



# **Amazon Chime Voice Connector**

## **SIP Trunking Configuration Guide:**

### **Cisco Unified Communications Manager (CUCM) and Oracle Acme Packet 4600 (Oracle AP4600)**

**September 2019**

## Document History

<b>Rev. No.</b>	<b>Date</b>	<b>Description</b>
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1.1	October-10-2019	Minor edits based on feedback
1.2	February-3-2020	Minor edits based on feedback

## Table of Contents

1	Audience .....	4
1.1	Amazon Chime Voice Connector.....	5
2	SIP Trunking Network Components .....	6
2.1	Hardware Components.....	7
2.2	Software Requirements.....	7
3	Features .....	7
3.1	Features Supported .....	7
3.2	Features Not Supported .....	7
3.3	Features Not Tested.....	8
3.4	Caveats and Limitations.....	8
4	Configuration.....	9
4.1	Configuration Checklist .....	9
4.2	IP Address Worksheet .....	9
4.3	Cisco UCM Configuration.....	10
4.3.1	Cisco UCM Login and Version.....	10
4.3.2	Cisco UCM SIP Profile Configuration .....	10
4.3.3	Cisco UCM Device Pool Configuration .....	15
4.3.4	Media Resources.....	20
4.3.5	SIP Trunk Security Profile.....	22
4.3.6	SIP Trunk to Oracle Acme Packet 4600 .....	24
4.3.7	Route Pattern.....	28
4.4	Oracle Acme Packet 4600 Configuration.....	31
4.4.1	Network Interface .....	31
4.4.2	Codecs.....	31
4.4.3	Media Realm.....	31
4.4.4	Steering Pool.....	32
4.4.5	Local Policy .....	33
4.4.6	SIP Trunk.....	34
4.4.7	SIP Interface .....	34
4.4.8	SIP Manipulations.....	35

4.4.9 SIP-TLS .....	40
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## Table of Figures

Figure 1 Network Topology .....	6
Figure 2: Cisco UCM software version .....	10
Figure 3 Cisco UCM SIP Profile.....	11
Figure 4 Cisco UCM SIP Profile Contd.,.....	12
Figure 5 Cisco UCM SIP Profile Contd.,.....	13
Figure 6 Cisco UCM SIP Profile Contd.,.....	14
Figure 7 Cisco UCM SIP Profile Contd.,.....	14
Figure 8 Cisco UCM SIP Profile Contd.,.....	15
Figure 9 Cisco UCM Audio Codec Preference List .....	16
Figure 10 Cisco UCM Region .....	17
Figure 11 Cisco UCM Device Pool.....	18
Figure 12 Cisco UCM Device Pool Contd.,.....	19
Figure 13 Cisco UCM Device Pool Contd.,.....	20
Figure 14 Cisco UCM Media Resources Group.....	21
Figure 15 Cisco UCM Media Resources Group List .....	22
Figure 16 Cisco UCM SIP Trunk Security Profile .....	23
Figure 17 Cisco UCM SIP Trunk Security Profile Contd., .....	23
Figure 18 Cisco UCM SIP Trunk Configuration.....	24
Figure 19 Cisco UCM SIP Trunk Configuration Contd.,.....	25
Figure 20 Cisco UCM SIP Trunk Configuration Contd.,.....	26
Figure 21 Cisco UCM SIP Trunk Configuration Contd.,.....	26
Figure 22 Cisco UCM SIP Trunk Configuration Contd.,.....	27
Figure 23 Cisco UCM SIP Trunk Configuration Contd.,.....	28
Figure 24 Cisco UCM SIP Trunk Configuration Contd.,.....	28
Figure 25 Cisco UCM Route Pattern Configuration .....	29
Figure 26 Cisco UCM Route Pattern Configuration Contd.,.....	30
Figure 27 Cisco UCM Route Pattern Configuration Contd.,.....	30

## 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 SIP trunks using **Cisco Unified Communications Manager (CUCM) and Oracle Acme Packet 4600 (Oracle AP4600)** to connect to **Amazon Chime Voice Connector** for inbound and/or outbound telephony capabilities.

## 1.1 Amazon Chime Voice Connector

Amazon Chime Voice Connector is a pay-as-you-go service that enables companies to make or receive secure phone calls over the internet or AWS Direct Connect using their existing telephone system or session border controller (SBC). The service has no upfront fees, elastically scales based on demand, supports calling both landline and mobile phone numbers in over 100 countries, and gives customers the option to enable inbound calling, outbound calling, or both.

Amazon Chime Voice Connector uses the industry-standard Session Initiation Protocol (SIP). Amazon Chime 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 API. Amazon Chime Voice Connector offers cost-effective rates for inbound and outbound calls. Calls into Amazon Chime meetings, as well as calls to other Amazon Chime Voice Connector customers are at no additional cost. With Amazon Chime 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 the SIP trunk reference configuration is illustrated below and is representative of Cisco UCM with Oracle Acme Packet 4600 configuration.

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

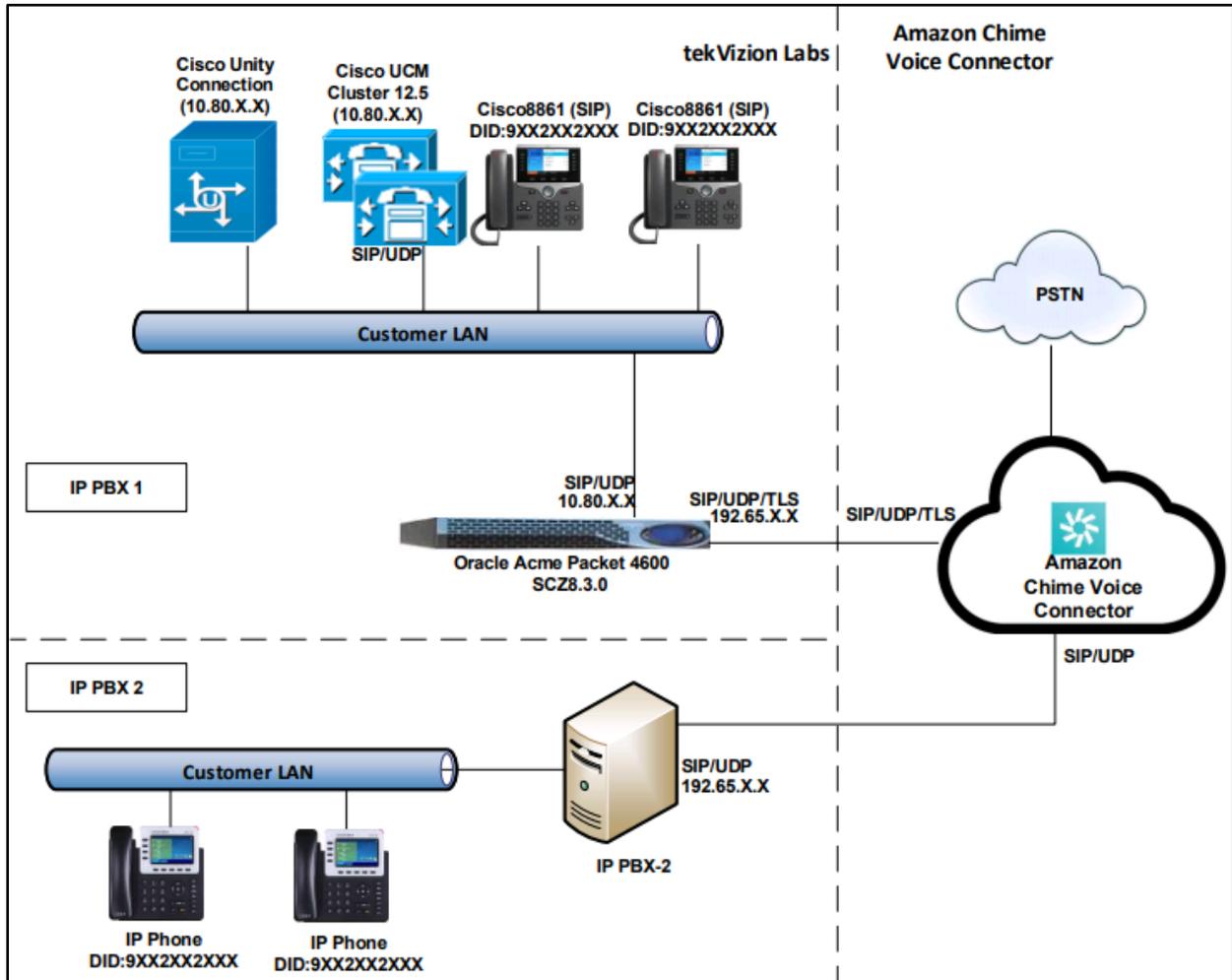


Figure 1 Network Topology

## 2.1 Hardware Components

- UCS-C240 VMWare server running ESXi 5.5 or later used for the following virtual machines
  - Cisco Unified Communications Manager (CUCM / Cisco UCM)
  - Cisco Unity Connection (CUC)
- Oracle Acme Packet 4600 (Oracle AP4600)
- Cisco IP Phone(s)-8861

## 2.2 Software Requirements

- Cisco Unified Communications Manager: 12.5.1.11900-146
- Cisco Unity Connection: 12.5.1.11900-57
- Oracle Acme Packet 4600: SCZ8.3.0 Patch 3 (Build 46)

# 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 Voice Connector,
  - Keep Alive – Double CRLF
  - Keep Alive – SIP OPTIONS

### **3.3 Features Not Tested**

- None

### **3.4 Caveats and Limitations**

- Amazon Chime 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

### 4.1 Configuration Checklist

In this section we present an overview of the steps that are required to configure **Cisco UCM and Oracle Acme Packet 4600** for SIP Trunking with **Amazon Chime Voice Connector**.

*Table 1 – PBX Configuration Steps*

Steps	Description	Reference
Step 1	Cisco UCM Configuration	<a href="#">Section 4.3</a>
Step 2	Oracle Acme Packet 4600 Configuration	<a href="#">Section 4.4</a>

### 4.2 IP Address Worksheet

The specific values listed in the table below and in 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*

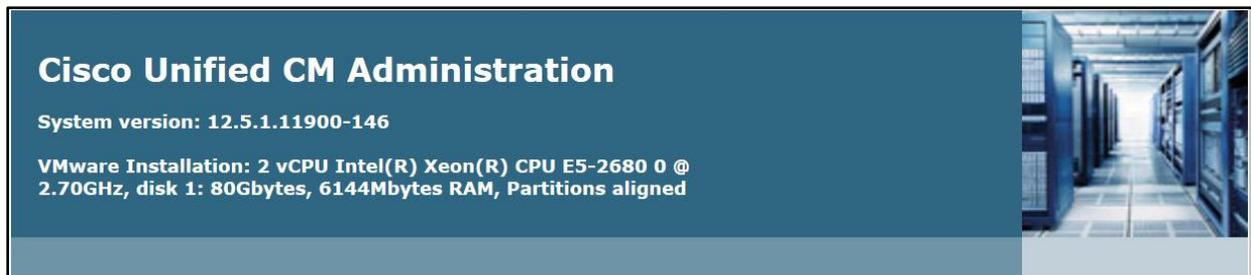
Component	Lab Value
<b>Oracle Acme Packet 4600</b>	
LAN IP Address	10.80.11.100
LAN Subnet Mask	255.255.255.0
<b>Cisco UCM</b>	
IP Address	10.80.12.2
Subnet Mask	255.255.255.0

## 4.3 Cisco UCM Configuration

This section with screenshots taken from CUCM used for the interoperability testing gives a general overview of the PBX configuration.

### 4.3.1 Cisco UCM Login and Version

Open an instance of a web browser and connect to the CUCM,  
Log in using an appropriate user ID and password. Verify the system version being tested.



*Figure 2: Cisco UCM software version*

### 4.3.2 Cisco UCM SIP Profile Configuration

1. Navigate to **Device ->Device Settings-> SIP Profile.**
2. On the screen that appears, copy the "**Standard SIP Profile**" and save the SIP Profile with the name **Standard SIP Profile – AmazonVC** and configure the SIP Profile as below.
3. Then click **Save** and then **Apply Config**

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
administrator | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

**SIP Profile Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**SIP Profile Information**

Name *	Standard SIP Profile-AmazonVC
Description	Standard SIP Profile-AmazonVC
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agent
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, and
Confidential Access Level Headers*	Disabled

- Redirect by Application
- Disable Early Media on 180
- Outgoing T.38 INVITE include audio mline
- Offer valid IP and Send/Receive mode only for T.38 Fax Relay
- Use Fully Qualified Domain Name in SIP Requests
- Assured Services SIP conformance
- Enable External QoS\*\*

Figure 3 Cisco UCM SIP Profile

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**SIP Profile Configuration** Related Links: Back To Find/List Go

Save ✖ Delete 📄 Copy 🔄 Reset 🔧 Apply Config + Add New

**SDP Information**

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites\* TIAS and AS ▾  
 SDP Transparency Profile Pass all unknown SDP attributes ▾  
 Accept Audio Codec Preferences in Received Offer\* Default ▾

Require SDP Inactive Exchange for Mid-Call Media Change  
 Allow RR/RS bandwidth modifier (RFC 3556)

**Parameters used in Phone**

Timer Invite Expires (seconds)\* 180  
 Timer Register Delta (seconds)\* 5  
 Timer Register Expires (seconds)\* 3600  
 Timer T1 (msec)\* 500  
 Timer T2 (msec)\* 4000  
 Retry INVITE\* 6  
 Retry Non-INVITE\* 10

Media Port Ranges  
 Common Port Range for Audio and Video  
 Separate Port Ranges for Audio and Video

Start Media Port\* 16384  
 Stop Media Port\* 32766

DSCP for Audio Calls Use System Default ▾  
 DSCP for Video Calls Use System Default ▾

Figure 4 Cisco UCM SIP Profile Contd.,

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation **Cisco Unified CM Administration** Go  
administrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**SIP Profile Configuration** Related Links: **Back To Find/List** Go

Save Delete Copy Reset Apply Config Add New

DSCP for Audio Portion of Video Calls	Use System Default ▾
DSCP for TelePresence Calls	Use System Default ▾
DSCP for Audio Portion of TelePresence Calls	Use System Default ▾
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None ▾
DTMF DB Level*	Nominal ▾
Call Hold Ring Back*	Off ▾
Anonymous Call Block*	Off ▾
Caller ID Blocking*	Off ▾
Do Not Disturb Control*	User ▾
Telnet Level for 7940 and 7960*	Disabled ▾
Resource Priority Namespace	< None > ▾
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

Figure 5 Cisco UCM SIP Profile Contd.,

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**SIP Profile Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

- Conference Join Enabled
- RFC 2543 Hold
- Semi Attended Transfer
- Enable VAD
- Stutter Message Waiting
- MLPP User Authorization

**Normalization Script**

Normalization Script: < None >

Enable Trace

	Parameter Name	Parameter Value	
1	<input type="text"/>	<input type="text"/>	+ -

**External Presentation Information**

Anonymous External Presentation

External Presentation Number:

External Presentation Name:

Figure 6 Cisco UCM SIP Profile Contd.,

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**SIP Profile Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on\*: Never

Resource Priority Namespace List: < None >

SIP Rel1XX Options\*: Send PRACK if 1xx Contains SDP

Video Call Traffic Class\*: Mixed

Calling Line Identification Presentation\*: Default

Session Refresh Method\*: Invite

Early Offer support for voice and video calls\*: Best Effort (no MTP inserted)

- Enable ANAT
- Deliver Conference Bridge Identifier
- Enable External Presentation Name and Number
- Reject Anonymous Incoming Calls
- Reject Anonymous Outgoing Calls
- Send ILS Learned Destination Route String
- Connect Inbound Call before Playing Queuing Announcement

Figure 7 Cisco UCM SIP Profile Contd.,

**SIP OPTIONS Ping**

Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)\*

Ping Interval for Out-of-service Trunks (seconds)\*

Ping Retry Timer (milliseconds)\*

Ping Retry Count\*

---

**SDP Information**

Send send-receive SDP in mid-call INVITE

Allow Presentation Sharing using BFCP

Allow iX Application Media

Allow multiple codecs in answer SDP

---

**i** \*- indicates required item.

**i** \*\*-. setting only takes effect if the External QoS Enabled Service Parameter is set to true.

Figure 8 Cisco UCM SIP Profile Contd.,

### 4.3.3 Cisco UCM Device Pool Configuration

#### 4.3.3.1 Codec Preference list

1. Navigate to **System** → **Region Information** → **Audio Codec Preference List**
2. Click **Add New**
3. Provide a Name and Description: **G711\_Preferred Codec List** was used in this test
4. Prioritize codecs as shown below

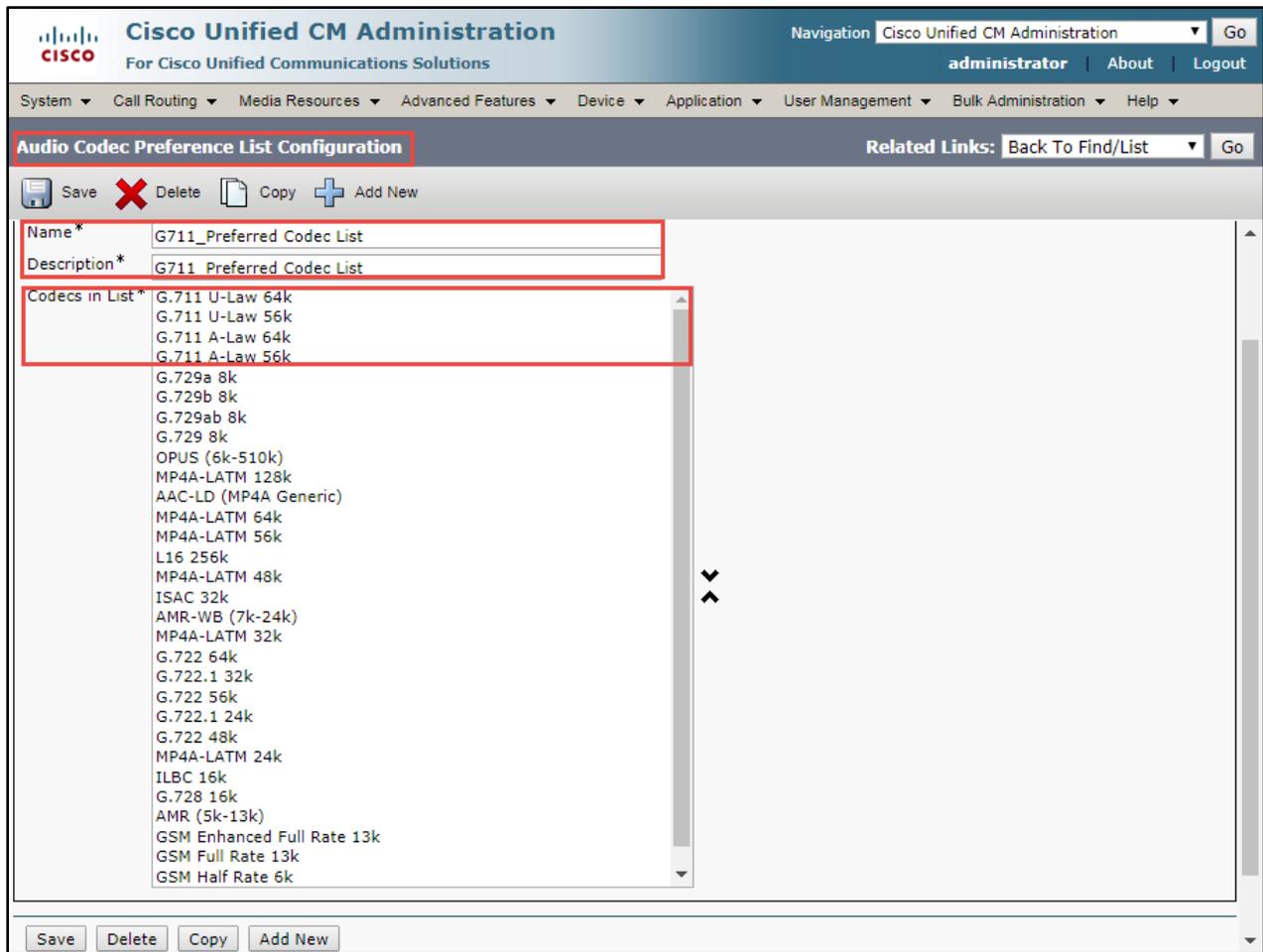


Figure 9 Cisco UCM Audio Codec Preference List

#### 4.3.3.2 New region

1. Navigate to **System** → **Region**
2. Click **Add New**
3. Provide a Name: **G711\_Region** was used in this test
4. Associate the codec preference list **G711\_Preferred Codec List** to this Region

The screenshot displays the Cisco Unified CM Administration interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user role "administrator". Below this is a menu bar with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Region Configuration" and includes a "Related Links" section with "Back To Find/List". A toolbar contains icons for Save, Delete, Reset, Apply Config, and Add New. The "Region Information" section shows a text input field for "Name\*" containing "G711\_Region". Below this is a "Region Relationships" table with the following data:

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	Default	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
G711_Region	G711_PREFERRED Codec List	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)

Figure 10 Cisco UCM Region

#### 4.3.3.3 Device Pool

1. Navigate to **System** → **Device Pool**
2. Click **Add New**
3. Provide a Device Pool Name: **G711\_pool** was used in this test
4. Associate the Region: **G711\_Region** to this Device Pool
5. Associate the Media resource Group List: **MRGL\_SW\_No\_MTP**
6. Leave all other parameters at their default settings
7. Click **Save**

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
administrator | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

**Device Pool Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Device Pool Settings**

Device Pool Name*	G711_pool
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	< None >
Adjunct CSS	< None >
Reverted Call Focus Priority	Default
Intercompany Media Services Enrolled Group	< None >
MRA Service Domain	< None >

**Roaming Sensitive Settings**

Date/Time Group*	CMLocal
Region*	G711_Region
Media Resource Group List	MRGL_SW_No_MTP
Location	< None >
Network Locale	< None >
SRST Reference*	Disable
Connection Monitor Duration***	
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >
Wireless LAN Profile Group	< None > <a href="#">View Details</a>

**Local Route Group Settings**

Standard Local Route Group	< None >
----------------------------	----------

Figure 11 Cisco UCM Device Pool

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

administrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**Device Pool Configuration** Related Links: [Back To Find/List](#)

**Device Mobility Related Information\*\*\*\***

Device Mobility Calling Search Space   
 AAR Calling Search Space   
 AAR Group   
 Calling Party Transformation CSS   
 Called Party Transformation CSS

**Geolocation Configuration**

Geolocation   
 Geolocation Filter

**Call Routing Information**

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
International Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>

Figure 12 Cisco UCM Device Pool Contd.,

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
administrator | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

**Device Pool Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Incoming Called Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >

**Phone Settings**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS < None >

**Connected Party Settings**

Connected Party Transformation CSS < None >

**Redirecting Party Settings**

Redirecting Party Transformation CSS < None >

Save Delete Copy Reset Apply Config Add New

**i** \*- indicates required item.

Figure 13 Cisco UCM Device Pool Contd.,

## 4.3.4 Media Resources

### 4.3.4.1 Media Resources Group

1. Navigate to Media Resources -> Media Resource Group.
2. Add New.
3. Provide a Name: **MRG With SW\_NOMTP** was used in this test
4. Select Media Resources from the Available Media Resources

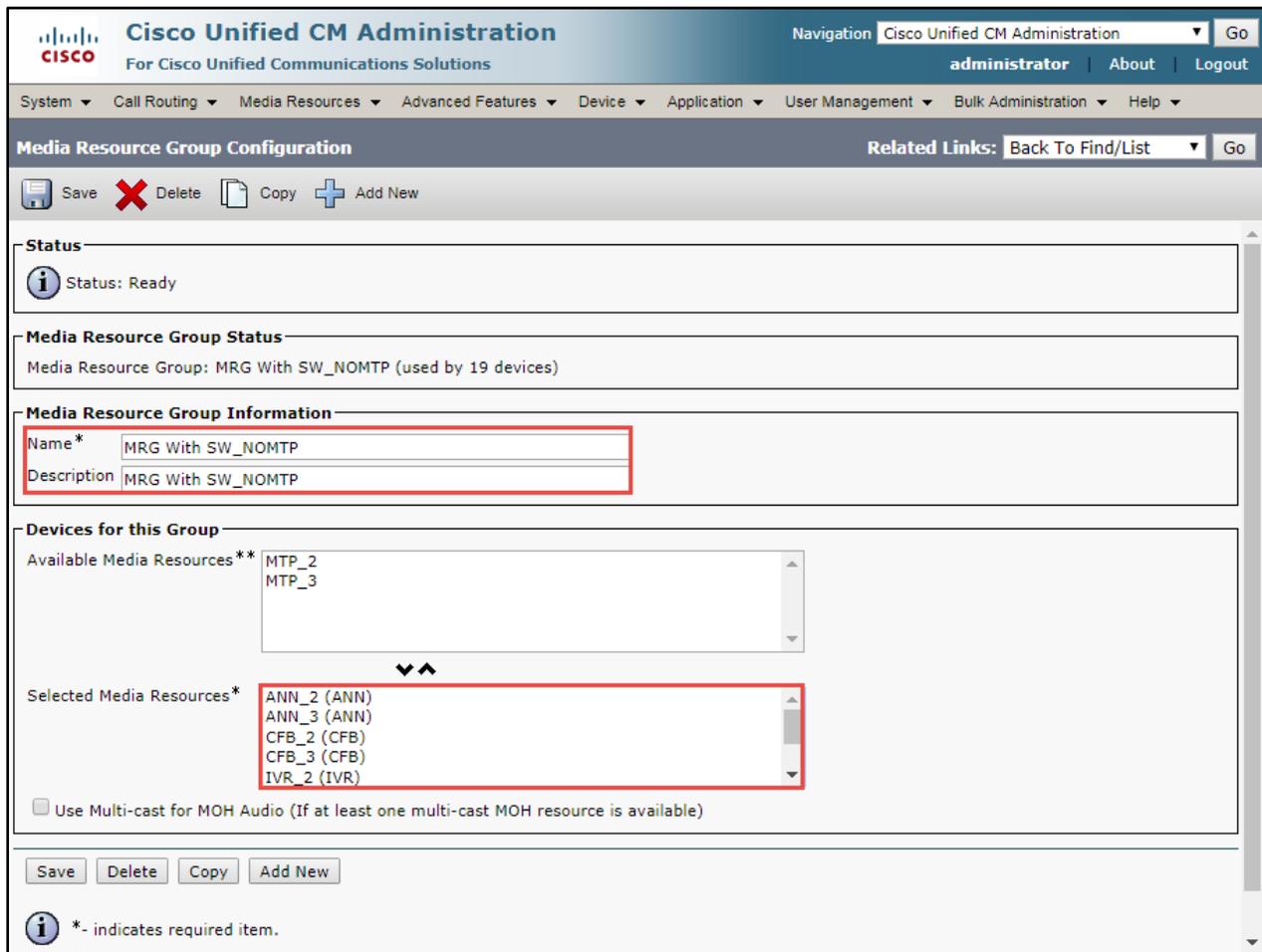


Figure 14 Cisco UCM Media Resources Group

#### 4.3.4.2 Media Resources Group List

1. Navigate to **Media Resources** -> **Media Resource Group List**
2. **Add New**
3. Provide a Name: **MRGL\_SW\_No\_MTP** was used in this test
4. Select the media resource group from the list of Available Media Resource Groups
5. Click on **Save**

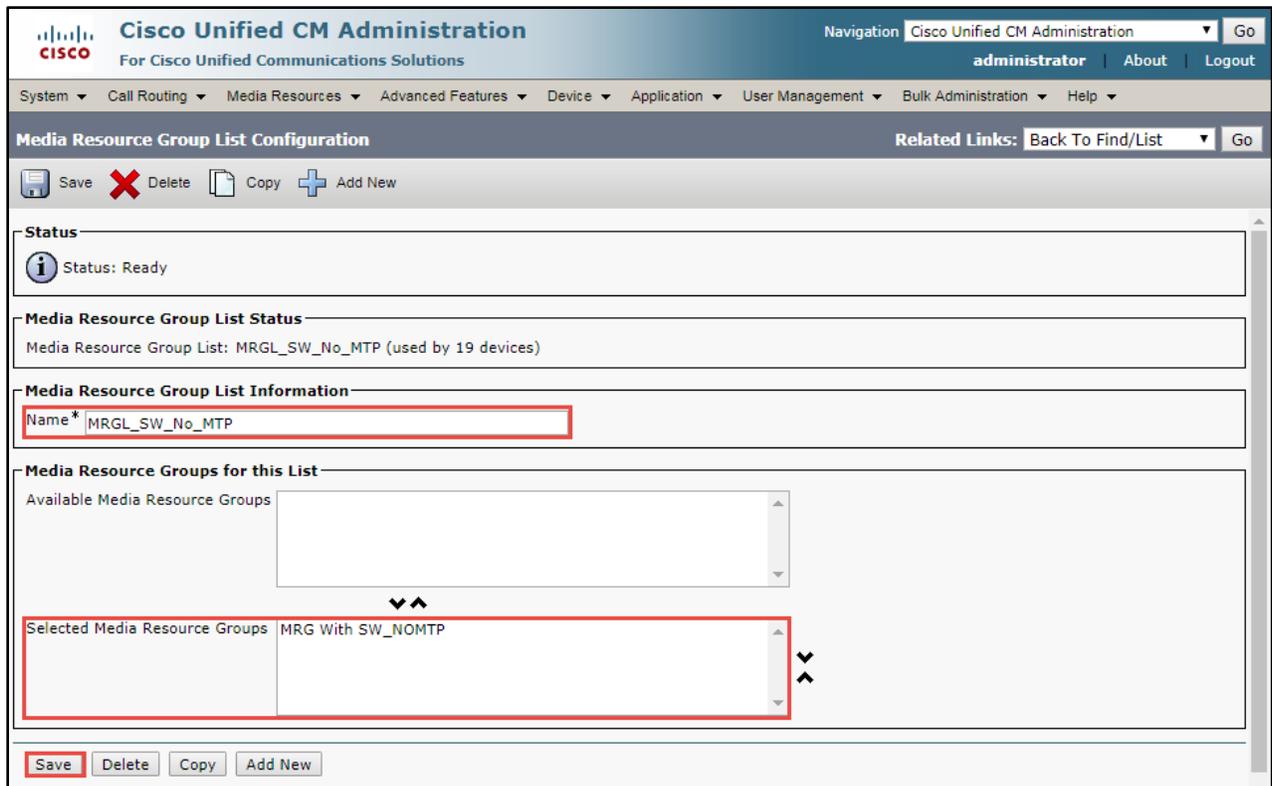


Figure 15 Cisco UCM Media Resources Group List

#### 4.3.5 SIP Trunk Security Profile

1. Navigate to: **System**→**Security**→ **Non Secure SIP Trunk Profile**
2. Provide a Name: **Non Secure SIP Trunk Profile-Amazon VC** was used for this test
3. Select Incoming Transport Type: **TCP+UDP** was used in this test
4. Select Outgoing Transport Type: **UDP** was used in this test
5. Click **Save**

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

### SIP Trunk Security Profile Configuration

Related Links: [Back To Find/List](#) Go

Save  Delete  Copy  Reset  Apply Config  Add New

**Status**  
Status: Ready

**SIP Trunk Security Profile Information**

Name*	Non Secure SIP Trunk Profile-AmazonVC
Description	Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	UDP

Enable Digest Authentication  
 Nonce Validity Time (mins)\*: 600  
 Secure Certificate Subject or Subject Alternate Name:

Incoming Port\*: 5060

Figure 16 Cisco UCM SIP Trunk Security Profile

Enable Application level authorization

Accept presence subscription

Accept out-of-dialog refer\*\*

Accept unsolicited notification

Accept replaces header

Transmit security status

Allow charging header

SIP V.150 Outbound SDP Offer Filtering\* Use Default Filter

Save

Figure 17 Cisco UCM SIP Trunk Security Profile Contd.,

### 4.3.6 SIP Trunk to Oracle Acme Packet 4600

1. Navigate to **Device**→ **Trunk**
2. Provide a **Device Name**: Amazon\_SIPTrunk\_To\_OracleACME
3. Provide a **Description**: Amazon\_SIPTrunk\_To\_OracleACME
4. Set **Device Pool**: G711\_pool
5. Set **Destination Address**: Set IP address of Oracle AP4600
6. Set **SIP Trunk Security Profile**: Non Secure SIP Trunk Profile-AmazonVC
7. Set **SIP Profile**: Standard SIP Profile – AmazonVC
8. Set **DTMF Signaling Method**: RFC2833

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and it includes navigation menus for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The "Device Information" section is highlighted with a red box and contains the following configuration details:

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Amazon_SIPTrunk_To_OracleACME
Description	Amazon_SIPTrunk_To_OracleACME
Device Pool*	G711_pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_SW_No_MTP
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

Below the table, there are several checkboxes for additional configuration options:

- Media Termination Point Required
- Retry Video Call as Audio
- Path Replacement Support
- Transmit UTF-8 for Calling Party Name
- Transmit UTF-8 Names in QSIG APDU
- Unattended Port
- SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so

Figure 18 Cisco UCM SIP Trunk Configuration

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

administrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**Trunk Configuration** Related Links: [Back To Find/List](#)

SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.  
 Consider Traffic on This Trunk Secure\*  ▾  
 Route Class Signaling Enabled\*  ▾  
 Use Trusted Relay Point\*  ▾  
 PSTN Access  
 Run On All Active Unified CM Nodes

**Intercompany Media Engine (IME)**

E.164 Transformation Profile  ▾

**MLPP and Confidential Access Level Information**

MLPP Domain  ▾  
 Confidential Access Mode  ▾  
 Confidential Access Level  ▾

**Call Routing Information**

Remote-Party-Id  
 Asserted-Identity  
 Asserted-Type\*  ▾  
 SIP Privacy\*  ▾  
 Trust Received Identity\*  ▾

Figure 19 Cisco UCM SIP Trunk Configuration Contd.,

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

**Trunk Configuration** Related Links: Back To Find/List Go

Save Delete Reset Add New

**Inbound Calls**

Significant Digits\* All  
 Connected Line ID Presentation\* Default  
 Connected Name Presentation\* Default  
 Calling Search Space < None >  
 AAR Calling Search Space < None >  
 Prefix DN

Redirecting Diversion Header Delivery - Inbound

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

Figure 20 Cisco UCM SIP Trunk Configuration Contd.,

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

**Trunk Configuration** Related Links: Back To Find/List Go

Save Delete Reset Add New

**Incoming Called Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Connected Party Settings**

Connected Party Transformation CSS < None >

Use Device Pool Connected Party Transformation CSS

Figure 21 Cisco UCM SIP Trunk Configuration Contd.,

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Trunk Configuration** Related Links: Back To Find/List Go

Save Delete Reset Add New

**Outbound Calls**

Called Party Transformation CSS < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS

Calling Party Selection\* Originator

Calling Line ID Presentation\* Default

Calling Name Presentation\* Default

Calling and Connected Party Info Format\* Deliver DN only in connected party

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

Use Device Pool Redirecting Party Transformation CSS

**Presentation Information**

Anonymous Presentation

Presentation Number

Presentation Name

Send Presentation Name and Number only in the FROM header and not in the other identity headers

Figure 22 Cisco UCM SIP Trunk Configuration Contd.,

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**Trunk Configuration** Related Links: Back To Find/List Go

Save Delete Reset Add New

**SIP Information**

**Destination**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	10.80.11.100		5060

MTP Preferred Originating Codec\* 711ulaw  
BLF Presence Group\* Standard Presence group  
SIP Trunk Security Profile\* Non Secure SIP Trunk Profile-AmazonVC  
Rerouting Calling Search Space < None >  
Out-Of-Dialog Refer Calling Search Space < None >  
SUBSCRIBE Calling Search Space < None >  
SIP Profile\* Standard SIP Profile-AmazonVC [View Details](#)  
DTMF Signaling Method\* RFC 2833

**Normalization Script**

Normalization Script < None >  
 Enable Trace

	Parameter Name	Parameter Value
1		

Figure 23 Cisco UCM SIP Trunk Configuration Contd.,

**Recording Information**

None  
 This trunk connects to a recording-enabled gateway  
 This trunk connects to other clusters with recording-enabled gateways

**Geolocation Configuration**

Geolocation < None >  
Geolocation Filter < None >  
 Send Geolocation Information

Save Delete Reset Add New

Figure 24 Cisco UCM SIP Trunk Configuration Contd.,

### 4.3.7 Route Pattern

1. Navigate to **Call Routing** -> **Route/Hunt**-> **Route Pattern**
2. Select **Add New** to create a new Route Pattern
3. The route pattern "9.XXXXXXXXXX" was configured to enable outbound dialing from CUCM to PSTN using the access code as "9".
4. Set **Gateway/Route List**: Amazon\_SIPTrunk\_To\_OracleACME

5. Set **Discard Digits**: *PreDot* was used in this test (configure this option to remove the prefix '97' from called party number while sending the call out to Oracle Acme Packet 4600)
6. Click on **Save**

The screenshot displays the Cisco Unified CM Administration interface for Route Pattern Configuration. The page title is "Route Pattern Configuration" and it includes a navigation menu at the top. The main content area is divided into sections: "Status" (Ready) and "Pattern Definition". The "Pattern Definition" section contains the following fields:

- Route Pattern\***: 9.XXXXXXXXXX
- Route Partition**: < None >
- Description**: AmazonRP\_NationalCalls
- Numbering Plan**: -- Not Selected --
- Route Filter**: < None >
- MLPP Precedence\***: Default
- Apply Call Blocking Percentage
- Resource Priority Namespace Network Domain**: < None >
- Route Class\***: Default
- Gateway/Route List\***: Amazon\_SIPTrunk\_To\_OracleACME (with an [\(Edit\)](#) link)
- Route Option**:
  - Route this pattern
  - Block this pattern No Error
- Call Classification\***: OffNet
- External Call Control Profile**: < None >
- Allow Device Override
- Provide Outside Dial Tone
- Allow Overlap Sending
- Urgent Priority
- Require Forced Authorization Code
- Authorization Level\***: 0
- Require Client Matter Code

Figure 25 Cisco UCM Route Pattern Configuration

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
administrator | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

**Route Pattern Configuration** Related Links: Back To Find/List Go

Save Delete Copy Add New

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\* Default

Calling Name Presentation\* Default

Calling Party Number Type\* Cisco CallManager

Calling Party Numbering Plan\* Cisco CallManager

**Connected Party Transformations**

Connected Line ID Presentation\* Default

Connected Name Presentation\* Default

**Called Party Transformations**

Discard Digits PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\* Cisco CallManager

Called Party Numbering Plan\* Cisco CallManager

Figure 26 Cisco UCM Route Pattern Configuration Contd.,

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol -- Not Selected --

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	<input type="text"/>

Save Delete Copy Add New

Figure 27 Cisco UCM Route Pattern Configuration Contd.,

## 4.4 Oracle Acme Packet 4600 Configuration

### 4.4.1 Network Interface

#### WAN Ethernet Interface

- **ip-address:** 192.65.XX.XX (IP address is based on customer's network)
- **dns-ip-primary:** 8.8.8.8 (Public DNS Server which can resolve the Outbound hostname provided by Amazon Chime Voice Connector)
- **dns-domain:** Outbound\_HostName from Amazon Chime Voice Connector

```
network-interface
  name                s0p0
  description         WAN network Interface
  ip-address          192.65.XX.XX
  netmask             255.255.XXX.XXX
  gateway             192.65.XX.XX
  dns-ip-primary      8.8.8.8
  dns-domain          <Outbound HostName
                    From Amazon Chime
                    Voice Connector>
  hip-ip-list         192.65.XX.XX
  icmp-address        192.65.XX.XX
```

#### LAN Ethernet Interface

- **ip-address:** 10.80.11.100 (IP address is based on customer's network)

```
network-interface
  name                s0p2
  description         LAN network Interface
  ip-address          10.80.11.100
  netmask             255.255.255.0
  gateway             10.80.11.1
  hip-ip-list         10.80.11.100
  icmp-address        10.80.11.100
```

### 4.4.2 Codecs

- **allow-codecs:** PCMU and telephone-event

```
codec-policy
  name                G711
  allow-codecs        PCMU telephone-event
```

### 4.4.3 Media Realm

#### Realm for PBX

- **realm identifier** pointing to Cisco UCM: LAN\_To\_CUCM
- Associate the LAN side **network-interface** : s0p2:0.4
- Associate **out-manipulationid**: To\_CiscoUCM
- Associate **codec-policy**: G711

```

realm-config
  identifier                LAN_To_CUCM
  description               LAN_To_CUCM
  network-interfaces       s0p2:0.4
  out-translationid        InboundCall
  out-manipulationid       To_CiscoUCM
  codec-policy              G711

```

### Realm for Amazon Chime Voice Connector

- **realm identifier** pointing to Amazon Chime Voice Connector: WAN\_To\_AmazonVC
- Associate the WAN side **network-interface** : s0p0:0.4
- Associate **out-manipulationid**: AmazonVC
- Associate the **codec-policy**: G711

```

realm-config
  identifier                WAN_To_AmazonVC
  description               WAN_To_AmazonVC
  network-interfaces       s0p0:0.4
  out-manipulationid       AmazonVC
  codec-policy              G711

```

#### 4.4.4 Steering Pool

##### Steering Pool for Cisco UCM

- Assign the LAN side **IP address**: 10.80.11.100
- Associate the **realm-id** : LAN\_To\_CUCM
- Associate the LAN side **network-interface**: s0p2:0.4

```

steering-pool
  ip-address                10.80.11.100
  start-port                49152
  end-port                  65535
  realm-id                  LAN_To_CUCM
  network-interface         s0p2:0.4

```

##### Steering Pool for Amazon Chime Voice Connector

- Assign the WAN side **IP address**: 192.65.XX.XX
- Associate the **realm-id** : WAN\_To\_AmazonVC
- Associate the LAN side **network-interface**: s0p0:0.4

```

steering-pool
  ip-address          192.65.XX.XX
  start-port         49152
  end-port           65535
  realm-id           WAN_To_AmazonVC
  network-interface  s0p0:0.4

```

#### 4.4.5 Local Policy

##### Local Policy for Cisco UCM

- **from address:** \*
- **to-address:** \*
- Associate **source-realm** with the realm-id: LAN\_To\_CUCM
- Associate **Policy-attribute -> next-hop** with the realm-id: WAN\_To\_AmazonVC

```

local-policy
  from-address      *
  to-address        *
  source-realm      LAN_To_CUCM
  description       LAN_To_CUCM
  activate-time
  deactivate-time
  state             enabled
  policy-priority   none
  policy-attribute
    next-hop        <Session_Agent_Amazon
                   Chime Voice Connector>
                   realm      WAN_To_AmazonVC
                   action     none
                   state      enabled

```

##### Local Policy for Amazon Chime Voice Connector

- **from address:** \*
- **to-address:** \*
- Associate **source-realm** with the realm-id: WAN\_To\_AmazonVC
- Associate **Policy-attribute -> next-hop** with the realm-id: 10.80.12.2

```

local-policy
  from-address      *
  to-address        *
  source-realm      WAN_To_AmazonVC
  description       WAN_To_AmazonVC
  activate-time
  deactivate-time
  state             enabled

```

```

policy-priority          none
policy-attribute
  next-hop               10.80.12.2
  realm                  LAN_To_CUCM
  action                 none
  state                  enabled

```

#### 4.4.6 SIP Trunk

##### Cisco UCM Trunk Configuration

- **hostname & ip-address:** 10.80.12.2 (IP address of Cisco UCM)
- **realm-id:** LAN\_To\_CUCM
- **rfc2833-mode:** preferred
- **rfc2833-payload:** 101
- **codec-policy:** G711

```

session-agent
  hostname               10.80.12.2
  ip-address             10.80.12.2
  realm-id               LAN_To_CUCM
  rfc2833-mode           preferred
  rfc2833-payload        101
  codec-policy           G711

```

##### Amazon Chime Voice Connector Trunk Configuration

- **hostname & ip-address:** 10.80.12.2 (IP address of Cisco UCM)
- **realm-id:** WAN\_To\_AmazonVC
- **ping-method:** OPTIONS
- **rfc2833-mode:** preferred
- **rfc2833-payload:** 101
- **codec-policy:** G711

```

session-agent
  hostname               <Outbound_HostName
                        from Amazon Chime
                        Voice Connector>
  realm-id               WAN_To_AmazonVC
  ping-method            OPTIONS
  ping-interval          60
  rfc2833-mode           preferred
  rfc2833-payload        101
  codec-policy           G711

```

#### 4.4.7 SIP Interface

- **Realm-id:** Realm for Amazon Chime Voice Connector
- **Address:** WAN IP address assigned to the device
- **Port:** 5060
- **transport-protocol:** UDP

```

sip-interface
  state                enabled
  realm-id             WAN_To_AmazonVC
  description          WAN_To_AmazonVC
  sip-port
    address            192.65.XX.XX
    port               5060
    transport-protocol UDP

```

#### 4.4.8 SIP Manipulations

##### SIP Manipulations for Amazon Chime 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 Voice Connector

```

sip-manipulation
  name                 AmazonVC
  description          AmazonVC
  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     ip
    new-value          <Outbound_HostName
                      from Amazon Chime
                      Voice Connector>

```

```

header-rule
  name                 ToHost
  header-name          to
  action              manipulate
  msg-type            request
  methods            INVITE
  element-rule
    name               tohost
    type               uri-host

```

action  
match-val-type  
new-value

replace  
ip  
<Outbound\_HostName  
from Amazon Chime  
Voice Connector>

**header-rule**

name  
header-name  
action  
msg-type  
element-rule  
name  
type  
action  
match-val-type  
new-value

**pai**  
**P-Asserted-Identity**  
manipulate  
request

pai  
uri-host  
replace  
ip  
<Outbound\_HostName  
from Amazon Chime  
Voice Connector>

**header-rule**

name  
header-name  
action  
comparison-type  
methods  
element-rule  
name  
type  
action  
comparison-type  
match-value  
new-value

**paiuser**  
**P-Asserted-Identity**  
manipulate  
pattern-rule  
INVITE

**paiheaderuser**  
**uri-user**  
replace  
pattern-rule  
(^919.\*)  
"+1"+\$ORIGINAL

**header-rule**

name  
header-name  
action  
msg-type  
element-rule  
name  
type  
action  
match-val-type  
new-value

**pairresponse**  
**P-Asserted-Identity**  
manipulate  
reply

pai  
uri-host  
replace  
ip  
\$LOCAL\_IP

**header-rule**

name  
header-name  
action  
element-rule  
name

**modsdowner**  
**Content-type**  
manipulate

**changeowner**

**parameter-name**  
type  
action  
match-value  
new-value

**application/sdp**  
mime  
find-replace-all  
CiscoSystemsCCM-SIP  
OracleACME

**header-rule**  
name  
header-name  
action  
element-rule  
name  
parameter-name  
type  
action

**removeremoteparty**  
Remote-Party-ID  
delete  
  
removeremoteparty  
Remote-Party-ID  
header-param-name  
delete-header

**header-rule**  
name  
header-name  
action  
comparison-type  
msg-type  
methods  
UPDATE  
element-rule  
name  
type  
action  
comparison-type  
match-value  
new-value

**modifyuseragent**  
User-Agent  
manipulate  
pattern-rule  
request  
ACK, BYE, INVITE, PRACK,

**modua**  
header-value  
replace  
pattern-rule  
^Cisco(.\*)  
OracleE\SBC/SCZ830

**header-rule**  
name  
header-name  
action  
comparison-type  
match-value

**removesupported**  
Supported  
delete  
pattern-rule  
X-cisco-srtp-fallback

**header-rule**  
name  
header-name  
action

**removeciscoguid**  
Cisco-Guid  
delete

**header-rule**  
name

**removecallinfo**

**header-name**  
action

**Call-Info**  
delete

**header-rule**

**name**  
**header-name**  
action  
comparison-type  
**element-rule**  
**name**  
**type**  
action  
comparison-type  
match-value  
new-value

**modcontact**  
**Contact**  
manipulate  
pattern-rule

**modcontact**  
**header-value**  
replace  
pattern-rule  
(.\*)  
sip:192.65.79.204:5060

**header-rule**

**name**  
**header-name**  
action

**removeserver**  
**Server**  
delete

**header-rule**

**name**  
**header-name**  
action  
comparison-type  
msg-type  
methods  
**element-rule**  
**name**  
**type**  
action  
comparison-type  
match-value  
new-value

**fromuser**  
**from**  
manipulate  
pattern-rule  
request  
INVITE

**fromuser**  
**uri-user**  
replace  
pattern-rule  
(^919.\*)  
"+1"+\$ORIGINAL

**header-rule**

**name**  
**header-name**  
action  
comparison-type  
msg-type  
methods  
**element-rule**  
**name**  
**type**  
action  
comparison-type  
match-value  
new-value  
**element-rule**

**ToUser**  
**To**  
manipulate  
pattern-rule  
request  
INVITE

**Touser**  
**uri-user**  
replace  
pattern-rule  
(^214.\*)  
"+1"+\$ORIGINAL

<pre> name type action comparison-type match-value new-value element-rule name type action comparison-type match-value new-value </pre>	<pre> Touser800 uri-user replace pattern-rule (^800.*) "+1"+\$ORIGINAL  TouserINTL uri-user replace pattern-rule (^9197.*) "+"+\$ORIGINAL </pre>
<pre> header-rule name header-name action comparison-type msg-type element-rule name type action comparison-type match-value new-value element-rule name type action comparison-type match-value new-value element-rule name type action comparison-type match-value new-value </pre>	<pre> RequestURI request-uri manipulate pattern-rule request  RequestURI uri-user replace pattern-rule (^214.*) "+1"+\$ORIGINAL  RequestURI800 uri-user replace pattern-rule (^800.*) "+1"+\$ORIGINAL  RequestURIINTL uri-user replace pattern-rule (^9197.*) "+"+\$ORIGINAL </pre>

## SIP Manipulations for Cisco UCM Trunk

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

<pre> sip-manipulation name description </pre>	<pre> To_CiscoUCM To_CiscoUCM </pre>
--	--------------------------------------

**header-rule**  
name  
header-name  
action  
msg-type  
element-rule  
name  
type  
action  
new-value

**fromhost**  
From  
manipulate  
request  
  
fromhost  
uri-host  
replace  
\$LOCAL\_IP

**header-rule**  
name  
header-name  
action  
msg-type  
element-rule  
name  
type  
action  
new-value

**tohost**  
To  
manipulate  
request  
  
tohost  
uri-host  
replace  
\$REMOTE\_IP

**header-rule**  
name  
header-name  
action  
element-rule  
name  
parameter-name  
type  
action  
match-value  
new-value

**modsdpowner**  
Content-type  
manipulate  
  
modsdpowner  
application/sdp  
mime  
find-replace-all  
Sonus\_UAC  
OracleACME

**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

#### 4.4.9 SIP-TLS

#### 4.4.9.1 Certificate Record for Oracle AP4600 Device

- Create a **certificate record** for the device
- Generate a **Certificate Signing Request(CSR)**
- Sign the certificate using the Certificate Authority
- Import the signed certificate to the **certificate record**

```
certificate-record
  name                Oracle_4600
  state               NA
  locality            NA
  organization        NA
  unit               NA
  common-name        IP:192.65.XX.XX
```

#### 4.4.9.2 Certificate Record for Amazon Chime Voice Connector

- Create a certificate record for Amazon Chime Voice Connector
- Download the certificate authority bundle from Amazon Chime Voice Connector portal
- Import the certificate bundle to the certificate record created

```
certificate-record
  name                AmazonVC
  state               NA
  locality            NA
  organization        Amazon
  common-name        Amazon Root CA 1
  extended-key-usage-list
                    serverAuth
                    clientAuth
```

#### 4.4.9.3 SDES Profile

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

```
sdes-profile
  name                SRTP
  crypto-list         AES_CM_128_HMAC_SHA1_80
  srtp-auth          enabled
  srtp-encrypt       enabled
  rtp-encrypt        enabled
  mki                disabled
  egress-offer-format
                    same-as-ingress
  use-ingress-session-params
  options
```

```

key
salt
srtp-rekey-on-re-invite           disabled
lifetime                           0

```

#### 4.4.9.4 Media SEC Policy

- Create a **media-sec-policy** for **RTP** and **SRTP**
- Associate **media-sec-policy: RTP** with the Realm config of Cisco UCM
- Associate **media-sec-policy: SRTP** with the Realm config of Amazon Chime Voice Connector

```

media-sec-policy
  name                               RTP
media-sec-policy
  name                               SRTP
  inbound
    profile                           SRTP
    mode                               srtp
    protocol                           sdes
  outbound
    profile                           SRTP
    mode                               srtp
    protocol                           sdes

```

#### 4.4.9.5 Session Agent

- **hostname**: Outbound Hostname provided by Amazon Chime Voice Connector
- **transport-method** : StaticTLS
- **port** : 5061

```

session-agent
  hostname                           <Outbound_HostName
                                     from Amazon Chime
                                     Voice Connector>
  port                               5061
  transport-method                   StaticTLS
  realm-id                           WAN_To_AmazonVC
  ping-method                         OPTIONS
  ping-interval                       60
  codec-policy                        G711

```

#### 4.4.9.6 TLS Profile

- **end-entity-certificate** : Certificate record created for the Device
- **trusted-ca-certificate**: Certificate record created Amazon Chime Voice Connector

```

tls-profile
  name                AmazonTLS_Profile
  end-entity-certificate Oracle_4600
  trusted-ca-certificates AmazonVC

```

#### 4.4.9.7 SIP Interface

- **Realm-id:** Realm for Amazon Chime Voice Connector
- **Address:** WAN IP address assigned to the device
- **Port:** 5061
- **transport-protocol:** TLS
- **tls-profile:** Associate the TLS profile created earlier

```

sip-interface
  realm-id            WAN_To_AmazonVC
  description         WAN_To_AmazonVC
  sip-port
  address             192.65.XX.XX
  port                5061
  transport-protocol TLS
  tls-profile         AmazonTLS_Profile

```