration	<input type="text"/>	
Session Timer Profile	<input type="text" value="ACVC_SessionTimer_Profile"/>	
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 38: 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

Activate Windows
Go to Settings to activate Windows.

OK Back

Figure 39: Oracle 4600 SIP Interface Asterisk Continuation

6.12 Session-Agent

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

1. Amazon Chime SDK Voice Connector
2. FreePBX Asterisk

Hostname: Enter the hostname

Port: Enter the 5060

APP Protocol: SIP

Transport Method: UDP

Realm Id: Select the Realm Id

Codec Policy: Select the Codec Policy

6.12.1 Session Agent for Amazon Chime SDK Voice Connector

Modify Session Agent Show Advanced Show Configuration

Hostname	fa2e5s3
IP Address	
Port	5060 (Range: 0,1025..65535)
State	<input checked="" type="checkbox"/> enable
App Protocol	SIP
App Type	
Transport Method	UDP
Realm ID	ACVCRecdServer
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)

Activate Windows

Figure 40: 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 41: 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 42: Oracle 4600 Session Agent Amazon Chime SDK Voice Connector Continuation

6.12.2 Session Agent for FreePBX Asterisk

Modify Session Agent
Show Advanced
Show Configuration

Hostname	<input type="text" value="172.16.0.1"/>
IP Address	<input type="text" value="172.16.0.1"/>
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	<input type="text"/>
Redirect Action	<input type="text"/>

Figure 43: Oracle 4600 Session Agent Asterisk

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Amazon Web Services

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

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"/>
Local Response Map	<input type="text"/>
Ping Response	<input type="checkbox"/> enable
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	0 (Range: 0.999999999)
Invalidate Registrations	<input type="checkbox"/> enable
Rfc2833 Mode	none
Rfc2833 Payload	0 (Range: 0.96.127)
Codec Policy	G711U
Refer Call Transfer	disabled
Refer Notify Provisional	none
Reuse Connections	NONE

Figure 44: 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 45: Oracle 4600 Session Agent Asterisk Continuation

6.13 Local Policy

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

1. FreePBX Asterisk
2. Amazon Chime SDK Voice Connector

From Address: *

To Address: *

Source Realm: Select the Realm

Next Hop: Select the Next Hop

6.13.1 Local Policy for Amazon Chime SDK Voice Connector

The screenshot shows the 'Modify Local Policy' interface. The 'From Address' and 'To Address' fields are both set to '*'. The 'Source Realm' is set to 'ACVCRecdServer'. The 'State' checkbox is checked and labeled 'enable'. The 'Parallel Forking' checkbox is unchecked. The 'Policy Priority' is set to 'none'. Below these fields is a table of '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

6.13.2 Local Policy for 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

6.14 TLS Configuration

6.14.1 Certificate Record for Oracle ESBC

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

Name: Enter a name for the Certificate Record

Common Name: Enter a 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.

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

6.14.2 Certificate Record for Amazon Chime SDK Voice Connector

Name: Enter a name for the Certificate Record

Common Name: Enter a common name for the Certificate Record

- Import Amazon's Root Certificate to the Certificate Record.
- Amazon Chime SDK Voice Connector Root Certificate can be downloaded from the 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:** (empty)
- Common Name:** Amazon Root CA 1
- Key Size:** 2048
- Alternate Name:** (empty)

Below these fields, the 'Trusted' checkbox is checked and labeled 'enable'. To the right, there is an 'Activate Windows' watermark. A sidebar on the left lists various configuration categories, with 'certificate-record' highlighted. The main content area below the sidebar shows the 'Key Usage List' configuration:

- 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:** (empty)
- Options:** (empty)

At the bottom of the sidebar, there is a 'Show All' toggle (unchecked) and 'OK' and 'Back' buttons.

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

6.14.3 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 for 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 interface. It is divided into three main sections:

- Top Section:** Contains fields for 'Name' (Amazon_TLS_Profile), 'End Entity Certificate' (Oracle_4600), 'Trusted Ca Certificates' (AmazonCert), and a 'Cipher List' with the following ciphers: 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.
- Middle Section:** Contains 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. Below this is 'Verify Depth' (10) and 'Mutual Authenticate' (disabled).
- Bottom Section:** Contains 'TLS Version' (tlsv12), 'Options' (empty), 'Cert Status Check' (disabled), 'Cert Status Profile List' (empty), 'Ignore Dead Responder' (disabled), and 'Allow Self Signed Cert' (disabled). 'OK' and 'Back' buttons are at the bottom left.

Figure 50: Oracle 4600 TLS Profile

6.14.4 SDES Profile

- Create a **sdes-profile** with the crypto supported by Amazon Chime SDK Voice Connector to enable media encryption
- Associate **sdes-profile** with the **media-sec-policy**

Name: Enter the name for the SDES Profile

Crypto List: Select the Crypto type supported by Amazon Chime SDK Voice Connector

Egress Offer Format: same-as-ingress

The screenshot shows the 'Modify Sdes Profile' configuration interface. The 'Name' field is set to 'SRTP' and the 'Crypto List' is set to 'AES_CM_128_HMAC_SHA1_80'. The 'Egress Offer Format' is set to 'same-as-ingress'. There are checkboxes for 'SrtP Auth', 'SrtP Encrypt', 'SrTCP Encrypt', and 'Mki', all of which are checked. The 'Lifetime' field is set to '0'. The page includes 'OK' and 'Back' buttons at the bottom left and an 'Activate Windows' watermark at the bottom right.

Figure 51: Oracle 4600 SDES Profile

6.14.5 Media SEC Policy for SRTP

- Associate **media-sec-policy: SRTP** with the Realm config of Amazon Chime SDK Voice Connector

Name: SRTP

Inbound

- **Profile:** SRTP
- **Mode:** srtp
- **Protocol:** sdes

Outbound

- **Profile:** SRTP
- **Mode:** srtp
- **Protocol:** sdes

Modify Media Sec Policy Show Advanced Show Configuration

Name

Pass Through enable

Options

∨ Inbound

Profile

Mode

Protocol

Hide Egress Media Update enable

∨ Outbound

Profile

Mode

Protocol

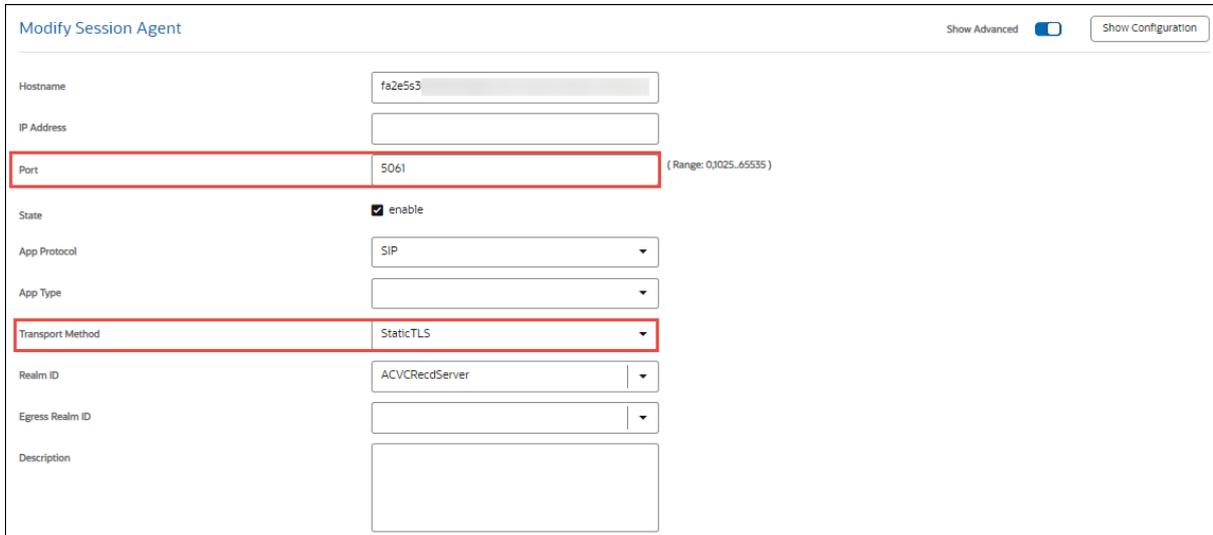
Activate Windows
Go to Settings to activate Windows.

Figure 52: Oracle 4600 Media SEC Policy for SRTP

6.14.6 Session Agent for Amazon Chime SDK Voice Connector-TLS

Port: 5061

Transport Method: Static TLS



Modify Session Agent Show Advanced Show Configuration

Hostname: fa2e5e3

IP Address:

Port: 5061 (Range: 0,1025..65535)

State: enable

App Protocol: SIP

App Type:

Transport Method: Static TLS

Realm ID: ACVCRcdServer

Egress Realm ID:

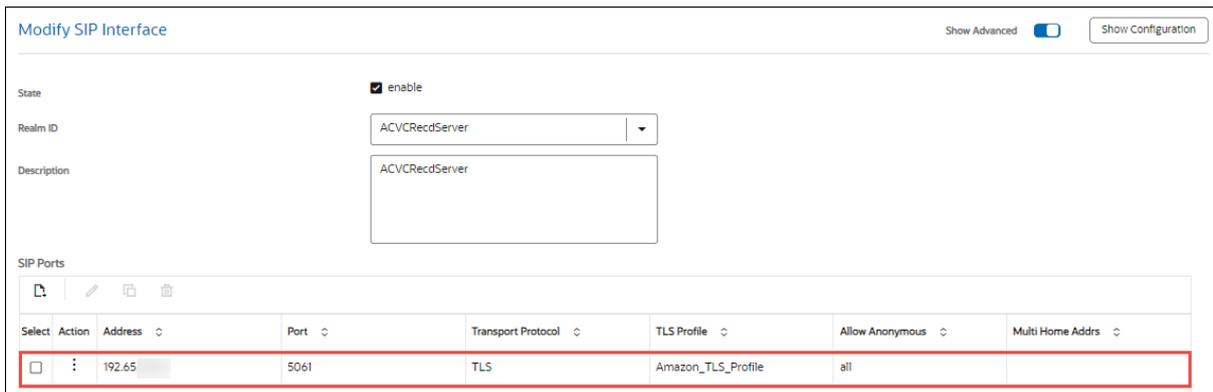
Description:

Figure 53: Oracle 4600 Session Agent Amazon Chime SDK Voice Connector-TLS

6.14.7 SIP Interface for Amazon Chime SDK Voice Connector-TLS

SIP Ports

- **Port:** 5061
- **Transport Protocol:** TLS
- **TLS Profile:** Select the TLS Profile



Modify SIP Interface Show Advanced Show Configuration

State: enable

Realm ID: ACVCRcdServer

Description: ACVCRcdServer

SIP Ports

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

Figure 54: Oracle 4600 SIP Interface Amazon Chime SDK Voice Connector-TLS

6.14.8 Realm for Amazon Chime SDK Voice Connector-TLS

Media Sec Policy: SRTP

QoS Enable	<input type="checkbox"/> enable
Max Bandwidth	<input type="text" value="0"/> (Range: 0.999999999)
Max Priority Bandwidth	<input type="text" value="0"/> (Range: 0.999999999)
Parent Realm	<input type="text"/>
DNS Realm	<input type="text"/>
Media Policy	<input type="text"/>
Nsep Media Policy	<input type="text"/>
Media Sec Policy	<input type="text" value="SRTP"/>
RTCP Mux	<input type="checkbox"/> enable

Figure 55: Oracle 4600 Realm Amazon Chime SDK Voice Connector-TLS