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
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1.11	Oct-11-2019	Minor Edits based on feedback
1.12	Feb-3-2020	Minor Edits based on feedback

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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

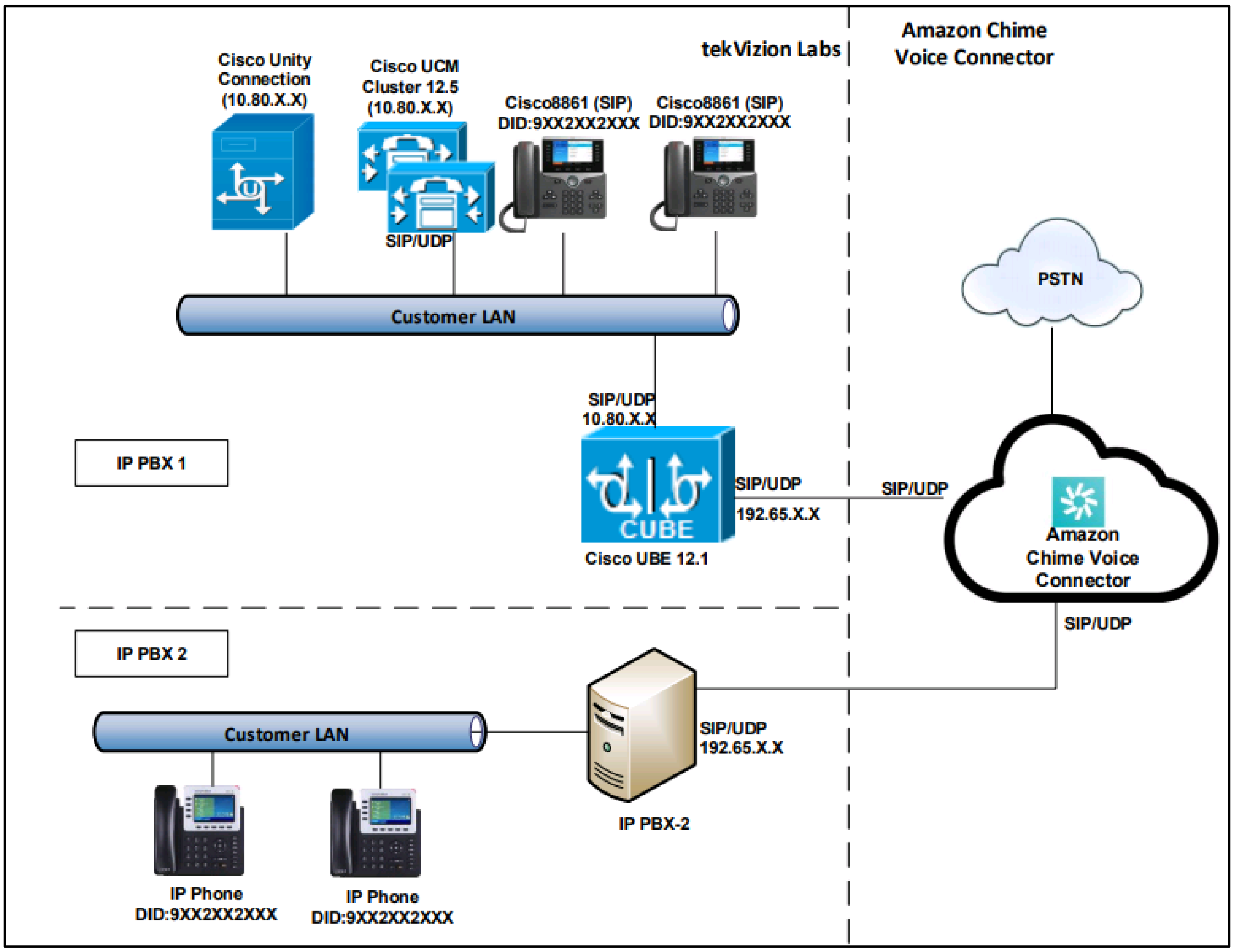

Figure 1 Network Topology

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.

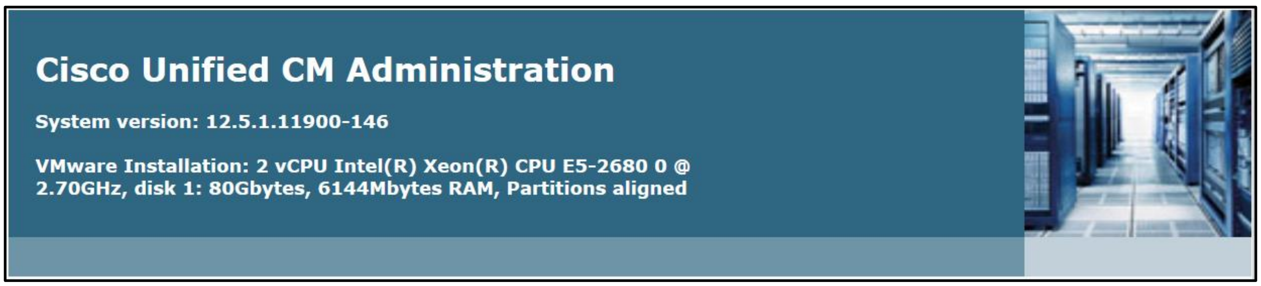

Figure 2: Cisco UCM software version

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**

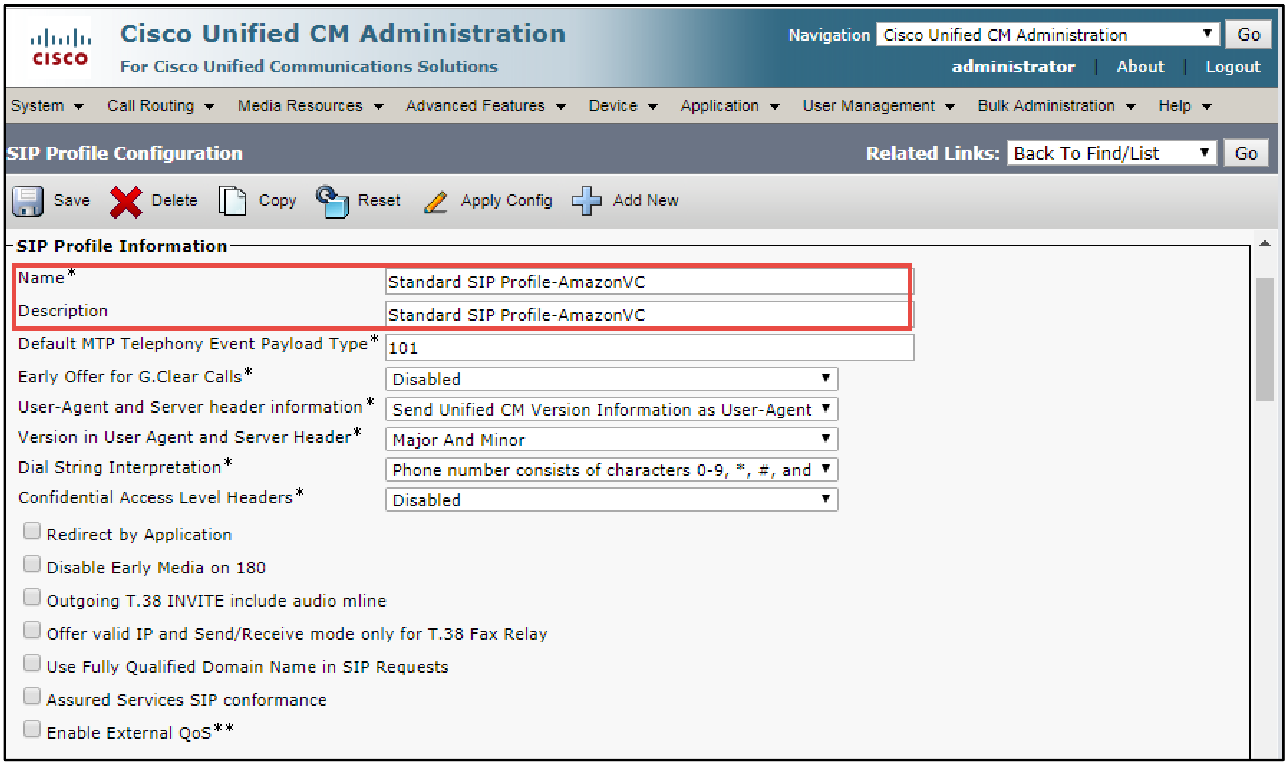

Figure 3 Cisco UCM SIP Profile

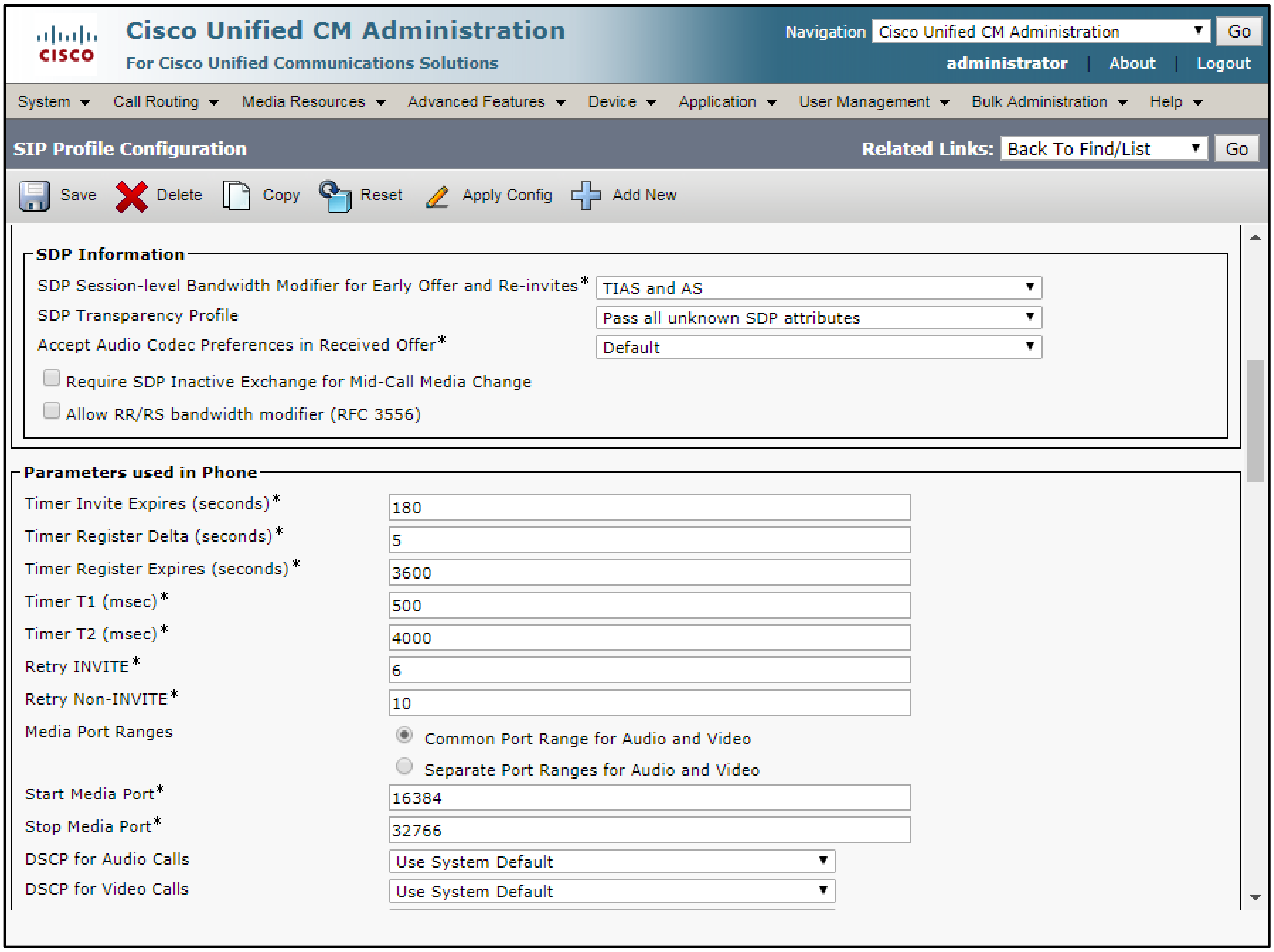

Figure 4 Cisco UCM SIP Profile Contd.,

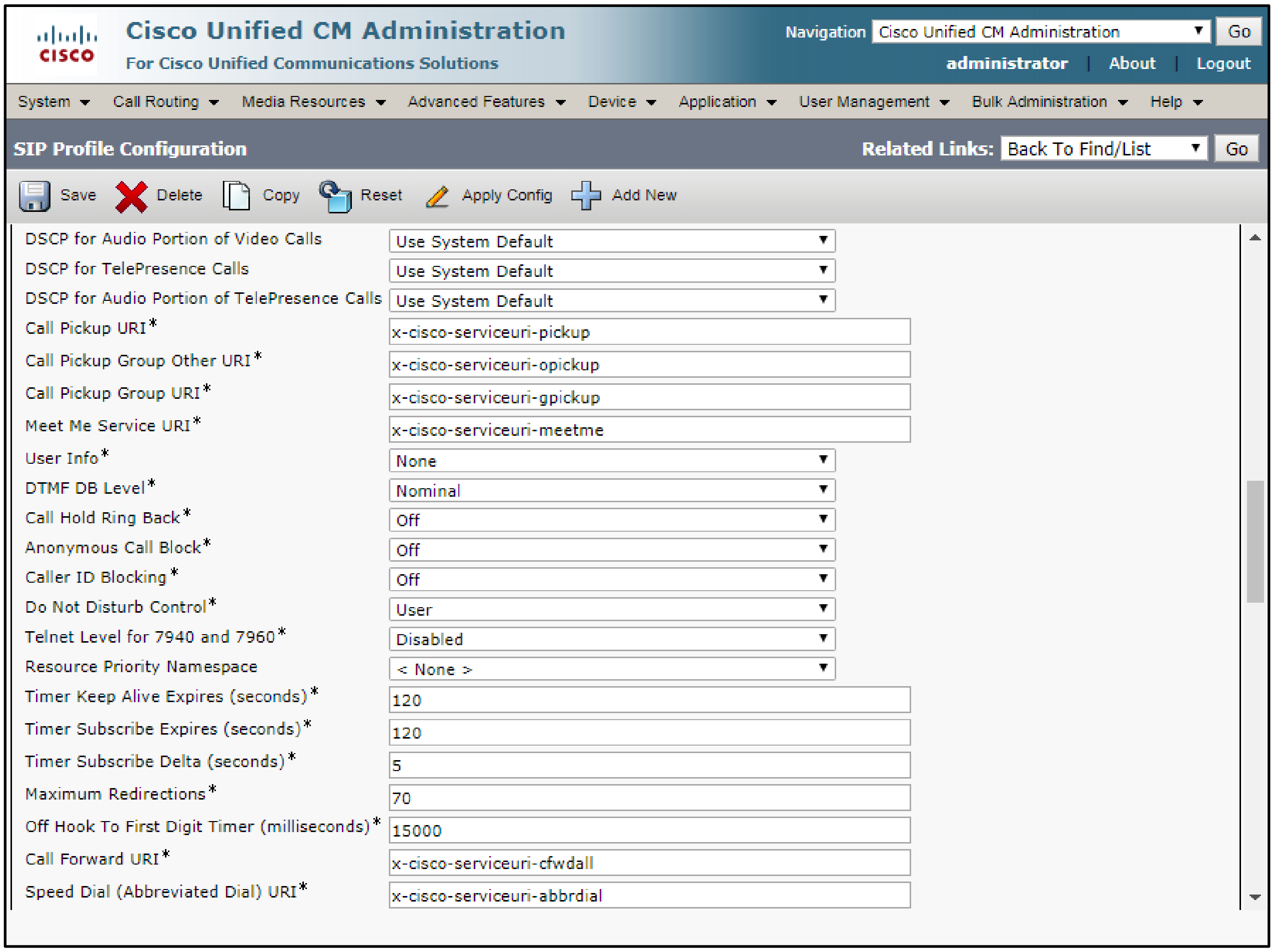

Figure 5 Cisco UCM SIP Profile Contd.,

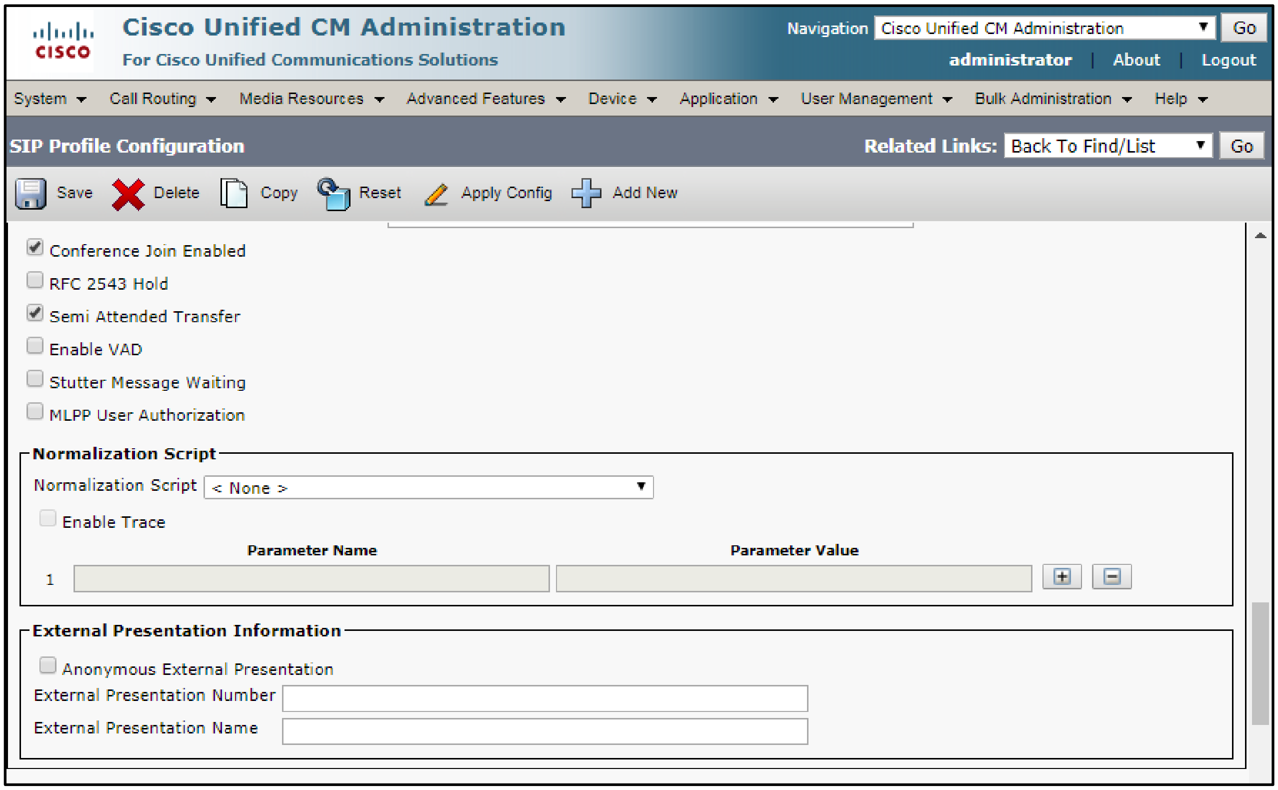

Figure 6 Cisco UCM SIP Profile Contd.,

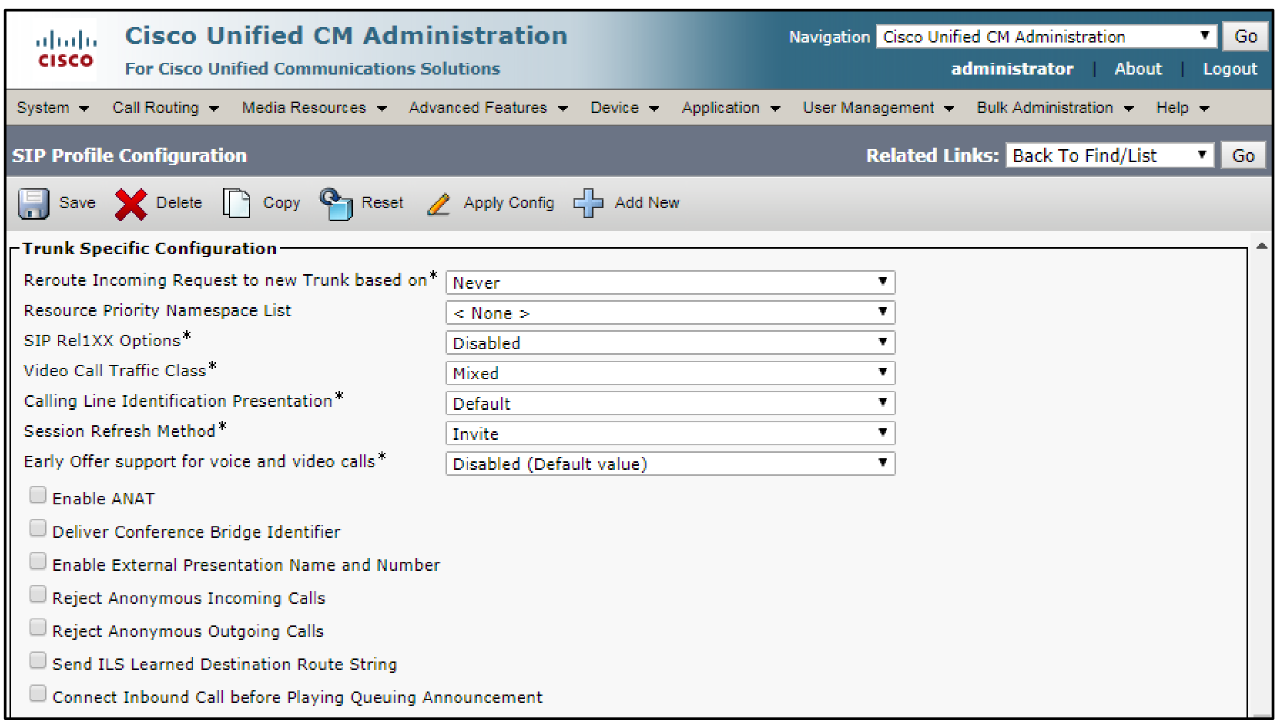

Figure 7 Cisco UCM SIP Profile Contd.,

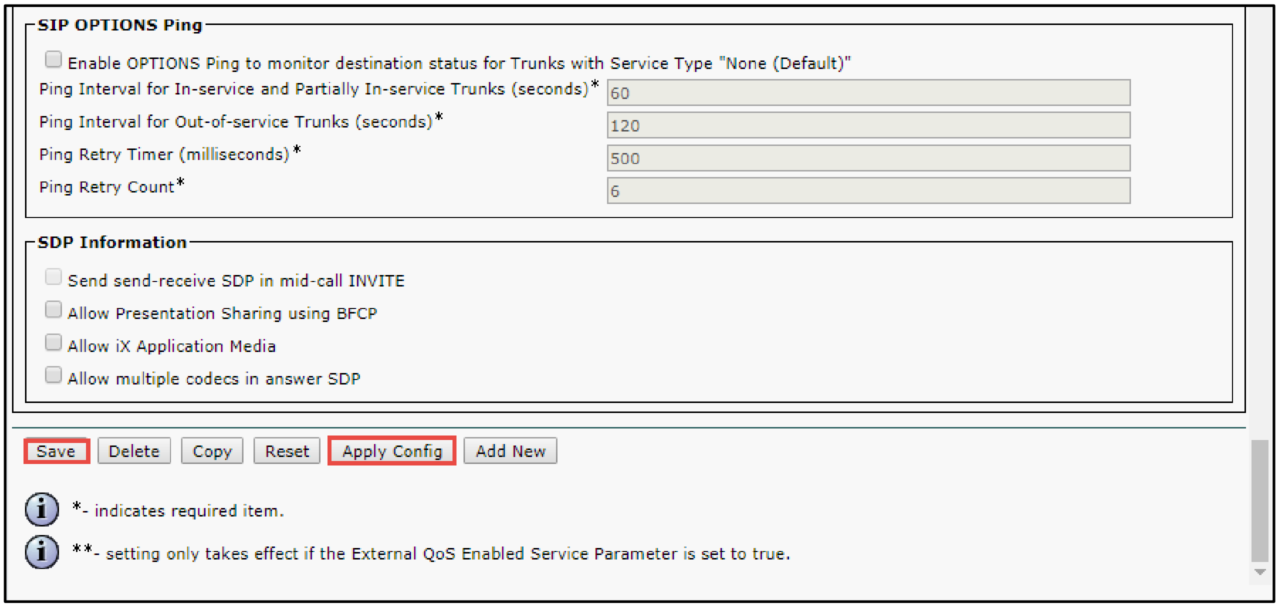

Figure 8 Cisco UCM SIP Profile Contd.,

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

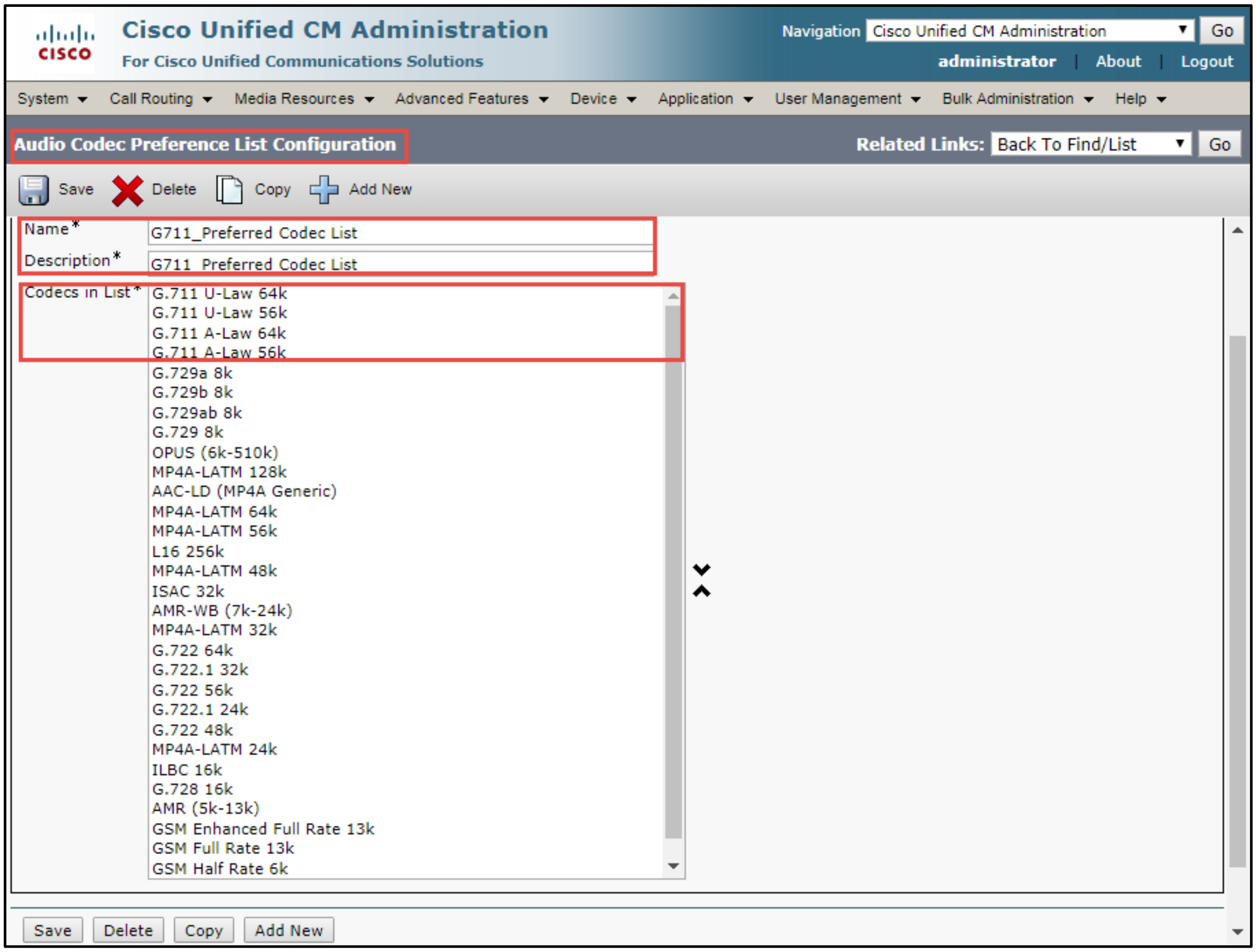

Figure 9 Cisco UCM Audio Codec Preference List

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

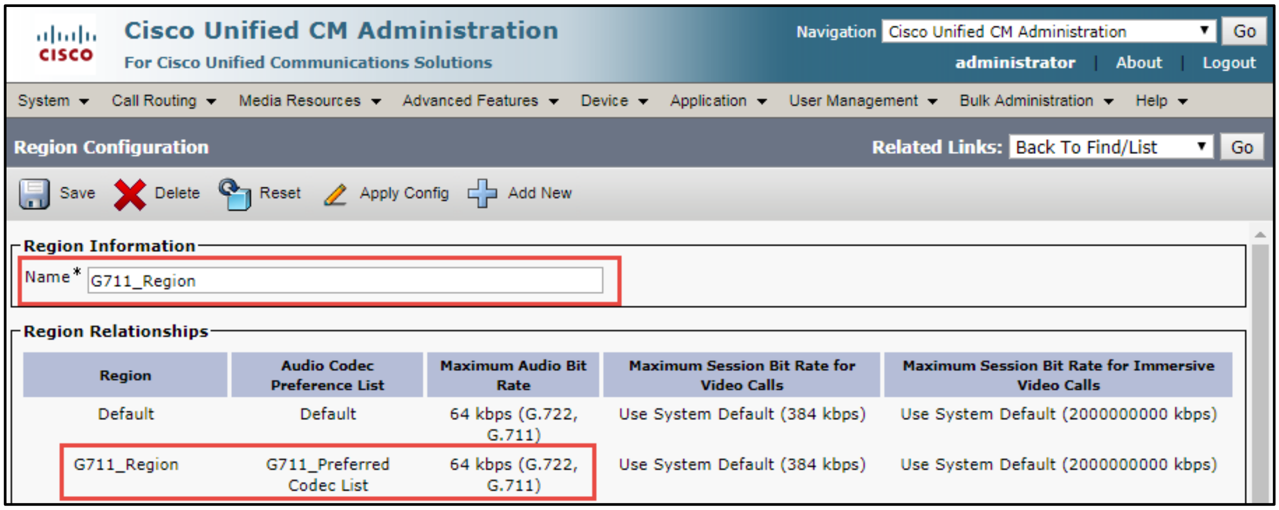

Figure 10 Cisco UCM Region

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**

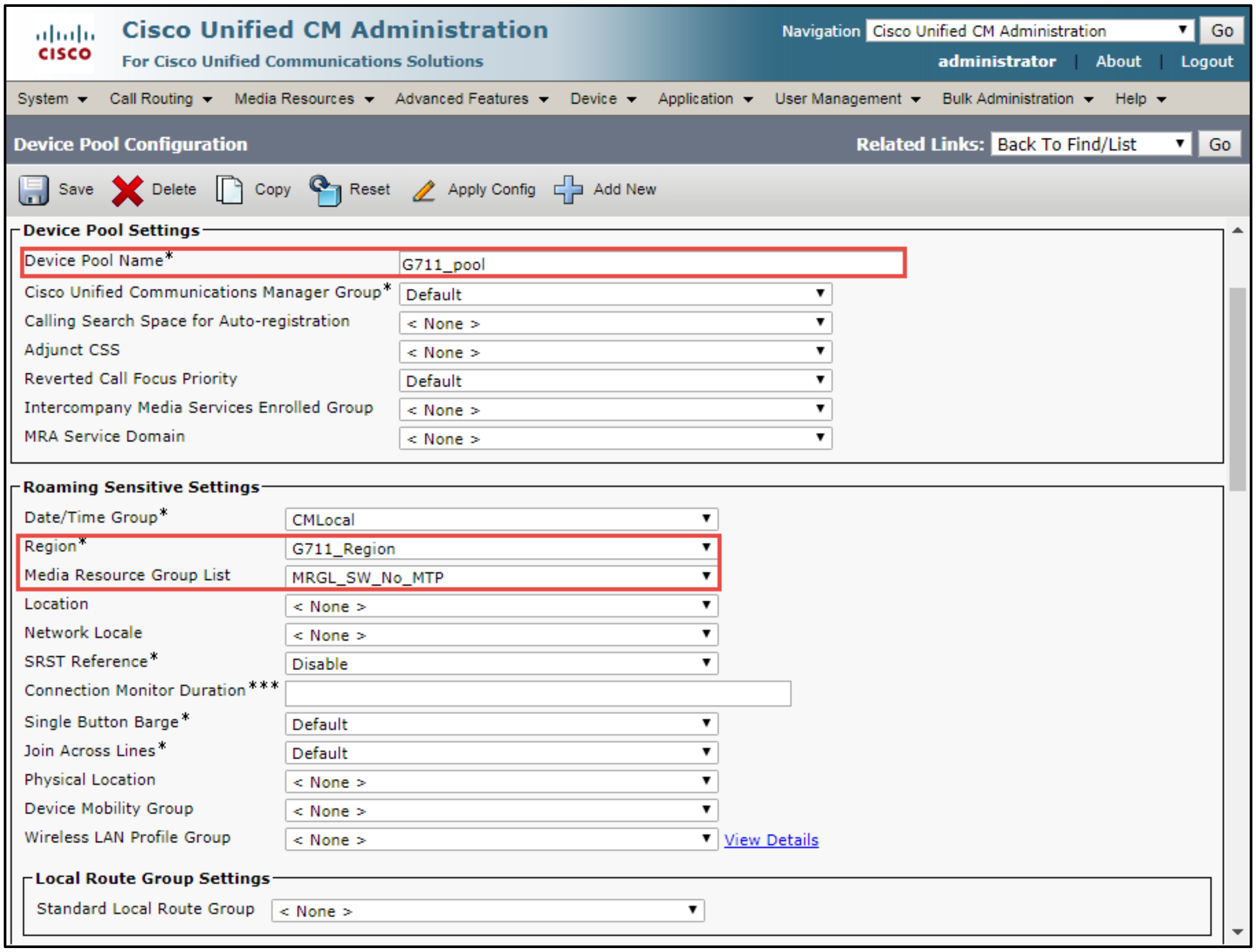

Figure 11 Cisco UCM Device Pool

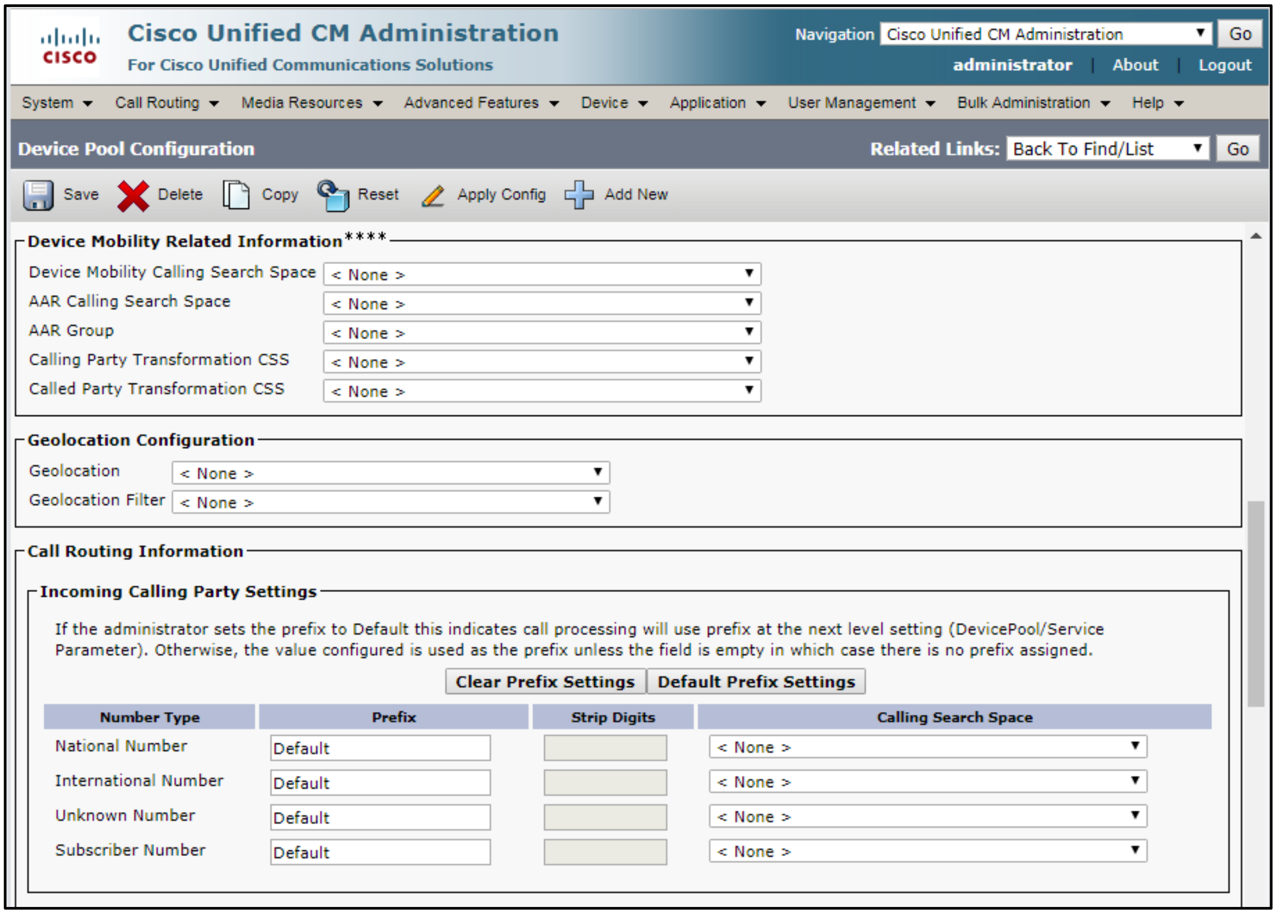

Figure 12 Cisco UCM Device Pool Contd.,

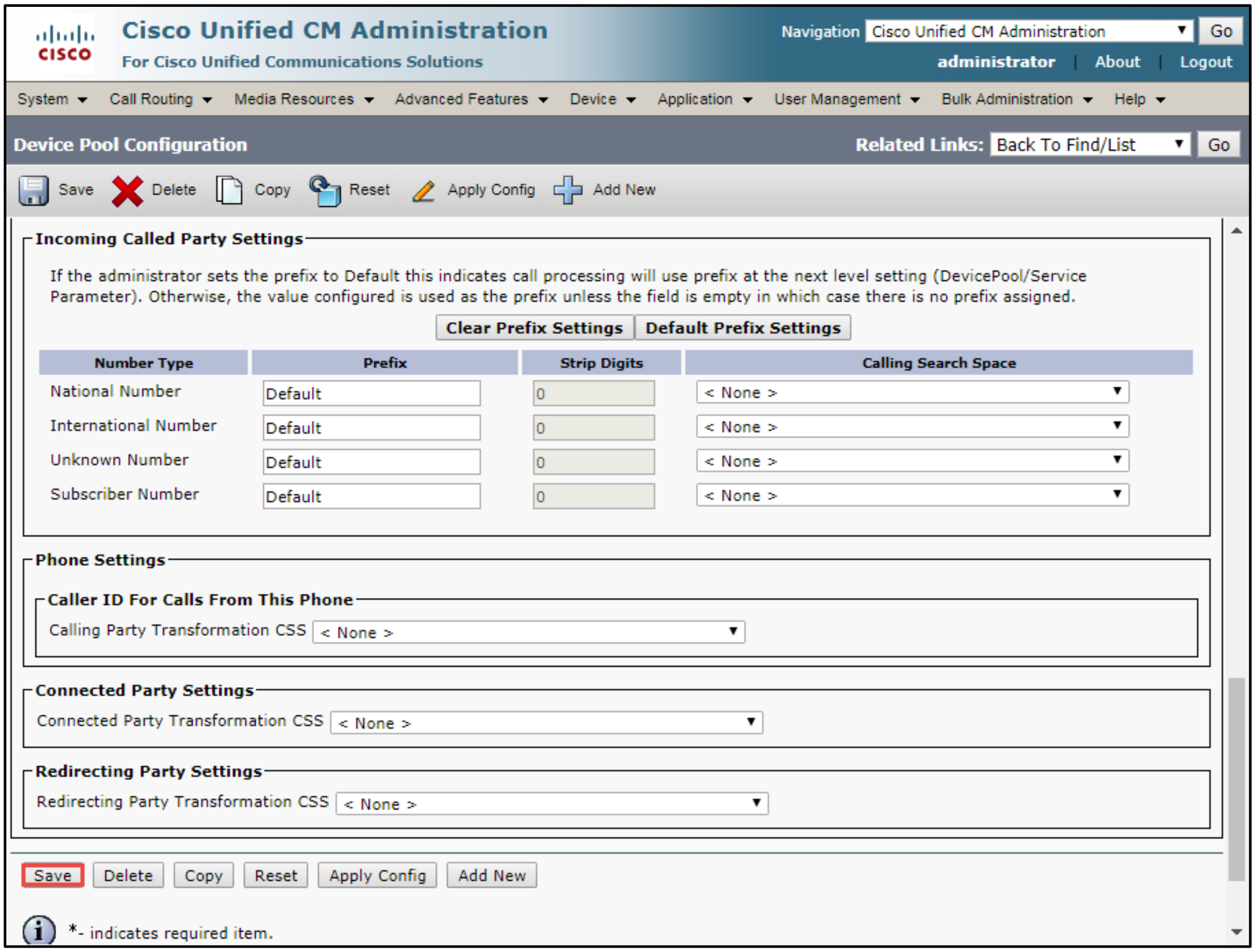

Figure 13 Cisco UCM Device Pool Contd.,

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

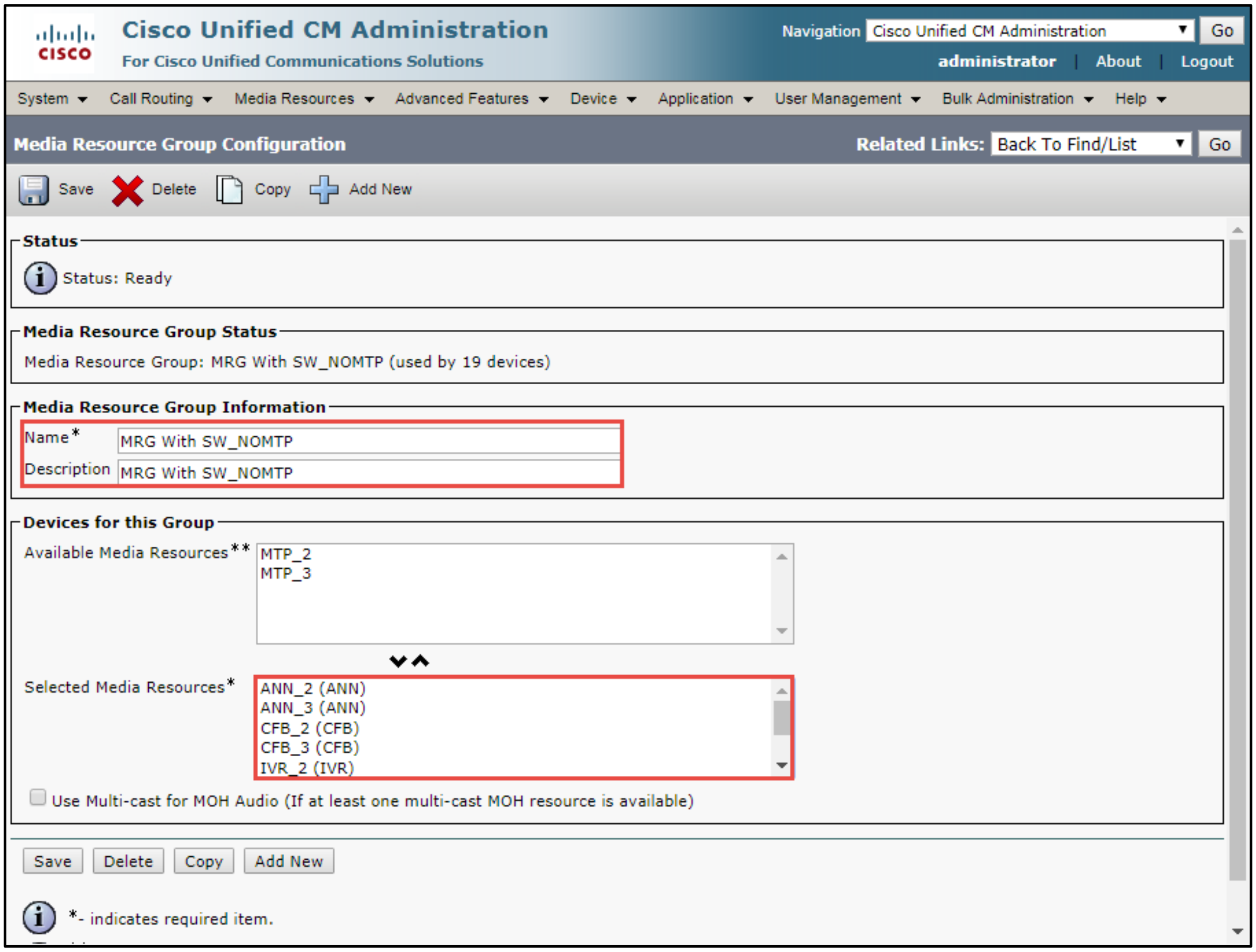

Figure 14 Cisco UCM Media Resources Group

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**

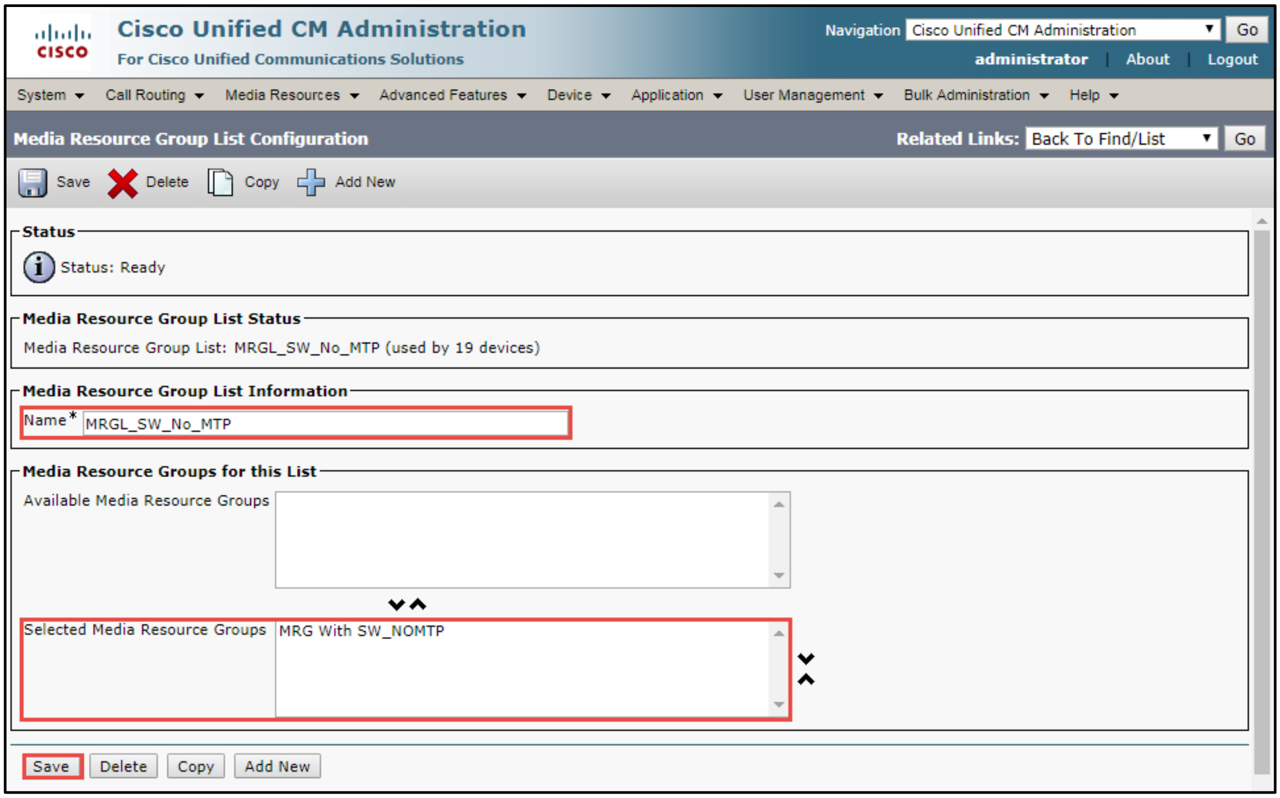

Figure 15 Cisco UCM Media Resources Group List

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**

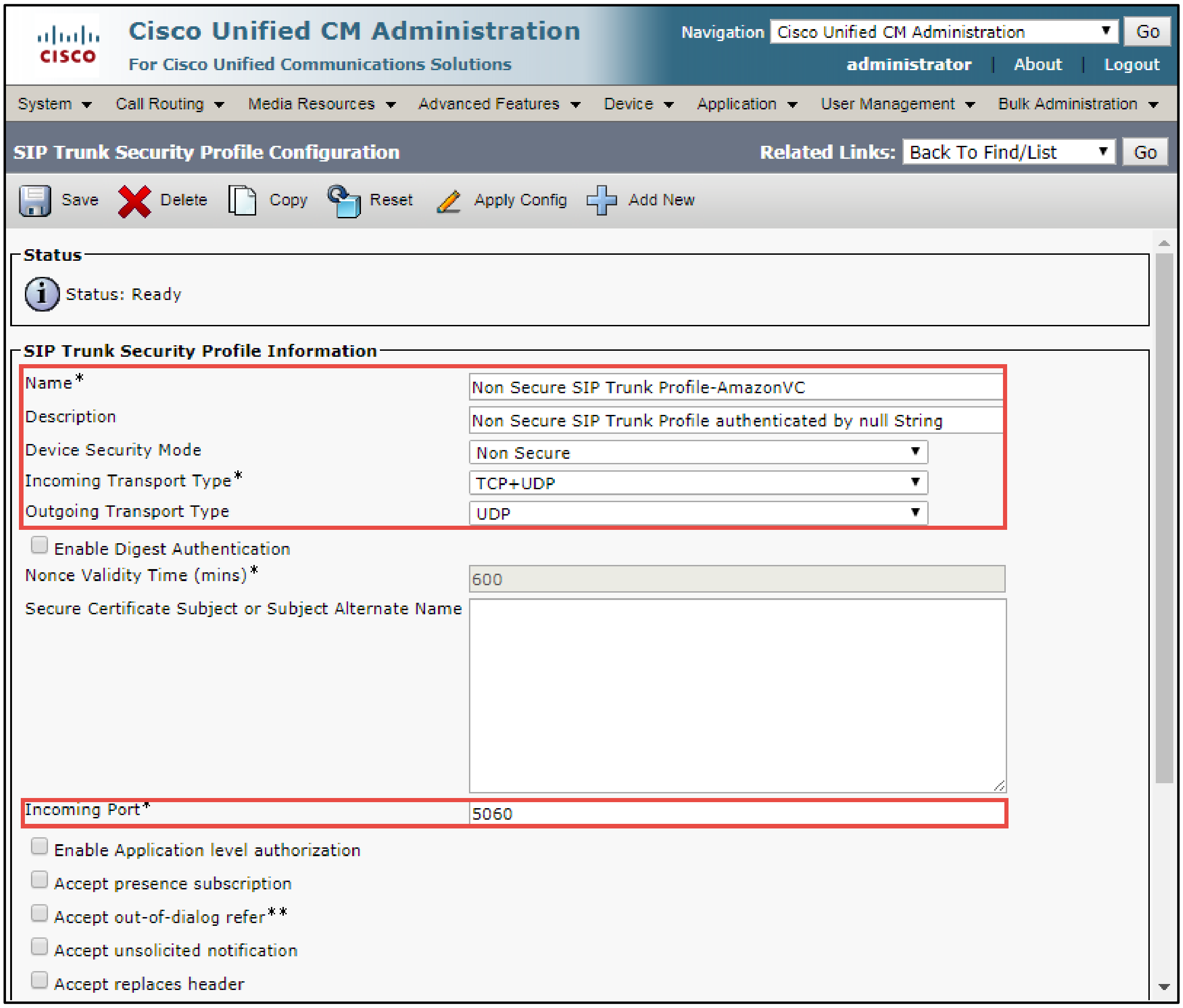

Figure 16 Cisco UCM SIP Trunk Security Profile

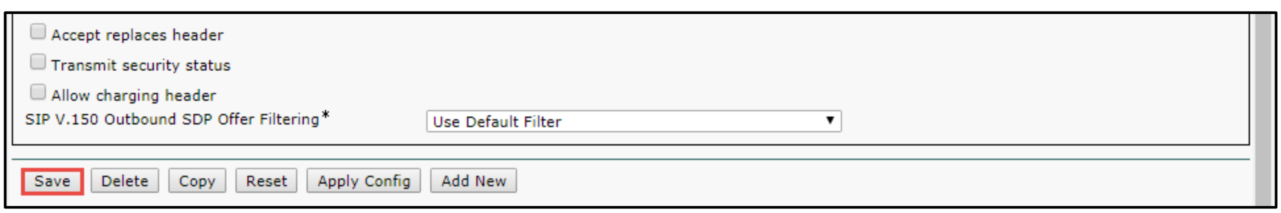

Figure 17 Cisco UCM SIP Trunk Security Profile Contd.,

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

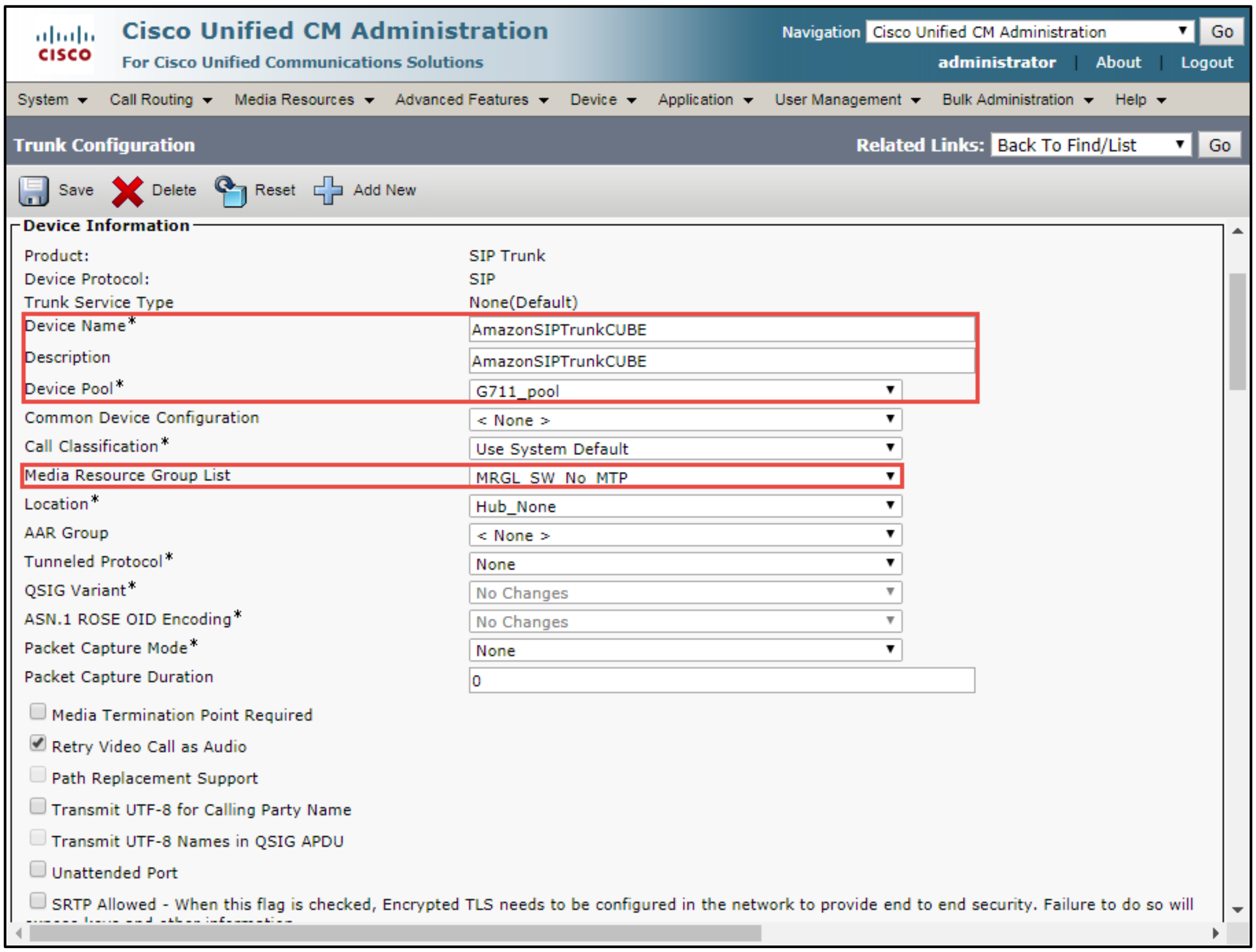

Figure 18 Cisco UCM SIP Trunk Configuration

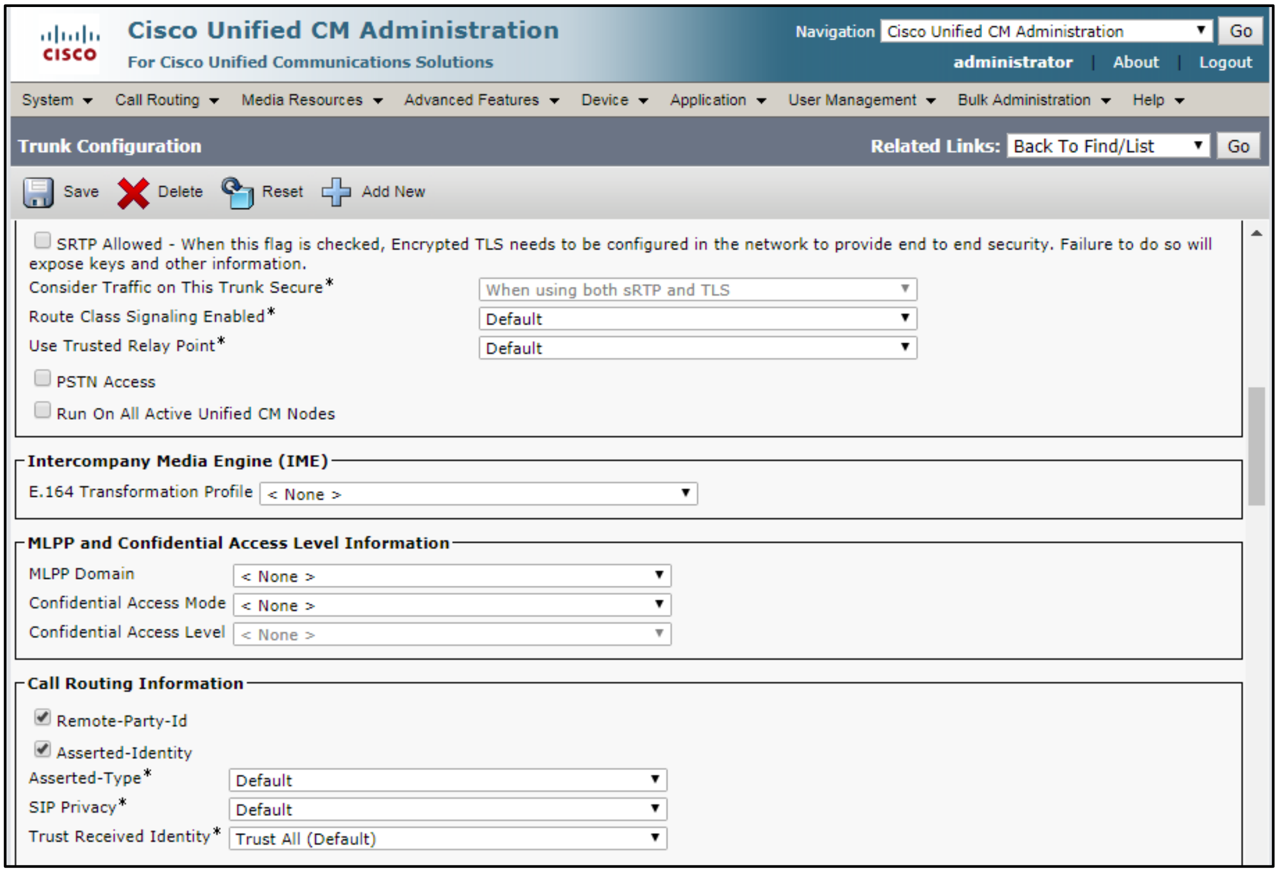

Figure 19 Cisco UCM SIP Trunk Configuration Contd.,

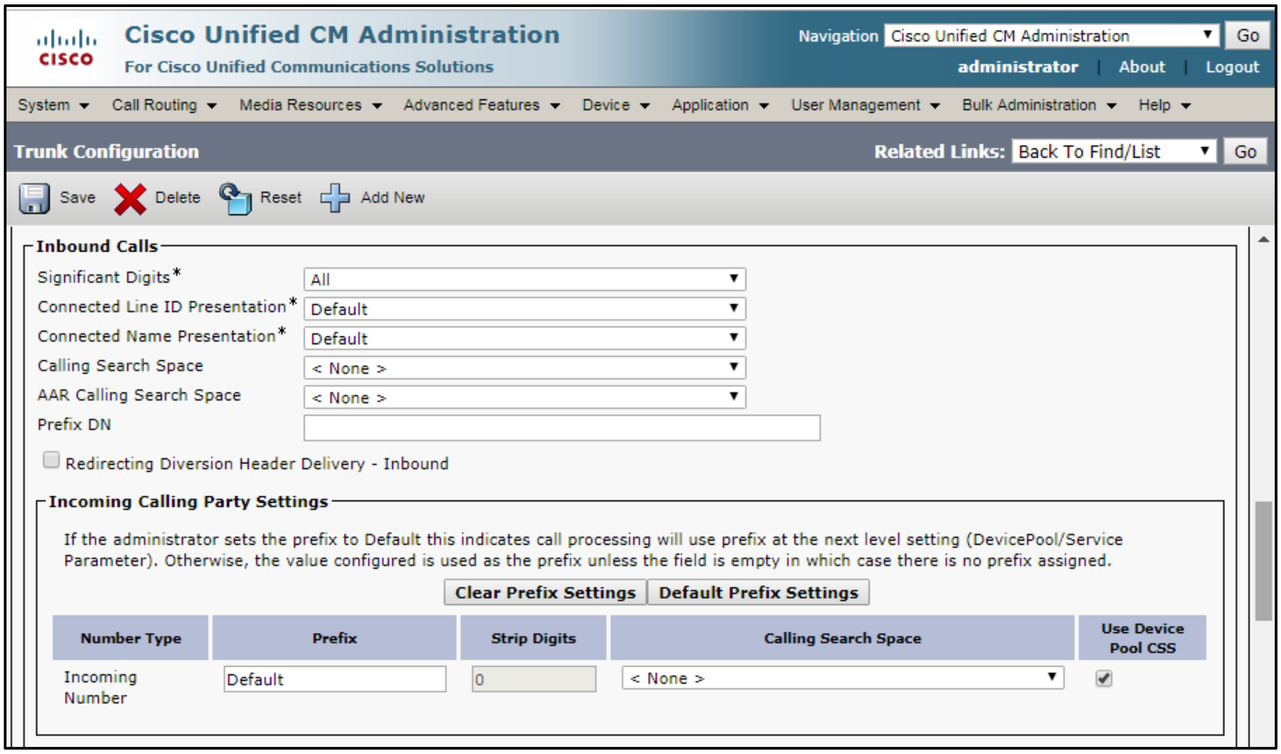

Figure 20 Cisco UCM SIP Trunk Configuration Contd.,

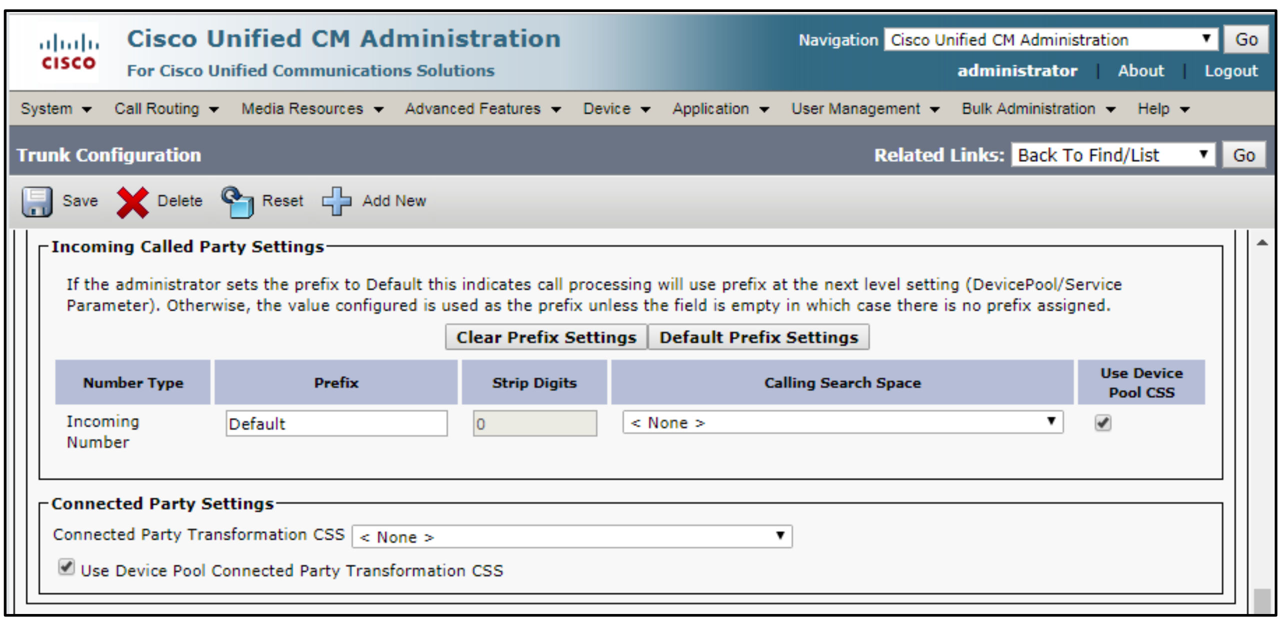

Figure 21 Cisco UCM SIP Trunk Configuration Contd.,

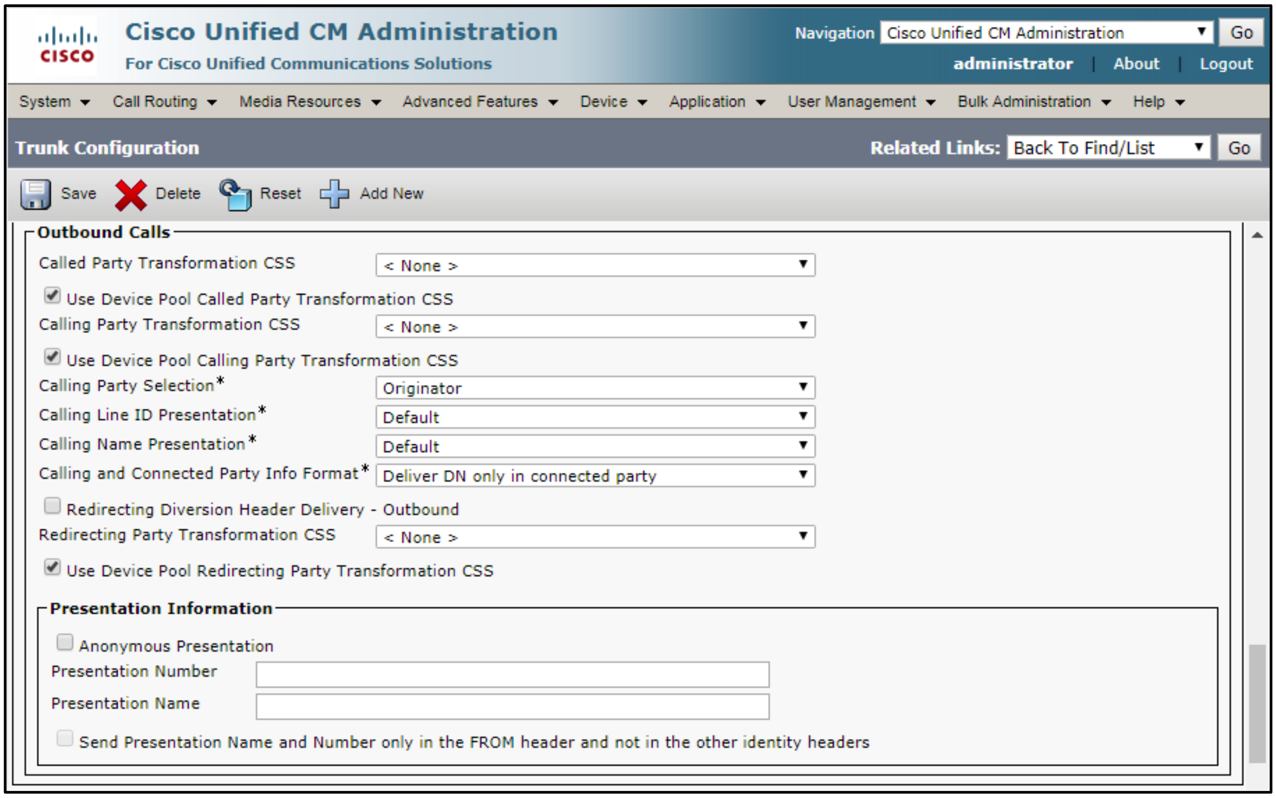

Figure 22 Cisco UCM SIP Trunk Configuration Contd.,

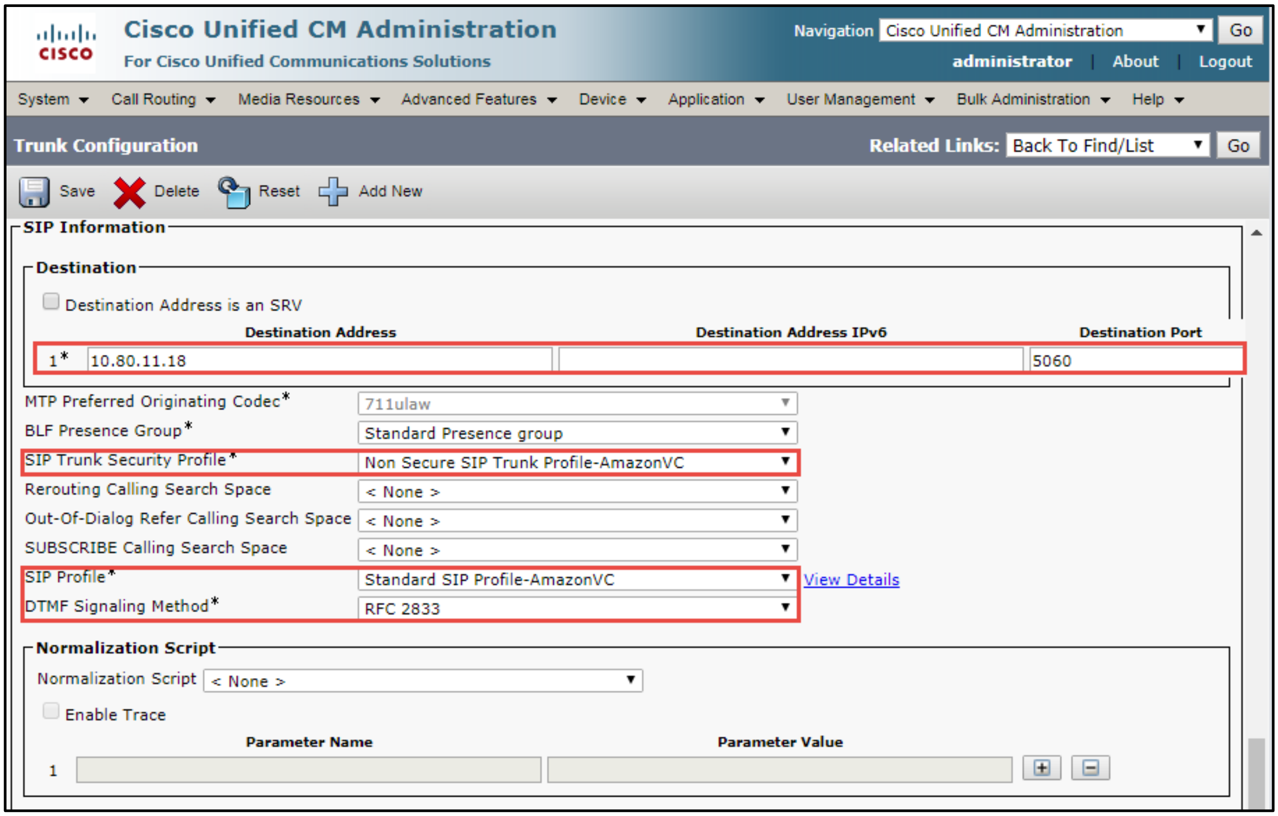

Figure 23 Cisco UCM SIP Trunk Configuration Contd.,

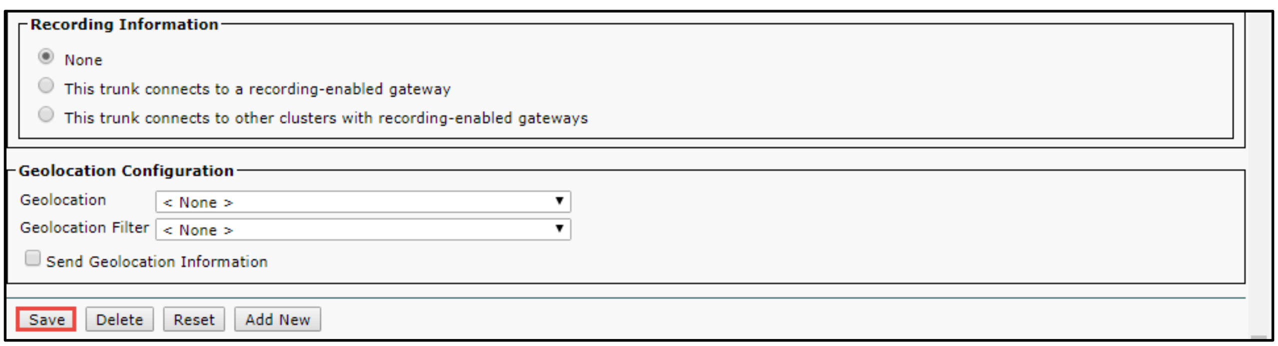

Figure 24 Cisco UCM SIP Trunk Configuration Contd.,

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**

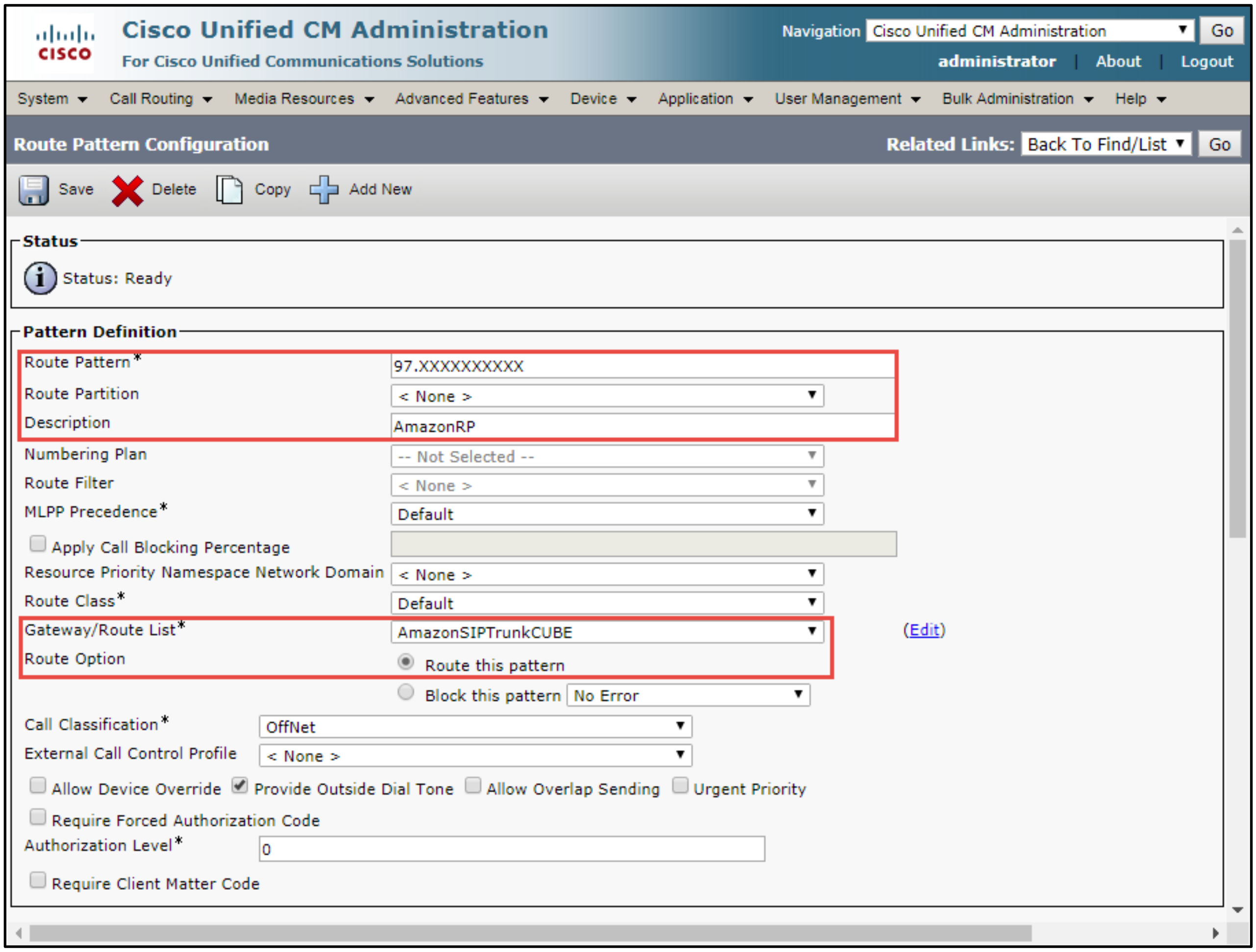

Figure 25 Cisco UCM Route Pattern Configuration

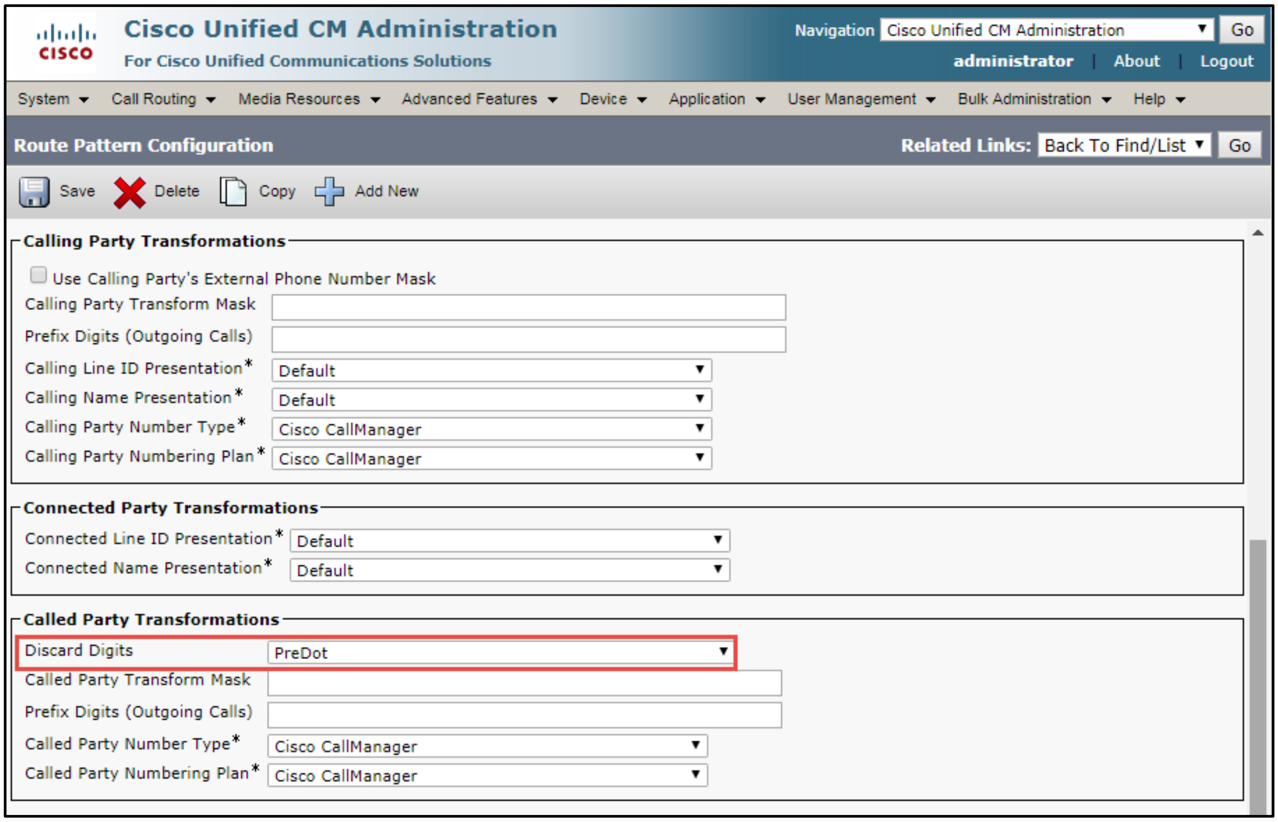

Figure 26 Cisco UCM Route Pattern Configuration Contd.,

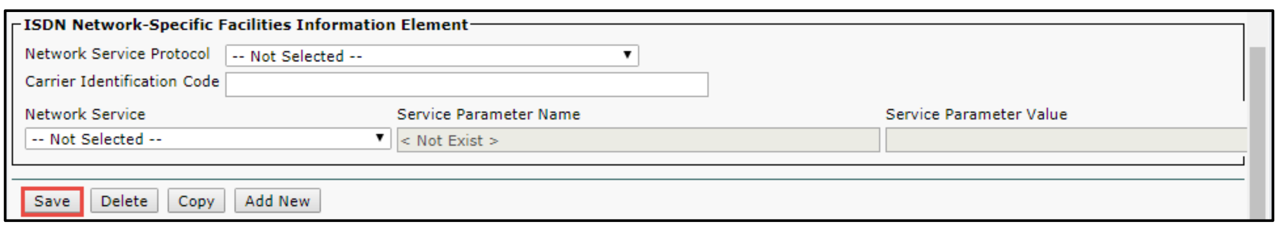

Figure 27 Cisco UCM Route Pattern Configuration Contd.,

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/
!
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
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 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
```