



# **Amazon Chime Voice Connector**

## **SIP Trunking Configuration Guide:**

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

**August 2019**

## Document History

Rev. No.	Date	Description
1.0	Aug-19-2019	Draft SIP Trunk Configuration Guide
1.7	Aug-22-2019	Minor Edits based on feedback
1.8	Aug-27-2019	Updated the Summary of Test Results
1.9	Sep-5-2019	Updated the Summary of Test Results
1.10	Sep-11-2019	Minor Edits based on feedback
1.11	Oct-11-2019	Minor Edits based on feedback
1.12	Feb-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 Cisco UBE .....	24
4.3.7	Route Pattern.....	28
4.4	Cisco UBE Configuration.....	31
4.4.1	Global Cisco UBE settings .....	31
4.4.2	Codecs.....	31
4.4.3	Sip-UA .....	31
4.4.4	Dial Peer .....	32
4.4.5	Cisco UBE Running Configuration.....	33

## 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 **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

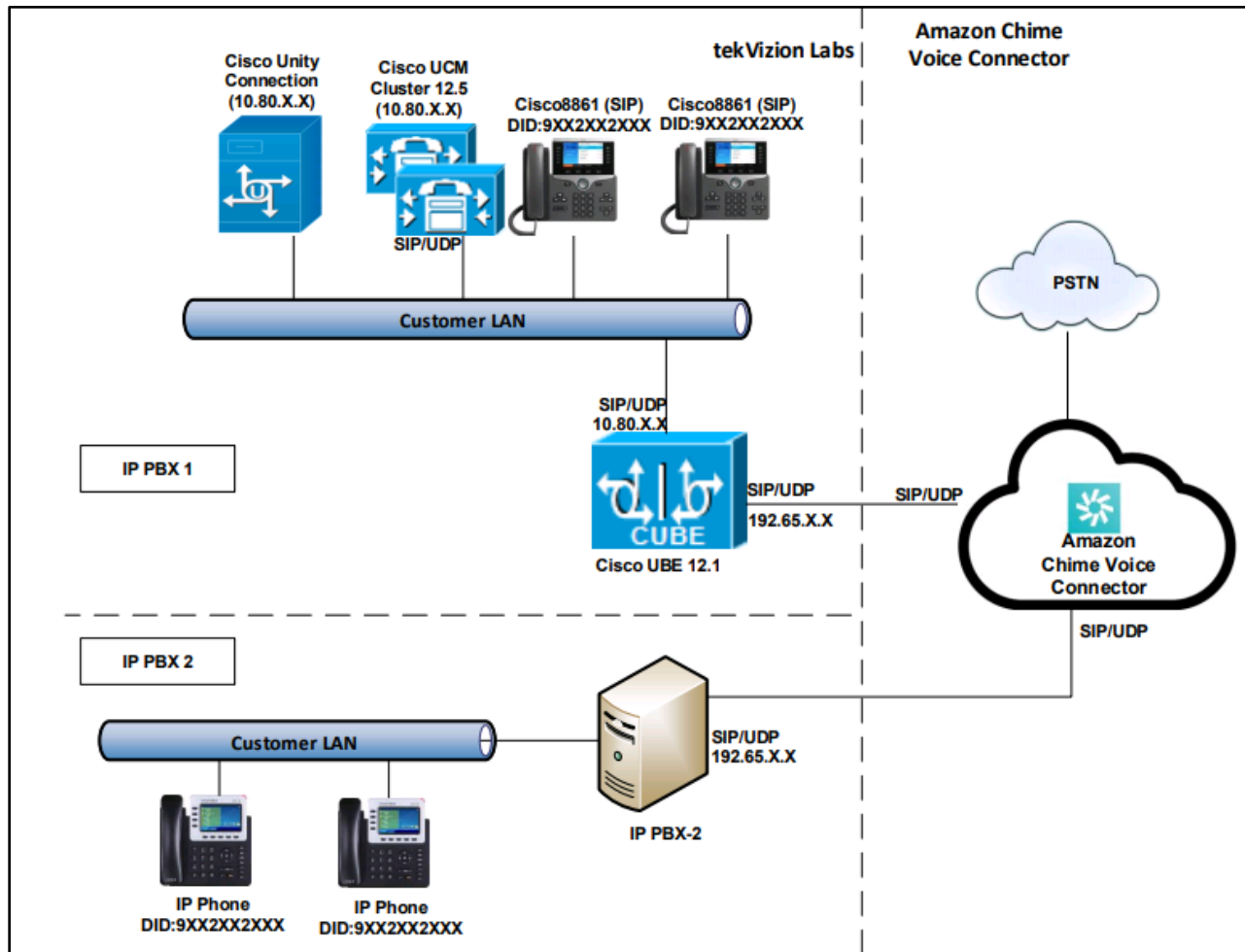


Figure 1 Network Topology

## 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	<a href="#">Section 4.3</a>
Step 2	Cisco UBE 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>Cisco UBE</b>	
LAN IP Address	10.80.11.18
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 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  
**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)*	<input type="text" value="180"/>
Timer Register Delta (seconds)*	<input type="text" value="5"/>
Timer Register Expires (seconds)*	<input type="text" value="3600"/>
Timer T1 (msec)*	<input type="text" value="500"/>
Timer T2 (msec)*	<input type="text" value="4000"/>
Retry INVITE*	<input type="text" value="6"/>
Retry Non-INVITE*	<input type="text" value="10"/>
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	<input type="text" value="16384"/>
Stop Media Port*	<input type="text" value="32766"/>
DSCP for Audio Calls	<span>Use System Default</span> ▾
DSCP for Video Calls	<span>Use System Default</span> ▾

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	<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=" &lt; None &gt;"/>
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

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		

**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\*: 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)\*
60

Ping Interval for Out-of-service Trunks (seconds)\*
120

Ping Retry Timer (milliseconds)\*
500

Ping Retry Count\*
6

**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

Save
Delete
Copy
Reset
Apply Config
Add New

*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

**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

**Audio Codec Preference List Configuration** Related Links: Back To Find/List Go

Save Delete Copy Add New

Name\* G711\_Prefered Codec List

Description\* G711 Preferred Codec List

Codecs in List\*

- G.711 U-Law 64k
- G.711 U-Law 56k
- G.711 A-Law 64k
- G.711 A-Law 56k
- G.729a 8k
- G.729b 8k
- G.729ab 8k
- G.729 8k
- OPUS (6k-510k)
- MP4A-LATM 128k
- AAC-LD (MP4A Generic)
- MP4A-LATM 64k
- MP4A-LATM 56k
- L16 256k
- MP4A-LATM 48k
- ISAC 32k
- AMR-WB (7k-24k)
- MP4A-LATM 32k
- G.722 64k
- G.722.1 32k
- G.722 56k
- G.722.1 24k
- G.722 48k
- MP4A-LATM 24k
- ILBC 16k
- G.728 16k
- AMR (5k-13k)
- GSM Enhanced Full Rate 13k
- GSM Full Rate 13k
- GSM Half Rate 6k

Save Delete Copy Add New

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\_Prefered Codec List** to this Region



**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 ▾

**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_Prefered 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 >

[View Details](#)

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

Device Mobility Calling Search Space < None >  
 AAR Calling Search Space < None >  
 AAR Group < None >  
 Calling Party Transformation CSS < None >  
 Called Party Transformation CSS < None >

**Geolocation Configuration**

Geolocation < None >  
 Geolocation Filter < None >

**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.

Clear Prefix Settings Default Prefix Settings

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

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

\*- 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

**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

**Media Resource Group Configuration** Related Links: Back To Find/List Go

Save Delete Copy Add New

**Status**  
Status: Ready

**Media Resource Group Status**  
Media Resource Group: MRG With SW\_NOMTP (used by 19 devices)

**Media Resource Group Information**

Name\* MRG With SW\_NOMTP  
Description MRG With SW\_NOMTP

**Devices for this Group**

Available Media Resources\*\* MTP\_2  
MTP\_3

Selected Media Resources\* ANN\_2 (ANN)  
ANN\_3 (ANN)  
CFB\_2 (CFB)  
CFB\_3 (CFB)  
IVR\_2 (IVR)

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Save Delete Copy Add New

\*- indicates required item.

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**

**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

**Media Resource Group List Configuration** Related Links: Back To Find/List Go

Save Delete Copy Add New

**Status**  
Status: Ready

**Media Resource Group List Status**  
Media Resource Group List: MRGL\_SW\_No\_MTP (used by 19 devices)

**Media Resource Group List Information**  
Name\* MRGL\_SW\_No\_MTP

**Media Resource Groups for this List**  
Available Media Resource Groups  
Selected Media Resource Groups MRG With SW\_NOMTP

Save Delete Copy Add New

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 ▾

**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

**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


**Device Information**

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

☐ 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 cause call quality and other information

Figure 18 Cisco UCM SIP Trunk 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 ▾

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

Save
 Delete
 Reset
 Add New

☐ **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.,

**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 *	10.80.11.18		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 web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The user is logged in as "administrator". The main menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The current page is "Route Pattern Configuration", with a "Related Links" section containing "Back To Find/List" and "Go".

The "Status" section shows "Status: Ready". The "Pattern Definition" section contains the following fields and options:

- Route Pattern\***: 97.XXXXXXXXXX
- Route Partition**: < None >
- Description**: AmazonRP
- 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\***: AmazonSIPTrunkCUBE (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 >	

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 g711ulaw
```

### 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
```