



Amazon Chime Voice Connector

SIP Trunking Configuration Guide:

Cisco Unified Communications Manager (CUCM) and Cisco Unified Border Element (CUBE)

August 2019

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1 Audience

This document is intended for technical staff and Value Added Resellers (VAR) with installation and operational responsibilities. This configuration guide provides steps for configuring SIP trunks using **Cisco Unified Communications Manager (CUCM) and Cisco Unified Border Element (CUBE)** 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 Cisco UBE configuration.

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

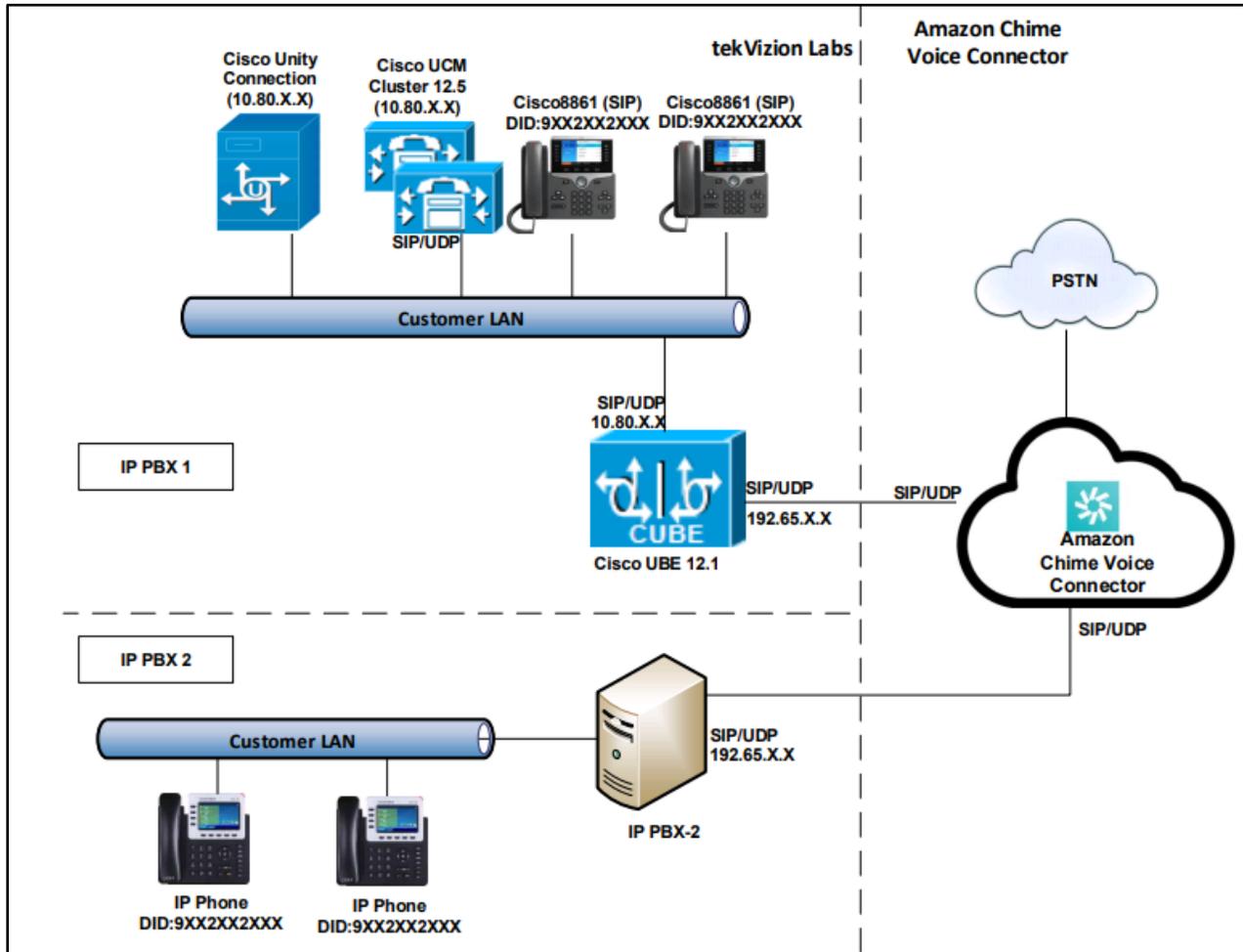


Figure 1 Network Topology

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 Unity Connection (CUC)
- Cisco UBE (CUBE) on Cisco ISR 4431 router
- Cisco IP Phone(s)-8861

2.2 Software Requirements

- Cisco UCM : 12.5.1.11900-146
- Cisco Unity Connection: 12.5.1.11900-57
- Cisco UBE: 12.1.0 running on IOS-XE 16.09.03(isr4400-universalk9.16.09.03.SPA.bin)

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
- 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,
 - Mutual TLS
 - Keep Alive – SIP OPTIONS
- Keep Alive – Double CRLF are not supported by Amazon Chime Voice Connector and Cisco UBE

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.
- Amazon Chime Voice Connector does not support Mutual TLS which is required for secure trunking with CUBE. Encrypted signaling and media with SRTP has not been tested.

4 Configuration

4.1 Configuration Checklist

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

Table 1 – PBX Configuration Steps

Steps	Description	Reference
Step 1	Cisco UCM Configuration	Section 4.3
Step 2	Cisco UBE Configuration	Section 4.4

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
Cisco UBE	
LAN IP Address	10.80.11.18
LAN Subnet Mask	255.255.255.0
Cisco UCM	
IP Address	10.80.12.2
Subnet Mask	255.255.255.0

4.3 Cisco UCM Configuration

This section with screen shots 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.

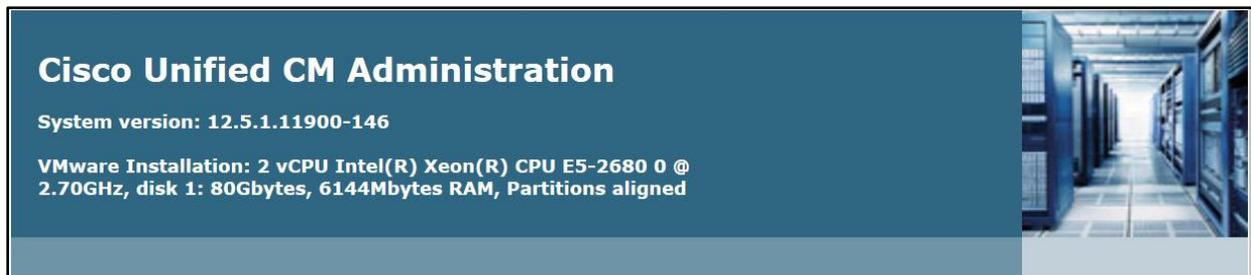


Figure 2: Cisco UCM software version

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
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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

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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.,

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SIP Profile Configuration

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

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

Figure 5 Cisco UCM SIP Profile Contd.,

Cisco Unified CM Administration For Cisco Unified Communications Solutions

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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

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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* Disabled

Video Call Traffic Class* Mixed

Calling Line Identification Presentation* Default

Session Refresh Method* Invite

Early Offer support for voice and video calls* Disabled (Default value)

- 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

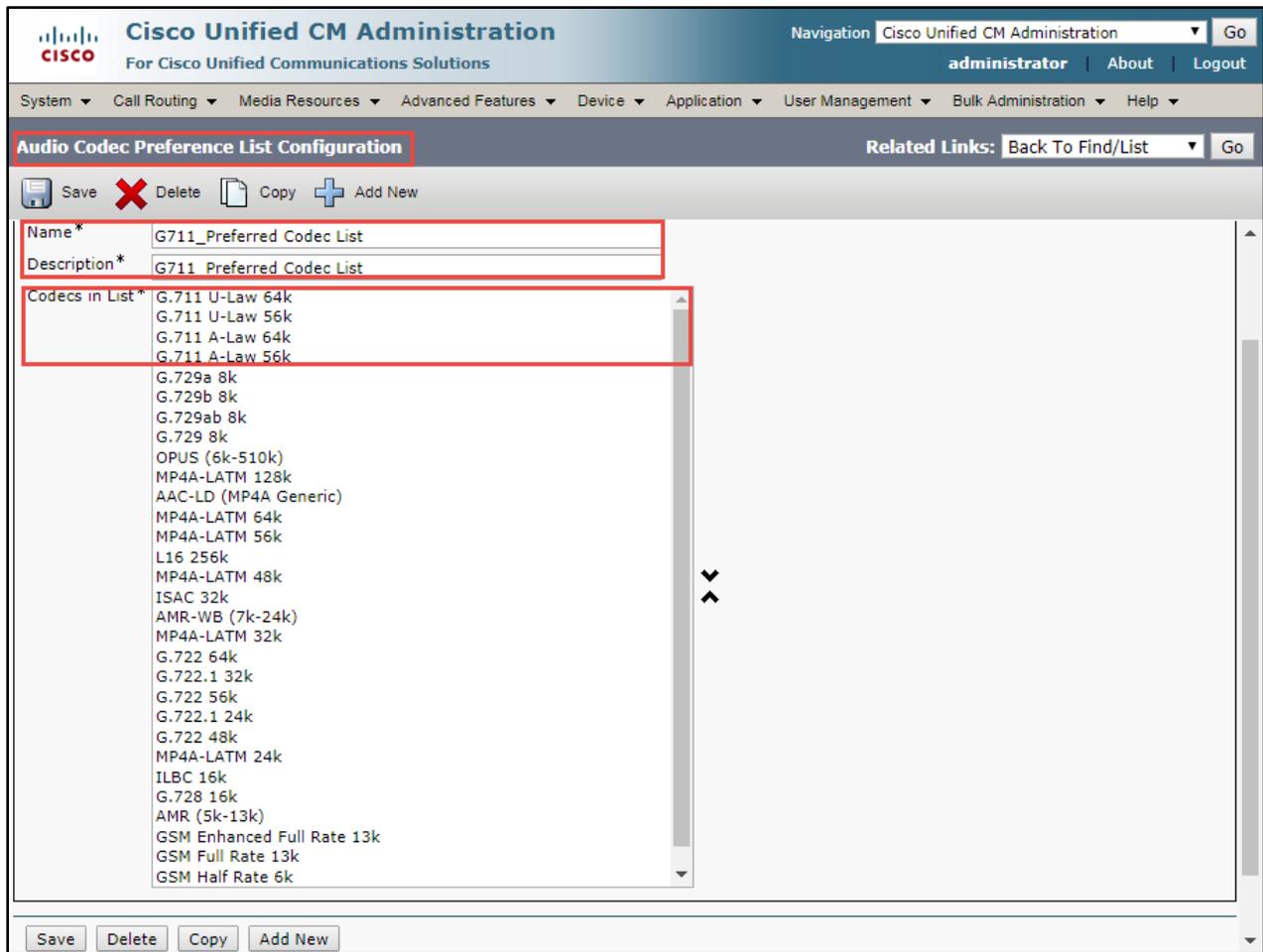


Figure 9 Cisco UCM Audio Codec Preference List

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

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Region Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Region Information

Name* G711_Region

Region Relationships

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**

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

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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 > View Details

Local Route Group Settings

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

Figure 11 Cisco UCM Device Pool

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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="< None >"/>
International Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="< None >"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="< None >"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="< None >"/>

Figure 12 Cisco UCM Device Pool Contd.,

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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

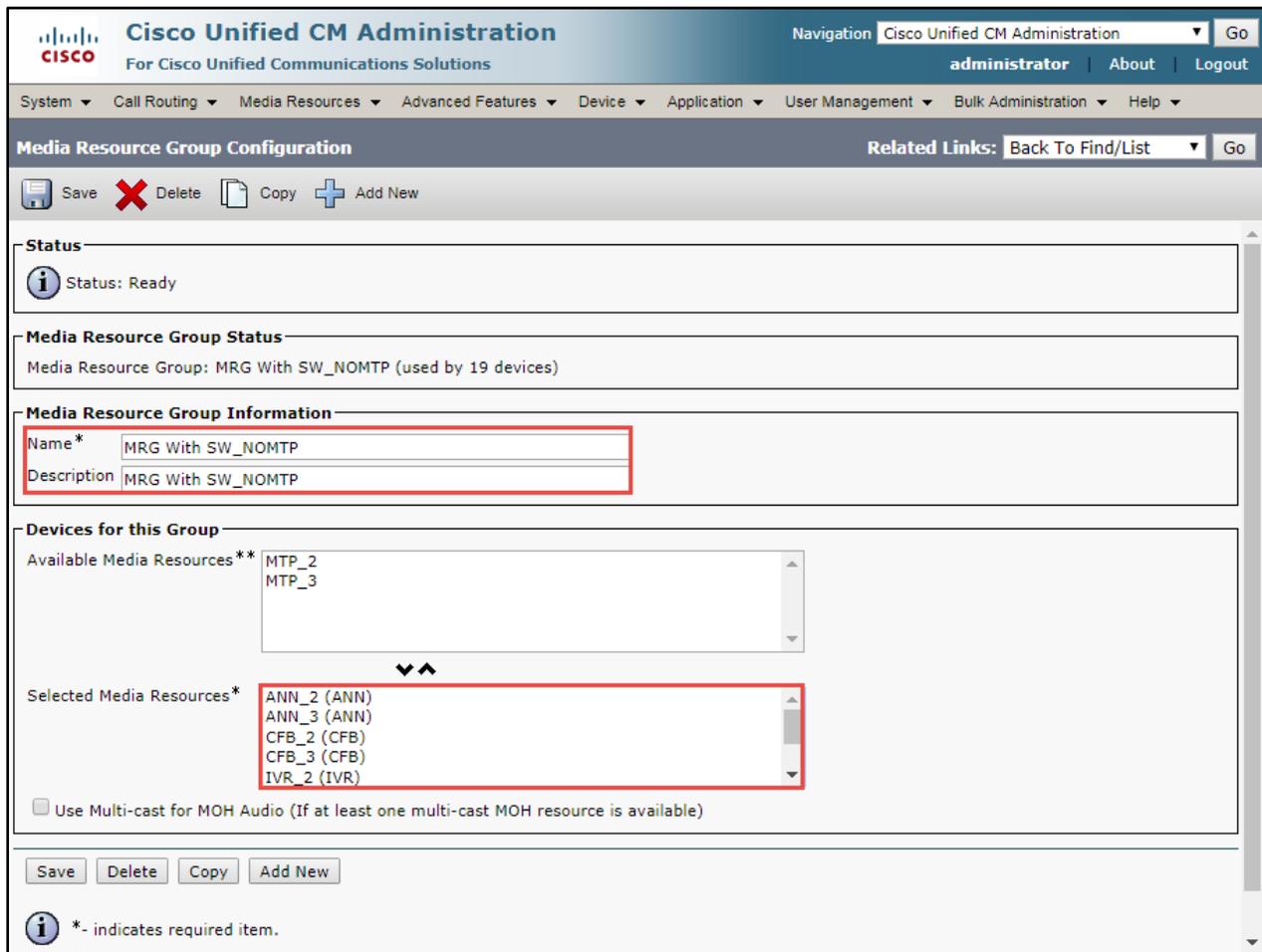


Figure 14 Cisco UCM Media Resources Group

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**

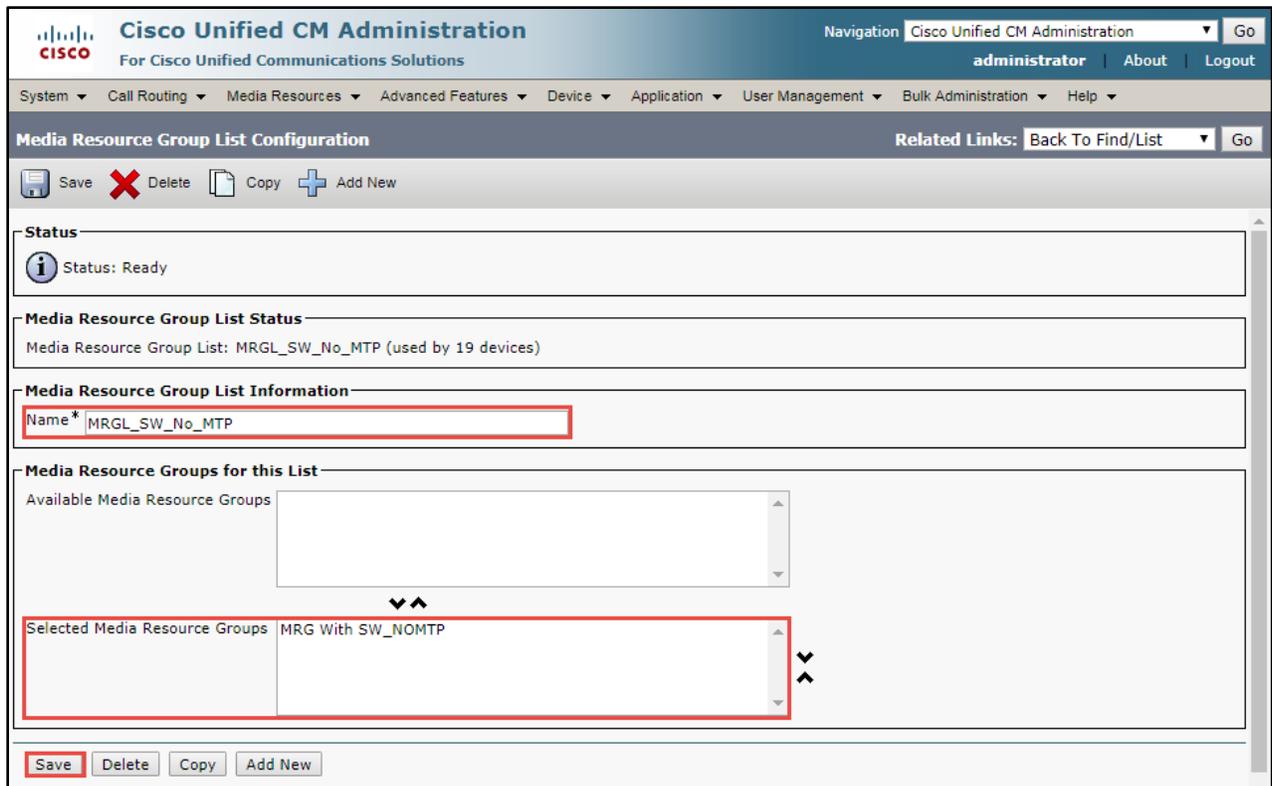


Figure 15 Cisco UCM Media Resources Group List

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**

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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
----------------	------

Enable Application level authorization
 Accept presence subscription
 Accept out-of-dialog refer**
 Accept unsolicited notification
 Accept replaces header

Figure 16 Cisco UCM SIP Trunk Security Profile

Accept replaces header
 Transmit security status
 Allow charging header
 SIP V.150 Outbound SDP Offer Filtering* Use Default Filter

Save Delete Copy Reset Apply Config Add New

Figure 17 Cisco UCM SIP Trunk Security Profile Contd.,

4.3.6 SIP Trunk to Cisco UBE

1. Navigate to **Device**→ **Trunk**
2. Provide a **Device Name**: AmazonSIPTrunkCUBE
3. Provide a **Description**: AmazonSIPTrunkCUBE
4. Set **Device Pool**: G711_pool
5. Set **Destination Address**: Set IP address of Cisco UBE
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 'Device Information' section is highlighted with a red box, showing the following configuration details:

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	AmazonSIPTrunkCUBE
Description	AmazonSIPTrunkCUBE
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

Additional configuration options include:

- 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 will...

Figure 18 Cisco UCM SIP Trunk Configuration

Cisco Unified CM Administration
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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* When using both sRTP and TLS ▾

Route Class Signaling Enabled* Default ▾

Use Trusted Relay Point* Default ▾

PSTN Access

Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile < None > ▾

MLPP and Confidential Access Level Information

MLPP Domain < None > ▾

Confidential Access Mode < None > ▾

Confidential Access Level < None > ▾

Call Routing Information

Remote-Party-Id

Asserted-Identity

Asserted-Type* Default ▾

SIP Privacy* Default ▾

Trust Received Identity* Trust All (Default) ▾

Figure 19 Cisco UCM SIP Trunk Configuration Contd.,

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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.,

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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.,

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Outbound Calls

Called Party Transformation CSS

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Calling and Connected Party Info Format*

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

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 *	10.80.11.18		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 "97.XXXXXXXXXX" was configured to enable outbound dialing from CUCM to PSTN using the access code as "97".
4. Set **Gateway/Route List:** AmazonSIPTrunkCUBE
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 Cisco UBE)

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 the user is logged in as "administrator". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is divided into sections: Status, Pattern Definition, and Call Classification. The Status section shows "Status: Ready". The Pattern Definition section contains the following fields:

Route Pattern*	97.XXXXXXXXXX
Route Partition	< None >
Description	AmazonRP
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	AmazonSIPTrunkCUBE
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error

Below the Pattern Definition section, there are additional fields for Call Classification, External Call Control Profile, and Authorization Level. The Call Classification is set to "OffNet", External Call Control Profile is "< None >", and Authorization Level is "0". There are also several checkboxes for "Allow Device Override", "Provide Outside Dial Tone", "Allow Overlap Sending", "Urgent Priority", "Require Forced Authorization Code", and "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 >	

Save Delete Copy Add New

Figure 27 Cisco UCM Route Pattern Configuration Contd.,

4.4 Cisco UBE Configuration

4.4.1 Global Cisco UBE settings

```
voice service voip
  no ip address trusted authenticate
  address-hiding
  mode border-element license capacity 100
  media disable-detailed-stats
  allow-connections sip to sip
  no supplementary-service sip handle-replaces
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback
  none
  sip
    session refresh
    asserted-id pai
    early-offer forced
    midcall-signaling passthru
    privacy-policy passthru
```

4.4.2 Codecs

```
voice class codec 1
  codec preference 1 g711u1aw
```

4.4.3 Sip-UA

```
sip-ua
  no remote-party-id
  sip-server dns:blg5XXXXX.g.voiceconnector.chime.aws:5060
```

4.4.4 Dial Peer

Inbound Dial Peer for Cisco UCM

```
dial-peer voice 100 voip
description *** Inbound Dial-Peer- from CUCM to CUBE ***
session protocol sipv2
session transport udp
incoming uri via CUCM
voice-class codec 1
dtmf-relay rtp-nte
no vad
```

Inbound Dial Peer for Amazon Chime Voice Connector

```
dial-peer voice 200 voip
description *** Inbound Dial-Peer- from Amazon to CUBE ***
translation-profile incoming Amazon-In
session protocol sipv2
session transport udp
incoming called e164-pattern-map 890
voice-class codec 1
dtmf-relay rtp-nte
no vad
```

Outbound Dial Peer to Amazon Chime Voice Connector

```
dial-peer voice 310 voip
description *** Outbound Dial-Peer to Amazon****
translation-profile outgoing Amazon-Out
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
```

```
voice-class sip localhost dns:blg5XXXXX.g.voiceconnector.chime.aws
voice-class sip options-keepalive
dtmf-relay rtp-nte
no vad
```

Outbound Dial Peer to Cisco UCM

```
dial-peer voice 311 voip
description *** Outbound Dial-Peer to CUCM****
destination-pattern 919.....
session protocol sipv2
session target ipv4:10.80.12.2:5060
session transport udp
voice-class codec 1
dtmf-relay rtp-nte
no vad
```

4.4.5 Cisco UBE Running Configuration

```
Current configuration : 10141 bytes
!
! Last configuration change at 17:32:20 UTC Tue Jul 30 2019 by cisco
!
version 16.9
service timestamps debug datetime msec
service timestamps log datetime msec
service call-home
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname AmazonCVC

boot-start-marker
boot system flash isr4400-universalk9.16.09.03.SPA.bin
```

```
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
call-home
! If contact email address in call-home is configured as sch-smart-
licensing@cisco.com
! the email address configured in Cisco Smart License Portal will be
used as contact email address to send SCH notifications.
contact-email-addr sch-smart-licensing@cisco.com
profile "CiscoTAC-1"
active
destination transport-method http
no destination transport-method email
!
ip name-server 8.8.8.8
!
login on-success log
!
subscriber templating
!
multilink bundle-name authenticated
!
password encryption aes
!
```

```

!
voice service voip
  no ip address trusted authenticate
  address-hiding
  mode border-element license capacity 100
  media disable-detailed-stats
  allow-connections sip to sip
  no supplementary-service sip handle-replaces
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback
  none
  sip
    session refresh
    asserted-id pai
    early-offer forced
    midcall-signaling passthru
    privacy-policy passthru
    g729 annexb-all
!
!
voice class uri CUCM sip
  host 10.80.12.2
voice class codec 1
  codec preference 1 g711ulaw
!

voice class e164-pattern-map 890
  e164 +191.....$
!
!
voice translation-rule 10
  rule 1 /\(^.....$\)/ /+1\1/
!
voice translation-rule 11

```

```
rule 1 /\(^.....$\)/ /+1\1/
!
voice translation-rule 12
rule 1 /\(^.....$\)/ /+1\1/
!
voice translation-rule 20
rule 1 /\^+1\(.*\)/ /\1/
!
!
voice translation-profile Amazon-INTL-Out
translate calling 11
translate called 12
!
voice translation-profile Amazon-In
translate called 20
!
voice translation-profile Amazon-Out
translate calling 11
translate called 10
!
voice-card 0/1
no watchdog
!
no license feature hseck9
license udi pid ISR4431/XXXXXXX
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level uck9
license boot level securityk9
no license smart enable
diagnostic bootup level minimal
!
```

```
spanning-tree extend system-id
memory free low-watermark processor 75392
!
!
!
redundancy
mode none
!

interface GigabitEthernet0/0/0
description WAN Interface to Amazon Voice Connector
ip address 192.65.X.X 255.255.X.X
negotiation auto
!
interface GigabitEthernet0/0/1
no ip address
shutdown
negotiation auto
!
interface GigabitEthernet0/0/2
description LAN side connected to MS3-1/0/39
ip address 10.80.11.18 255.255.255.0
negotiation auto
!
interface GigabitEthernet0/0/3
no ip address
shutdown
negotiation auto
!
interface Service-Engine0/1/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
```

```
no ip address
negotiation auto
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.x.x
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 10.80.12.0 255.255.255.0 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
!
ip ssh server algorithm encryption aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes192-ctr aes256-ctr
!
!
!
!
!
control-plane
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
```

```
!  
!  
dial-peer voice 100 voip  
  description *** Inbound Dial-Peer- from CUCM to CUBE ***  
  session protocol sipv2  
  session transport udp  
  incoming uri via CUCM  
  voice-class codec 1  
  dtmf-relay rtp-nte  
  no vad  
!  
dial-peer voice 311 voip  
  description *** Outbound Dial-Peer to CUCM****  
  destination-pattern 919.....  
  session protocol sipv2  
  session target ipv4:10.80.12.2:5060  
  session transport udp  
  voice-class codec 1  
  dtmf-relay rtp-nte  
  no vad  
!  
dial-peer voice 312 voip  
  description *** Outbound Dial-Peer to Amazon****  
  translation-profile outgoing Amazon-INTL-Out  
  destination-pattern 91978.....  
  session protocol sipv2  
  session target sip-server  
  session transport udp  
  voice-class codec 1  
  voice-class sip localhost dns:blg5XXXXX.g.voiceconnector.chime.aws  
  voice-class sip options-keepalive  
  dtmf-relay rtp-nte  
  no vad
```

```
!  
dial-peer voice 200 voip  
  description *** Inbound Dial-Peer- from Amazon to CUBE ***  
  translation-profile incoming Amazon-In  
  session protocol sipv2  
  session transport udp  
  incoming called e164-pattern-map 890  
  voice-class codec 1  
  dtmf-relay rtp-nte  
  no vad  
!  
dial-peer voice 310 voip  
  description *** Outbound Dial-Peer to Amazon****  
  translation-profile outgoing Amazon-Out  
  destination-pattern .T  
  session protocol sipv2  
  session target sip-server  
  voice-class codec 1  
  voice-class sip localhost dns:blg5XXXXX.g.voiceconnector.chime.aws  
  voice-class sip options-keepalive  
  dtmf-relay rtp-nte  
  no vad  
!  
!  
sip-ua  
  no remote-party-id  
  sip-server dns:blg5XXXXX.g.voiceconnector.chime.aws:5060  
!  
line con 0  
  transport input none  
  stopbits 1  
line aux 0  
  stopbits 1
```

```
line vty 0 4
  exec-timeout 180 0
  login local
  transport input telnet
!
End
```