Application Note

Connecting Cisco Unified Communication Manager [v12.0.1] to Cox Business SIP Trunks via Cisco Unified Border Element v12.0 [IOS-XE 16.06.01]

January 24, 2018

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Introduction

Service Providers today, such as Cox, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and centralized IP to TDM POP gateways to provide on-net and off-net services.

Cox is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager (Cisco UCM) and Cox network, Cisco Unified Border Element (Cisco UBE) ISR 4331/K9 running IOS 16.6.1 can be used. The Cisco Unified Border Element 16.6.1 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.0.1 connected to Cox network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco Unified Communications Manager. Only configuration settings specifically required for Cox interoperability are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure Cisco UCM 12.0.1, and Cisco UBE on ISR 4331/K9 [IOS – 16.6.1] for connectivity to Cox SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (Cox) via Cisco Border UBE v12.0 [IOS-XE] 16.6.1.
- Testing was performed in accordance to Cox generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and High Availability.
- The Cisco UBE configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Cox SIP network and Cisco Unified Communications. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying Cisco UCM to interoperate to Cox SIP Trunking network.

Network Topology

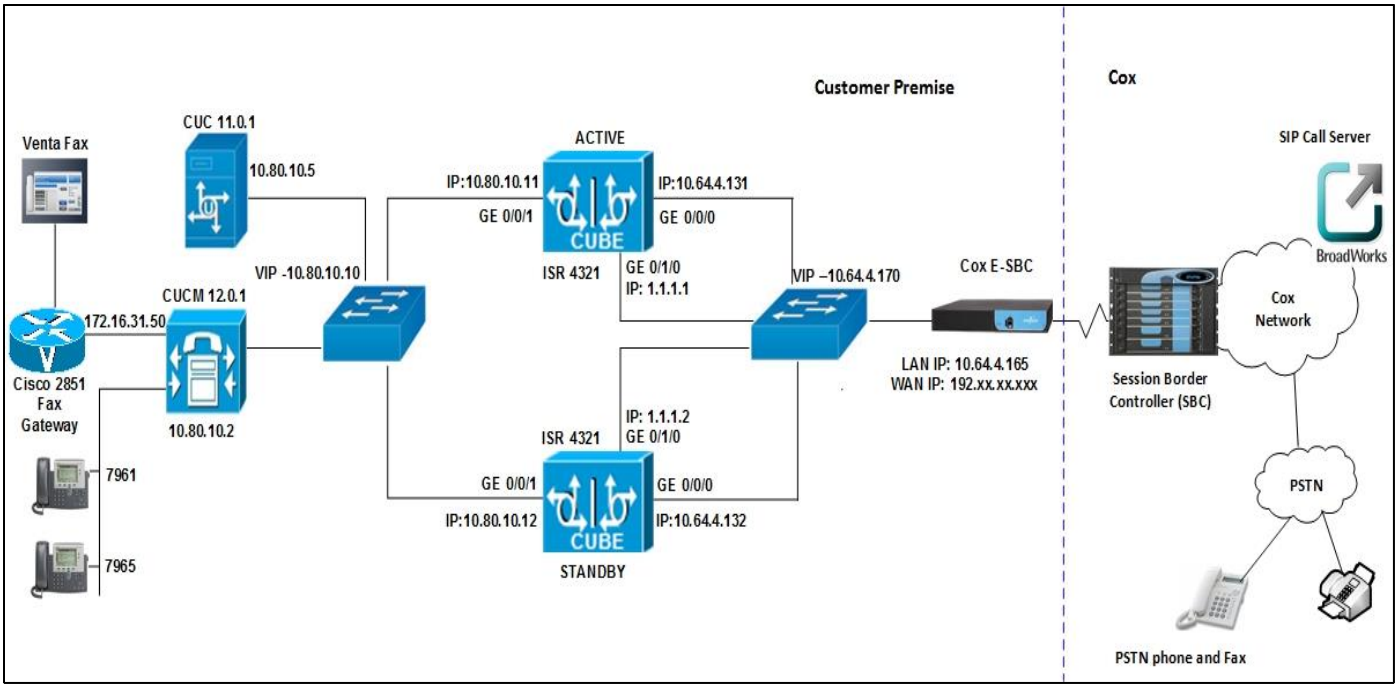


Figure 1 Network Topology

- The network topology includes the Cisco UCM Cluster, Unity Voicemail system, Cisco Fax gateway and 2 Cisco Endpoints. Cisco UCM has a trunk configured to Cisco UBE's Virtual IP Address. Cox was used as the service provider with SIP trunk to the Cisco UBE using the WAN Virtual IP Address.
- 2 Cisco Unified Border Elements are used here for High Availability.
- SIP Trunk transport type used between Cisco Unified Border Element and Cisco UCM is UDP and to Cox is UDP.

Cisco UCM and Cisco UBE Settings:

Setting	Value
Transport from Cisco UBE to Cisco UCM	UDP with RTP
Transport from Cisco UBE to Cox	UDP with RTP
Voice Mail Support	YES
Session Refresh	YES
Early Media support with PRACK	YES

System Components

Hardware Requirements

- Cisco UBE on Cisco ISR 4331 router
- CUCM cluster on UCS, 1 Publisher node and 2 Subscriber nodes
- Cisco 2851 with FXS ports and Analog Fax machine
- Generic Cisco IP-Phones

Software Requirements

- CUBE-Version: 12.0 running IOS-XE 16.6.1
- CUCM UCOS 12.0.1.10000-10 for 1 Publisher and 2 Subscriber
- Cisco IOS v15.1 for the fax gateway

Features

Features Supported

- Incoming and outgoing off-net calls using G711ulaw and G729 voice codecs
- International Calls and digit manipulations
- Call Conference
- CPE Voice Mail support
- Call hold & Resume with and without MoH
- Unattended and Attended Call transfer
- Call forward (all, busy and no answer)
- DTMF (RFC2833)
- Fax (T38 and G711 Pass-through)
- IP-PBX Calling number privacy
- High Availability

Features Not Supported

- Cisco UCM does not support Blind Call transfer
- In HA Redundancy mode, the Primary Cisco UBE will not take over the Primary/Active role after a reboot/network outage
- Cox does not support G729 codec

Caveats

- Caller ID does not get updated on attended and unattended transfer scenarios.
- Only one IP PBX used for the testing.
- The Cisco UBE HA tested is layer 2 box to box Cisco UBE redundancy.
- The maximum fax rate achieved using T.38 (G3 or SG3) fax protocol is only 14400 kbps
- For T38 test related scenario (G3 or SG3) achieved using “fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none” on dial peer.
- For G711Passthrough test achieved using “fax protocol pass-through g711ulaw

Configuration

Configuring Cisco Unified Border Element

The following configuration snippet contains a sample configuration of Cisco Unified Border Element.

Network Interface

The IP address used are for illustration only, the actual IP address can vary. The Active/Standby pair share the same virtual IP address and continually exchange status messages.

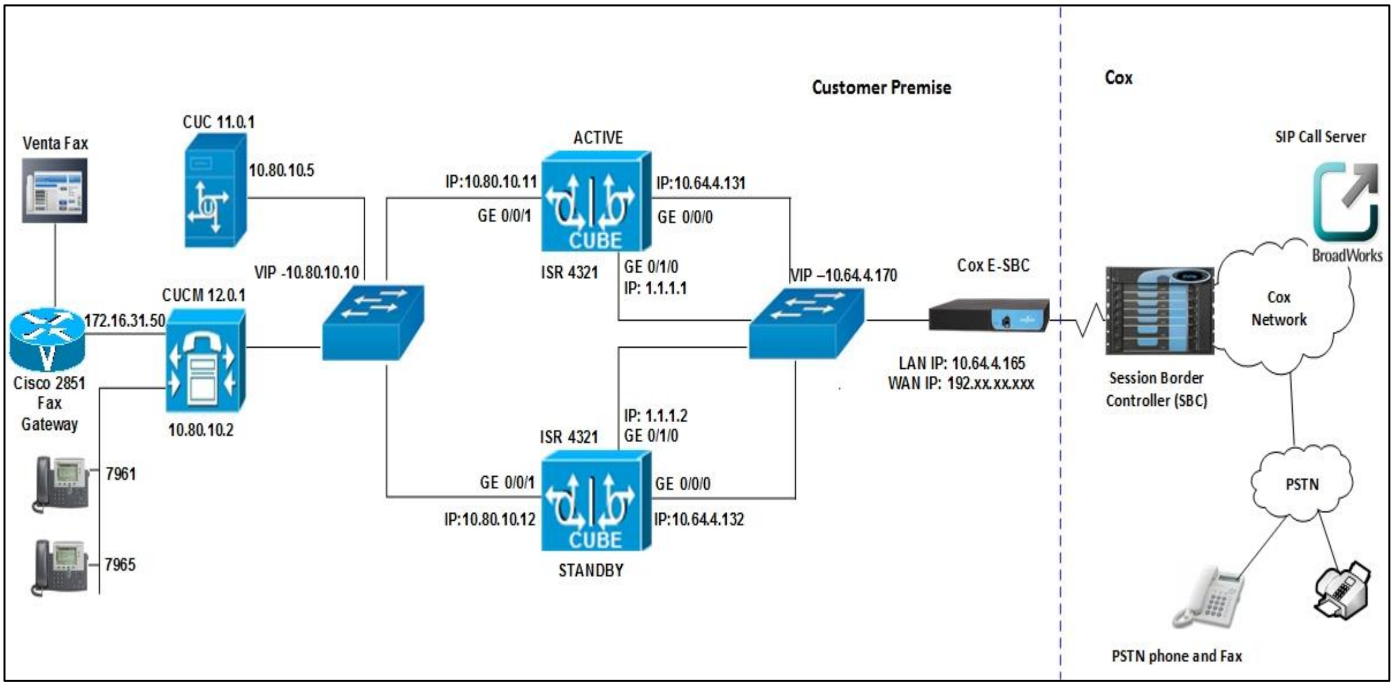


Figure 2 High Availability topology

Cisco UBE 1:

```
interface GigabitEthernet0/0/0
 description Cox WAN
 ip address 10.64.4.131 255.255.0.0
 negotiation auto
 redundancy rii 4
 redundancy group 1 ip 10.64.4.170 exclusive
!
!
interface GigabitEthernet0/0/1
description Cox LAN
 ip address 10.80.10.11 255.255.255.0
 negotiation auto
 redundancy rii 3
 redundancy group 1 ip 10.80.10.10 exclusive
!
!
interface GigabitEthernet0/0/2
 description Cox CubeHA Interface
 ip address 1.1.1.1 255.255.255.252
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
```

Cisco UBE 2:

```
interface GigabitEthernet0/0/0
 description Cox WAN
 ip address 10.64.4.132 255.255.0.0
 negotiation auto
 redundancy rii 4
 redundancy group 1 ip 10.64.4.170 exclusive
!
interface GigabitEthernet0/0/1
 description Cox LAN
 ip address 10.80.10.12 255.255.255.0
 negotiation auto
 redundancy rii 3
 redundancy group 1 ip 10.80.10.10 exclusive
!
interface GigabitEthernet0/0/2
 description Cox CubeHA Interface
 ip address 1.1.1.2 255.255.255.252
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
```

Global Cisco UBE settings

In order to enable Cisco UBE IP2IP SBC functionality, following command has to be entered:

```
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
 bind control source-interface GigabitEthernet0/0/0
  rel1xx disable
  session refresh
  header-passing
  asserted-id pai
  privacy pstn
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
```

Explanation

Command	**Description**
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
redundancy-group 2	Enable High Availability for the VoIP service
fax protocol	Specifies the fax protocol
asserted-id	Specifies the privacy header in the outgoing SIP requests and response messages

Codecs

G711ulaw and G729 voice codecs are used for this testing. Codec preferences used to change according to the test plan description.

```
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class codec 3
 codec preference 1 g729r8
!
```

Dial peer

Outbound Dial-peer to Cox:

```
dial-peer voice 510 voip
 description outgoing call to Cox - Lan Facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
!
```

```
dial-peer voice 520 voip
 description outgoing call to Cox - WAN Facing
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip profiles 101
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
```

Inbound Dial-peer from Cox:

```
dial-peer voice 610 voip
 description incoming call to PBX-WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number 402.......
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
```

```
!
dial-peer voice 620 voip
 description incoming call to PBX-LAN facing
 huntstop
 destination-pattern 402.......
 session protocol sipv2
 session target ipv4:10.80.10.2
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
```

Configuration example

The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

Active Cisco UBE:

```
User Access Verification

Username: cisco

Password:

CoxCertCube7#sh run


version 16.6

service timestamps debug datetime msec

service timestamps log datetime msec

service password-encryption

platform qfp utilization monitor load 80

no platform punt-keepalive disable-kernel-core

!

hostname CoxCertCube7

!

boot-start-marker

boot-end-marker

!

vrf definition Mgmt-intf

 !

 address-family ipv4

 exit-address-family

 !

 address-family ipv6

 exit-address-family

!

enable secret 5 $1$k8fq$3FBrZ5Ub4lioq2gWvC3e0
```

```
no aaa new-model
no ip domain lookup
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-2194658987
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2194658987
 revocation-check none
 rsakeypair TP-self-signed-2194658987
!
!
crypto pki certificate chain TP-self-signed-2194658987
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  bind control source-interface GigabitEthernet0/0/0
  rel1xx disable
  session refresh
  header-passing
  asserted-id pai
  privacy pstn
```

```
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class codec 3
 codec preference 1 g729r8
!
!
voice class sip-profiles 100
!
voice class sip-profiles 101
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>"
"<sip:402932\1@\2>"
!
voice translation-rule 1
 rule 1 /^.*\(..........\)/ /\1/
!
!
voice translation-profile Cox
 translate called 1
!
voice-card 0/4
```

```
 no watchdog
!
license udi pid ISR4331/K9 sn FDO21381GMV
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 0010160D320A11575F2F
!
redundancy
 mode none
 application redundancy
  group 1
   name coxcubecucm
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/2 protocol 1
   data GigabitEthernet0/0/2
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
 description Cox WAN
 ip address 10.64.4.131 255.255.0.0
 negotiation auto
 redundancy rii 4
```

```
 redundancy group 1 ip 10.64.4.170 exclusive
!
interface GigabitEthernet0/0/1
 ip address 10.80.10.11 255.255.255.0
 negotiation auto
 redundancy rii 3
 redundancy group 1 ip 10.80.10.10 exclusive
!
interface GigabitEthernet0/0/2
 ip address 1.1.1.1 255.255.255.252
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.0.0 255.255.0.0 10.80.10.1
ip route 172.16.0.0 255.255.0.0 10.80.10.1
ip route 172.16.24.0 255.255.248.0 10.80.10.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
```

```
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
control-plane
!
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 510 voip
 description outgoing call to Cox - Lan Facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 520 voip
 description outgoing call to Cox - WAN Facing
 huntstop
 destination-pattern .T
 session protocol sipv2
```

```
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip profiles 101
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 610 voip
 description incoming call to PBX-WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number 402.......
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 620 voip
 description incoming call to PBX-LAN facing
 huntstop
```

```
 destination-pattern 402.......
 session protocol sipv2
 session target ipv4:10.80.10.2
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 700 voip
 description PBX to PBX - LAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 1402.......
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 710 voip
 description PBX to PBX - WAN facing
```

```
 translation-profile outgoing Cox
 huntstop
 destination-pattern 1402.......
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
sip-ua
 keepalive target ipv4:10.64.4.165
 retry invite 3
 timers keepalive active 10
 sip-server ipv4:10.64.4.165
 connection-reuse
!
line con 0
 exec-timeout 0 0
 password 7 01100F175804
 login
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
```

```
 password 7 070C285F4D06
 login local
 transport input telnet
!
ntp server 34.208.249.133
!
end
```

Standby Cisco UBE:

```
CoxCertCUBE8#sh run


version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname CoxCertCUBE8
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
enable secret 5 $1$qL.R$XtXGz/0yrmPBmBl5S1b.1
!
no aaa new-model
no ip domain lookup
subscriber templating
```

```
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-3616943619
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-3616943619
 revocation-check none
 rsakeypair TP-self-signed-3616943619
!
crypto pki certificate chain TP-self-signed-3616943619
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  bind control source-interface GigabitEthernet0/0/0
  rel1xx disable
  session refresh
  header-passing
  asserted-id pai
  privacy pstn
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
```

```
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class codec 3
 codec preference 1 g729r8
!
voice class sip-profiles 100
!
voice class sip-profiles 101
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>"
"<sip:402932\1@\2>"
!
voice translation-rule 1
 rule 1 /^.*\(..........\)/ /\1/
!
voice translation-profile Cox
 translate called 1
!
voice-card 0/4
 no watchdog
!
license udi pid ISR4331/K9 sn FDO21381FEY
diagnostic bootup level minimal
spanning-tree extend system-id
!
```

```
username cisco privilege 15 password 7 0010160D320A11575F2F
!
redundancy
 mode none
 application redundancy
  group 1
   name coxcubecucm
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/2 protocol 1
   data GigabitEthernet0/0/2
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
 description Cox WAN
 ip address 10.64.4.132 255.255.0.0
 negotiation auto
 redundancy rii 4
 redundancy group 1 ip 10.64.4.170 exclusive
!
interface GigabitEthernet0/0/1
 description Cox LAN
 ip address 10.80.10.12 255.255.255.0
 negotiation auto
 redundancy rii 3
```

```
 redundancy group 1 ip 10.80.10.10 exclusive
!
interface GigabitEthernet0/0/2
 ip address 1.1.1.2 255.255.255.252
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 10.80.0.0 255.255.0.0 10.80.10.1
ip route 172.16.24.0 255.255.248.0 10.80.10.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
control-plane
!
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 510 voip
 description outgoing call to Cox - Lan Facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 520 voip
 description outgoing call to Cox - WAN Facing
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
```

```
 voice-class sip profiles 101
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 610 voip
 description incoming call to PBX-WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number 402.......
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 620 voip
 description incoming call to PBX-LAN facing
 huntstop
 destination-pattern 402.......
 session protocol sipv2
 session target ipv4:10.80.10.2:5060
 session transport udp
 voice-class codec 1
```

```
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 700 voip
 description PBX to PBX - LAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 1402.......
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 710 voip
 description PBX to PBX - WAN facing
 translation-profile outgoing Cox
 huntstop
 destination-pattern 1402.......
 session protocol sipv2
 session target sip-server
```

```
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
sip-ua
 keepalive target ipv4:10.64.4.165
 retry invite 3
 timers keepalive active 10
 sip-server ipv4:10.64.4.165
 connection-reuse
!
line con 0
 exec-timeout 0 0
 password 7 00071A150754
 login
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 password 7 104D000A0618
 login local
 transport input telnet
!
ntp server 34.202.215.187
```

```
ntp server pool.ntp.org
!
end
```

Configuring Cisco UCM 11.5 Cluster

Cisco UCM Version

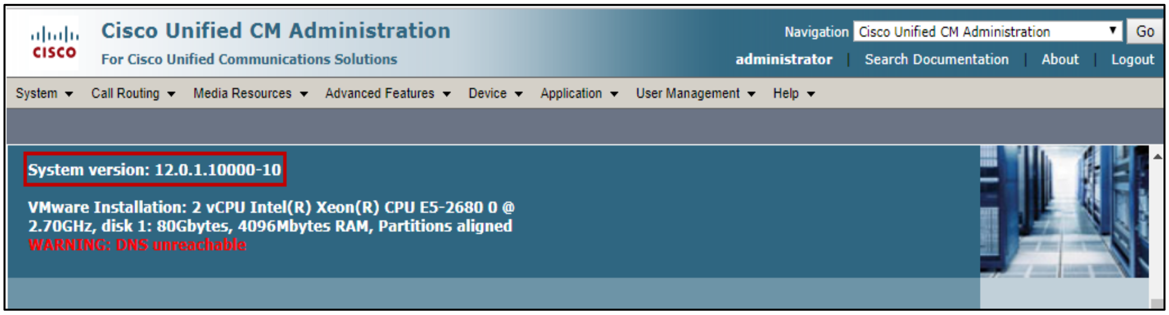


Figure 3: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

- Select Server* = **Clus20pub--CUCM Voice/Video (Active)**
- Select Service*= **Cisco CallManager (Active)**
- Duplex Streaming Enabled* = **True**
- All other fields are set to default values

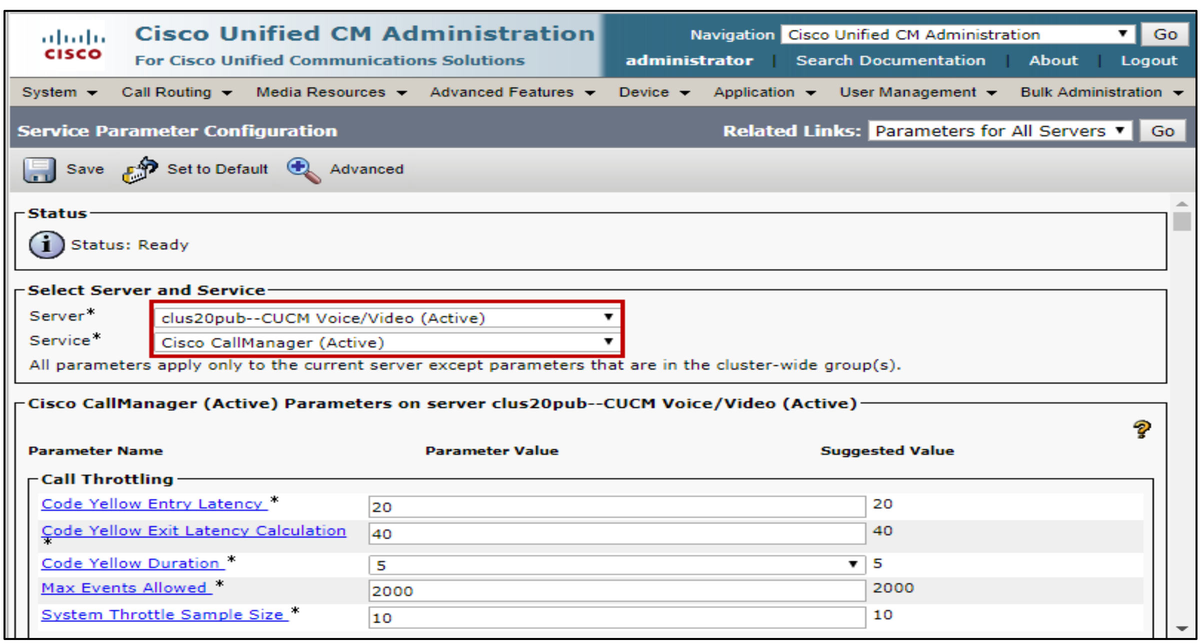


Figure 4: Service Parameters

Clusterwide Parameters (Service)

Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
Duplex Streaming Enabled *	True	False
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	12	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500
Media Resource Allocation Timer *	12	12
MTP and Transcoder Resource Throttling Percentage *	95	95
Intercluster Capabilities Mismatch Timer *	1000	1000
Silence Suppression *	True	False
Silence Suppression for Gateways *	True	False
Strip G.729 Annex B (Silence Suppression) from Capabilities *	True	False
Enable Source IP Address Verification for Software Media Devices *	True	True

Figure 5: Service Parameters (Cont.)

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

- Name*= **Non Secure SIP Trunk Profile-Cox-cube-cucm** is used as an example
- Description = **Non Secure SIP Trunk Profile authenticated by null String** is used as an example
- Device Security Mode = **Non Secure**
- Incoming Transport Type* = **TCP + UDP**
- Outgoing Transport Type = **UDP**
- **Accept unsolicited Notification** is enabled

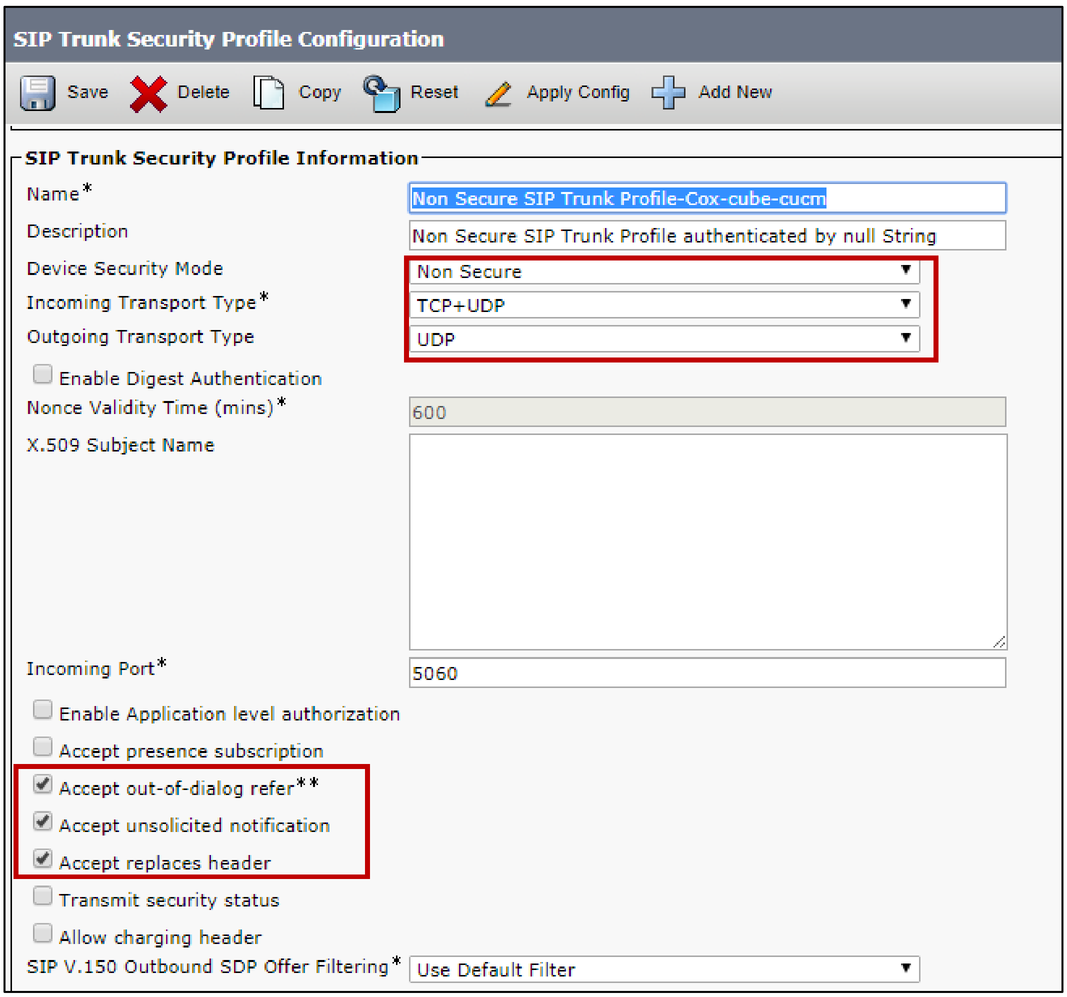


Figure 6: SIP Trunk Security Profile

SIP Profile

Navigation: Device → Device Settings → SIP Profile

- Name*= **Standard SIP Profile_CUBE_CUCM_COX** is used as an example
- Description = **SIP Profile for cox-cucm-cube-certification** is used as an example

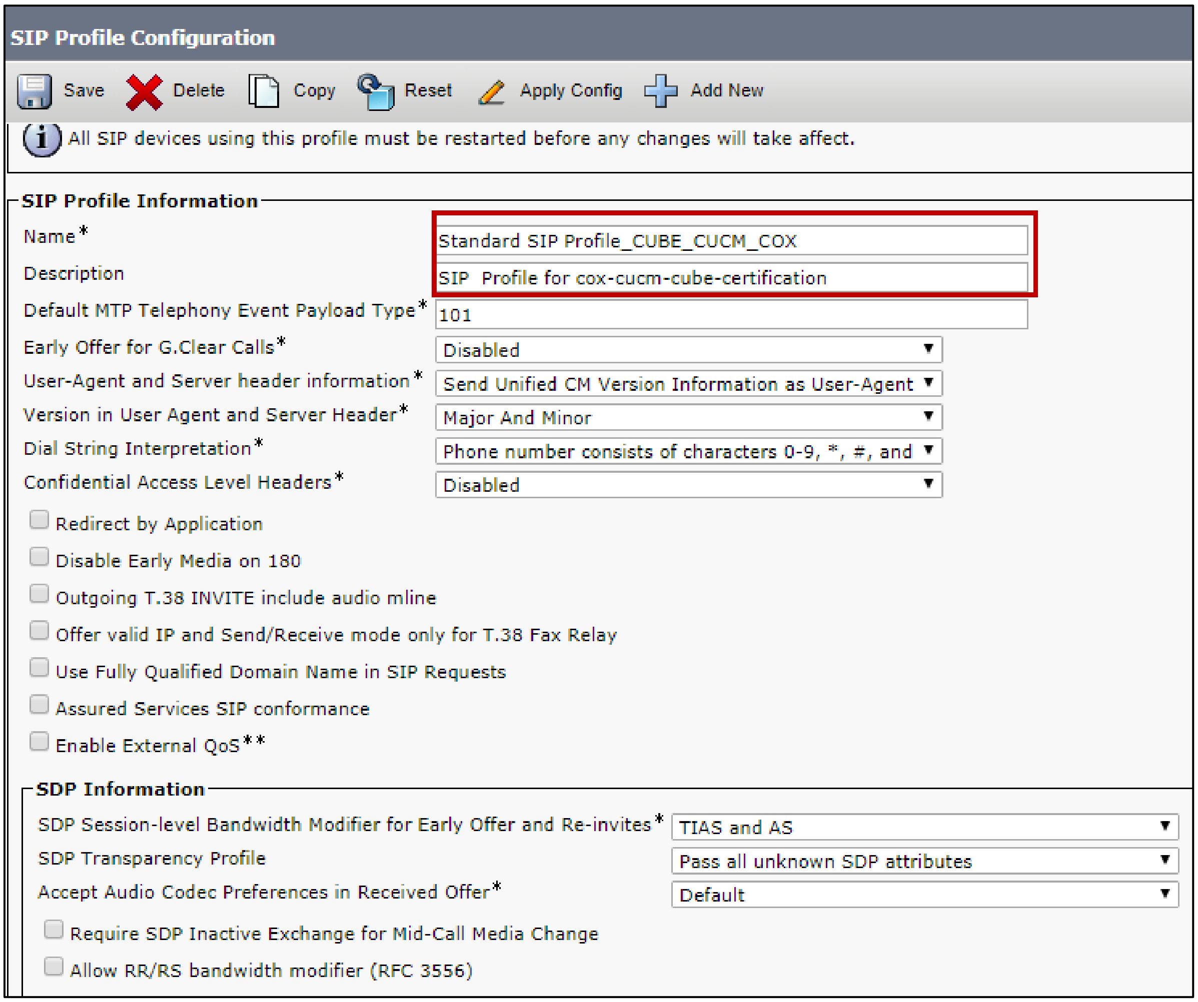
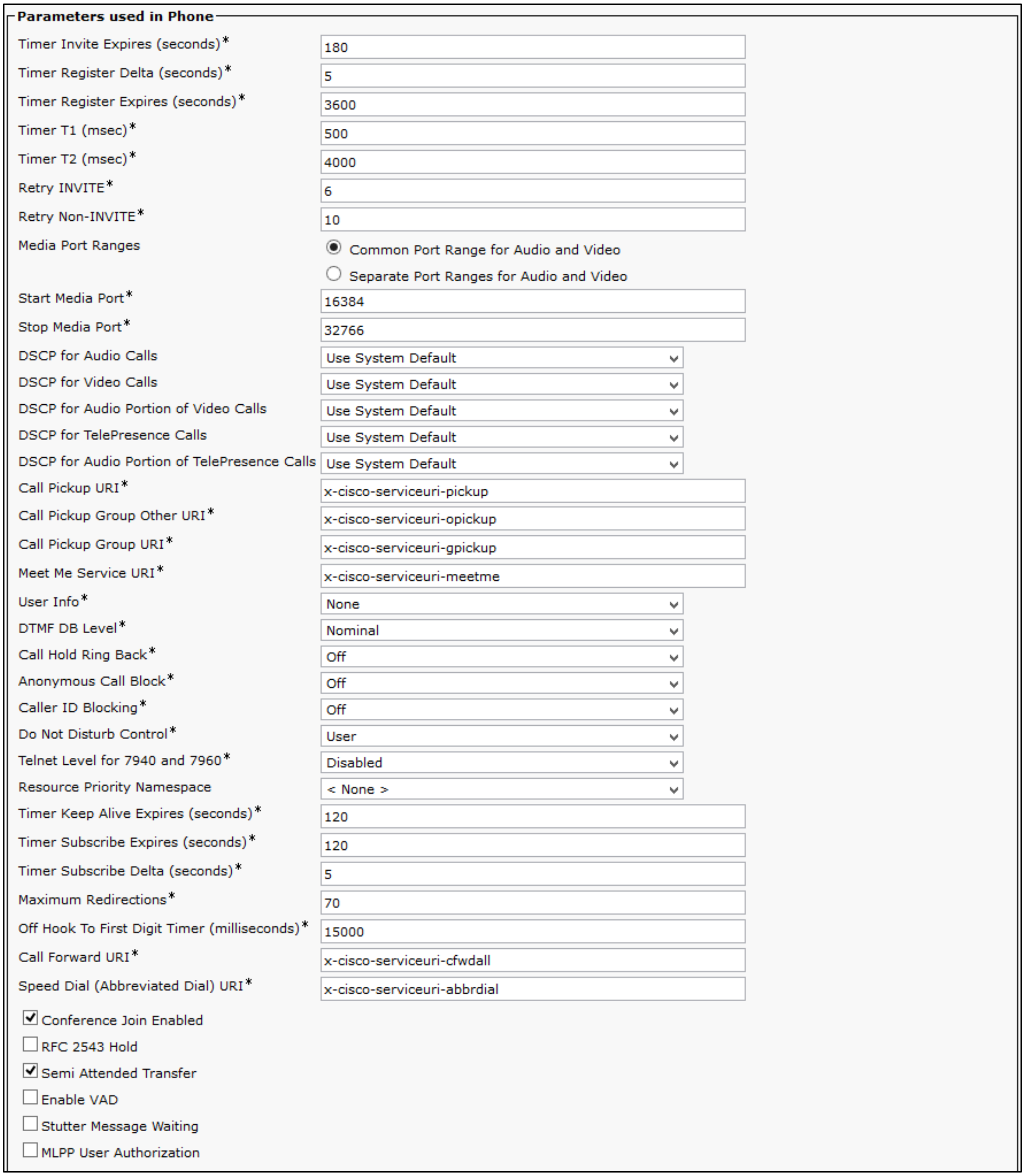


Figure 7: SIP Profile



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

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

Figure 8: SIP Profile (Cont.)

- "SIP Rel1XXX Options" is set to "**Send PRACK if 1xx Contains SDP**"

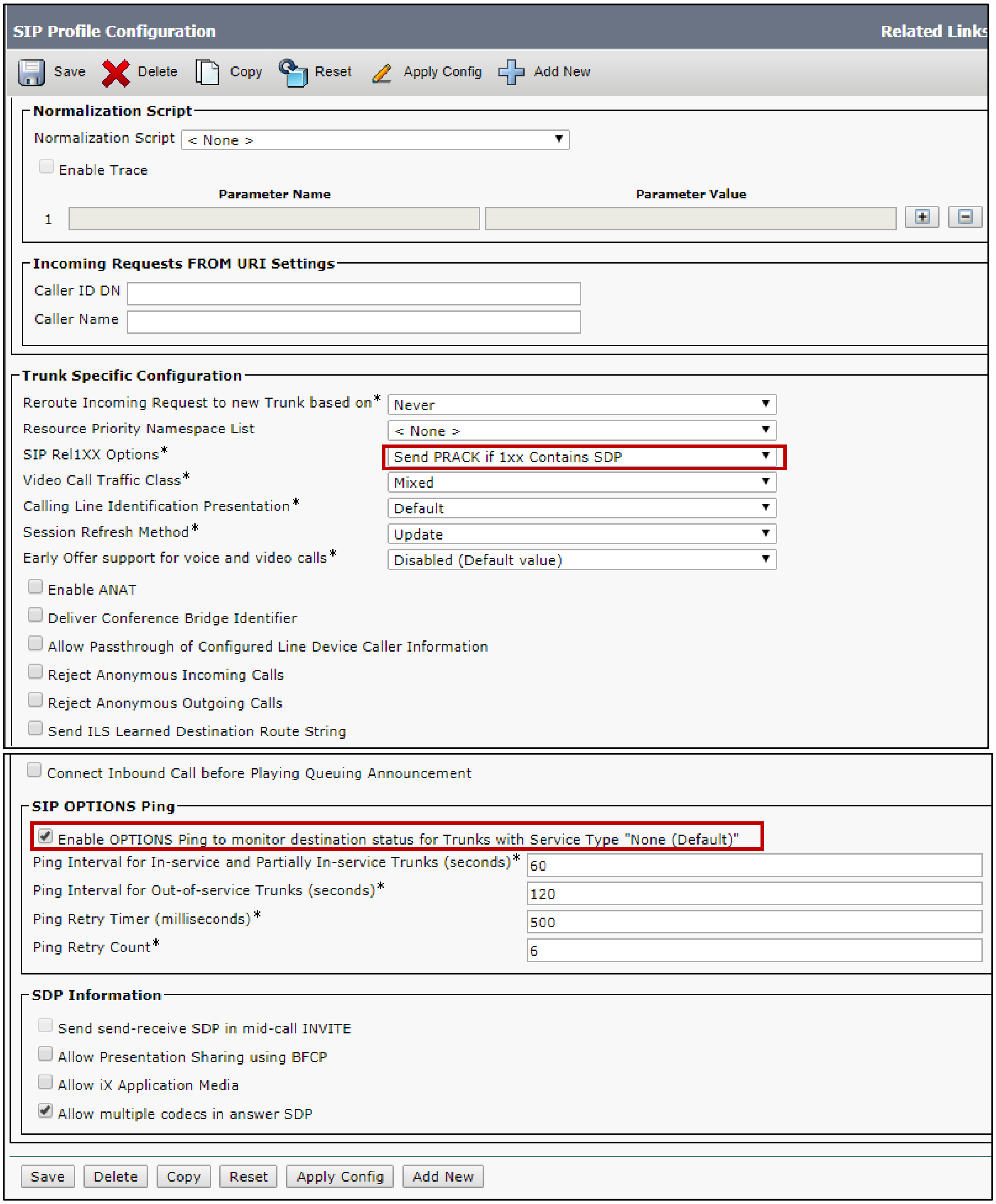


Figure 9: SIP Profile (Cont.)

Trunk configuration

Trunk configuration from Cisco UCM to the LAN side of the Cisco UBE:

Navigation: Device → Trunk → Add New

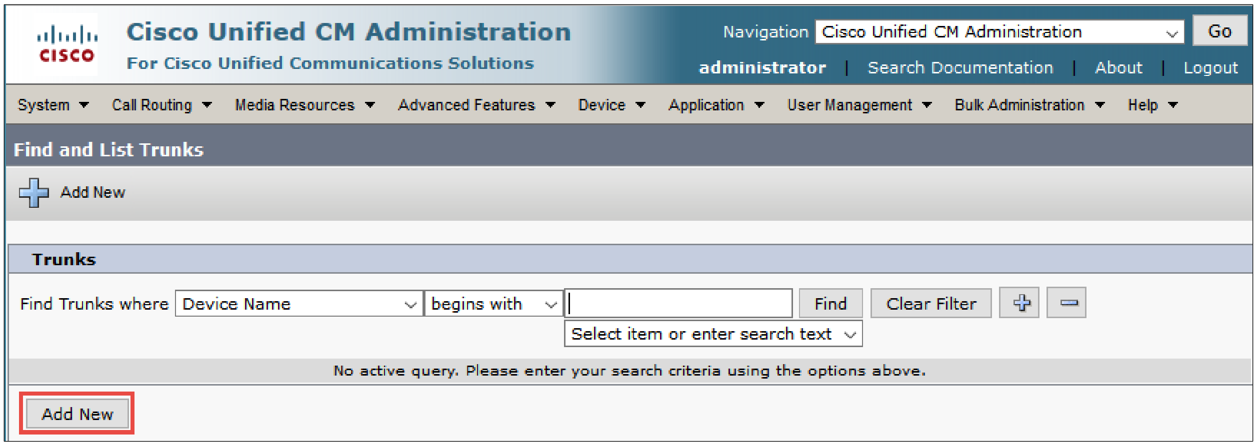


Figure 10: Add New Trunk to Cisco UBE

- Select 'Trunk Type' as **SIP Trunk** and 'Device Protocol' as **SIP** and select '**Next**' as shown below.

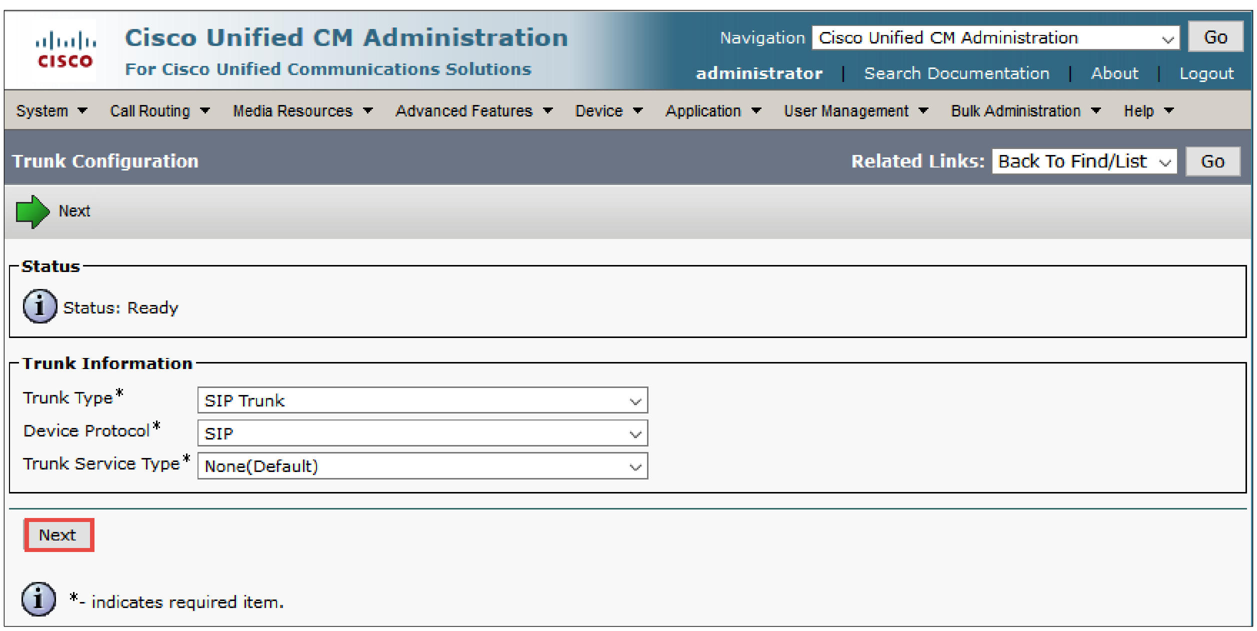


Figure 11: Add SIP Trunk Type

- "Media Resource Group List" is set to **MRGL_NoMTP** for resources with ANN, CFB, MOH

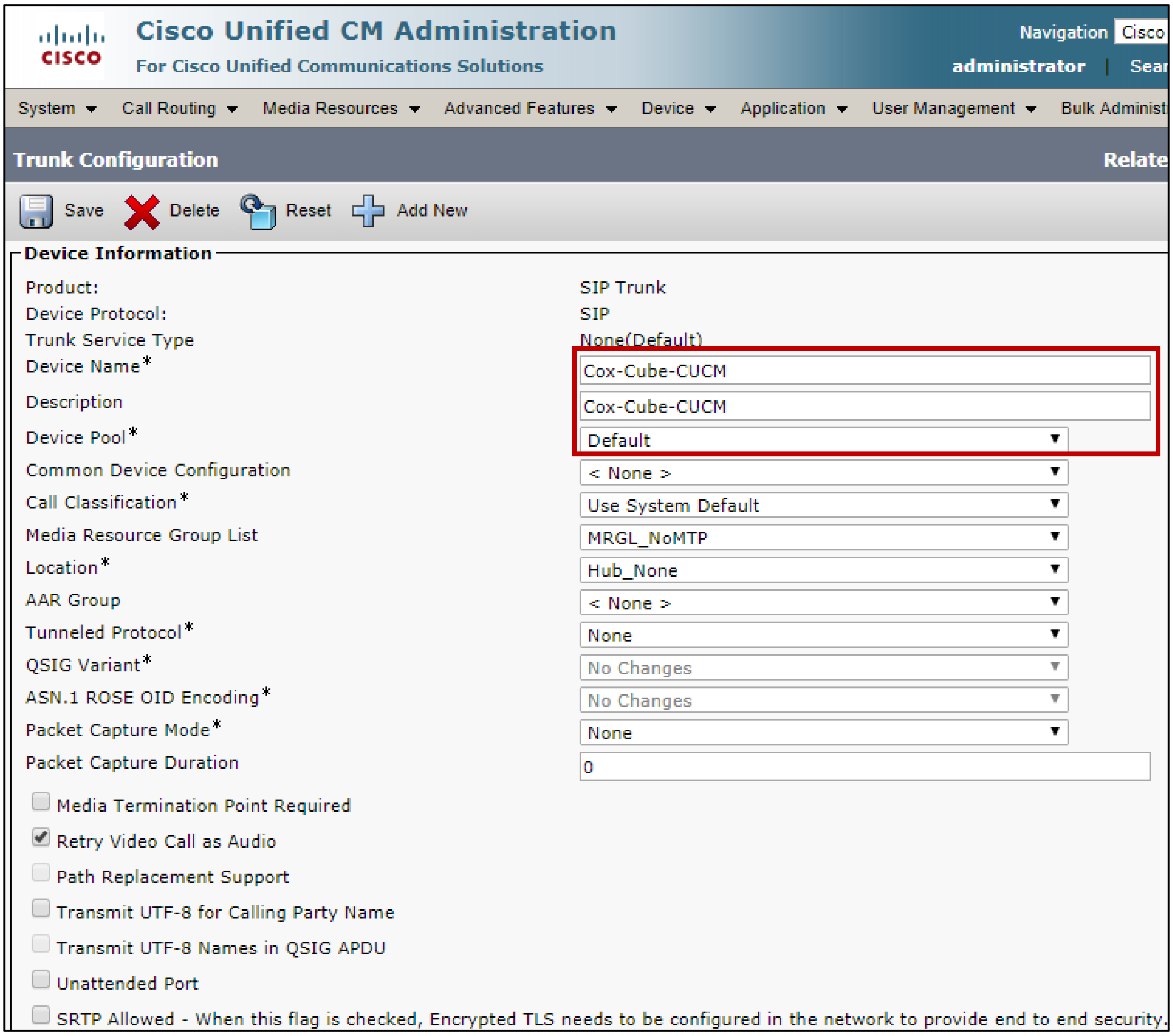


Figure 12: SIP Trunk to Cisco UBE

- "Significant Digits" set to **4.** 4 digits Extension for all CPE phones.

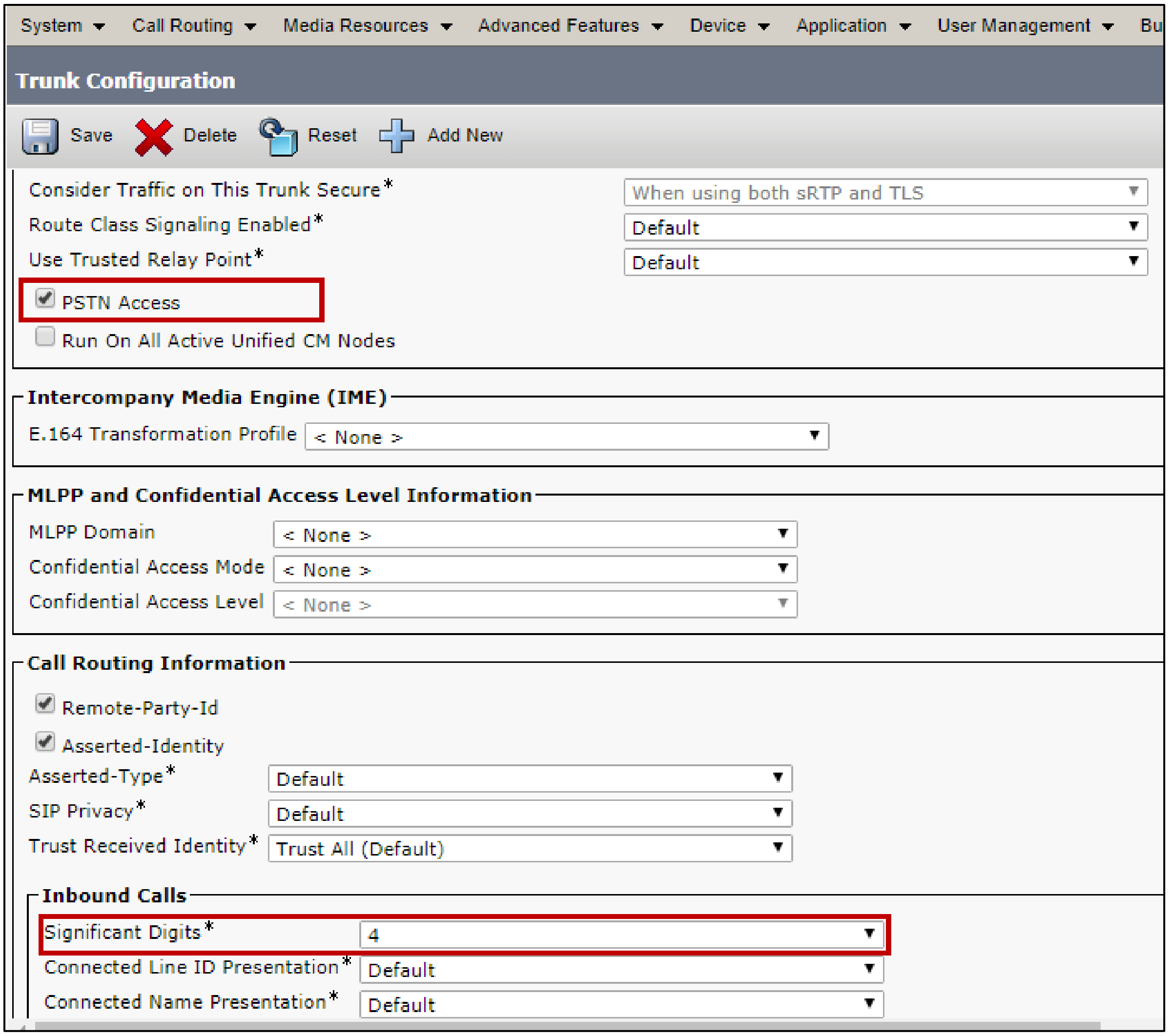
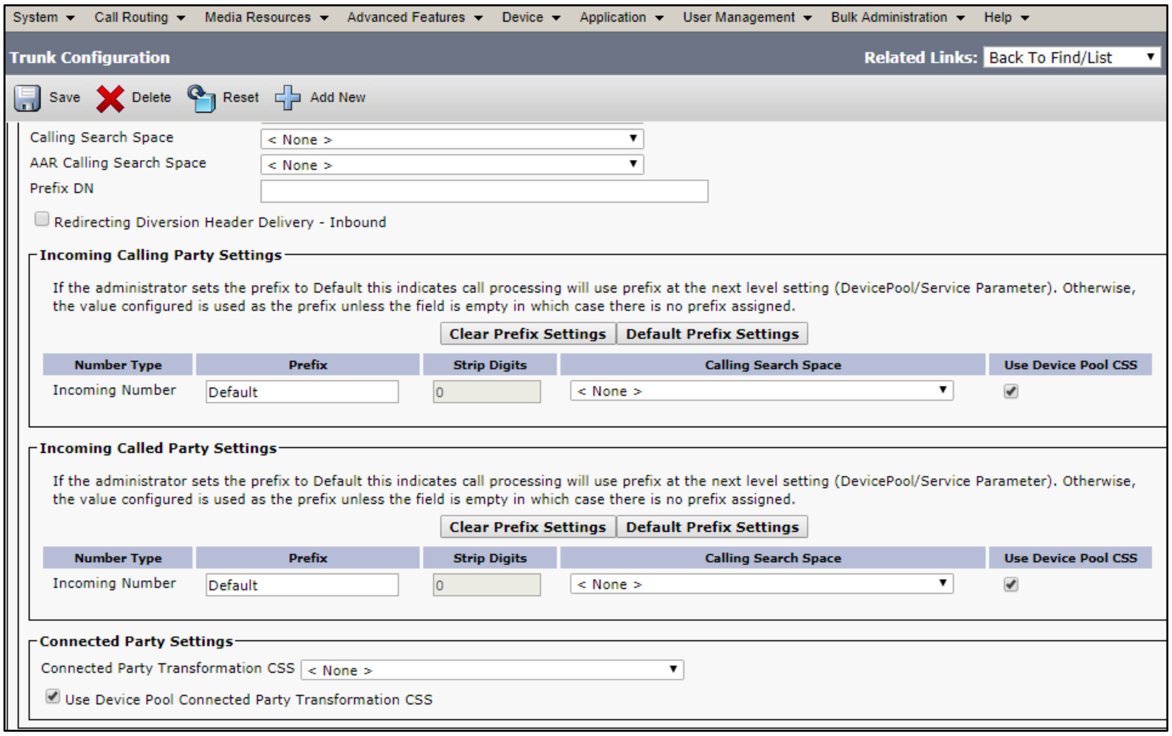


Figure 13: SIP Trunk to Cisco UBE (Cont.)

- Configure the Virtual LAN IP address of the Cisco UBE and the Destination Port
- Configure the SIP Trunk Security Profile and SIP Profile as shown below
- **Redirecting Diversion Header Delivery-Outbound** is enabled to send the Diversion information during outbound calls
- The rest of the configuration is all default
- Click Save and Reset after completion



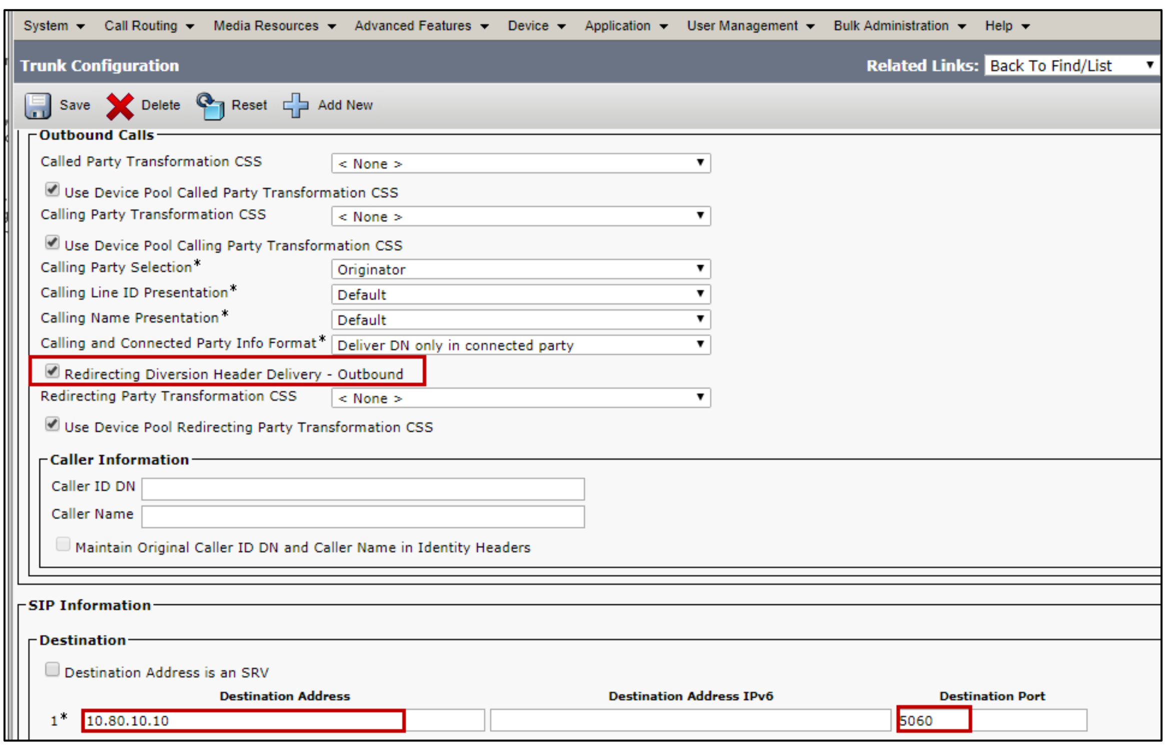


Figure 14: SIP Trunk to Cisco UBE (Cont.)

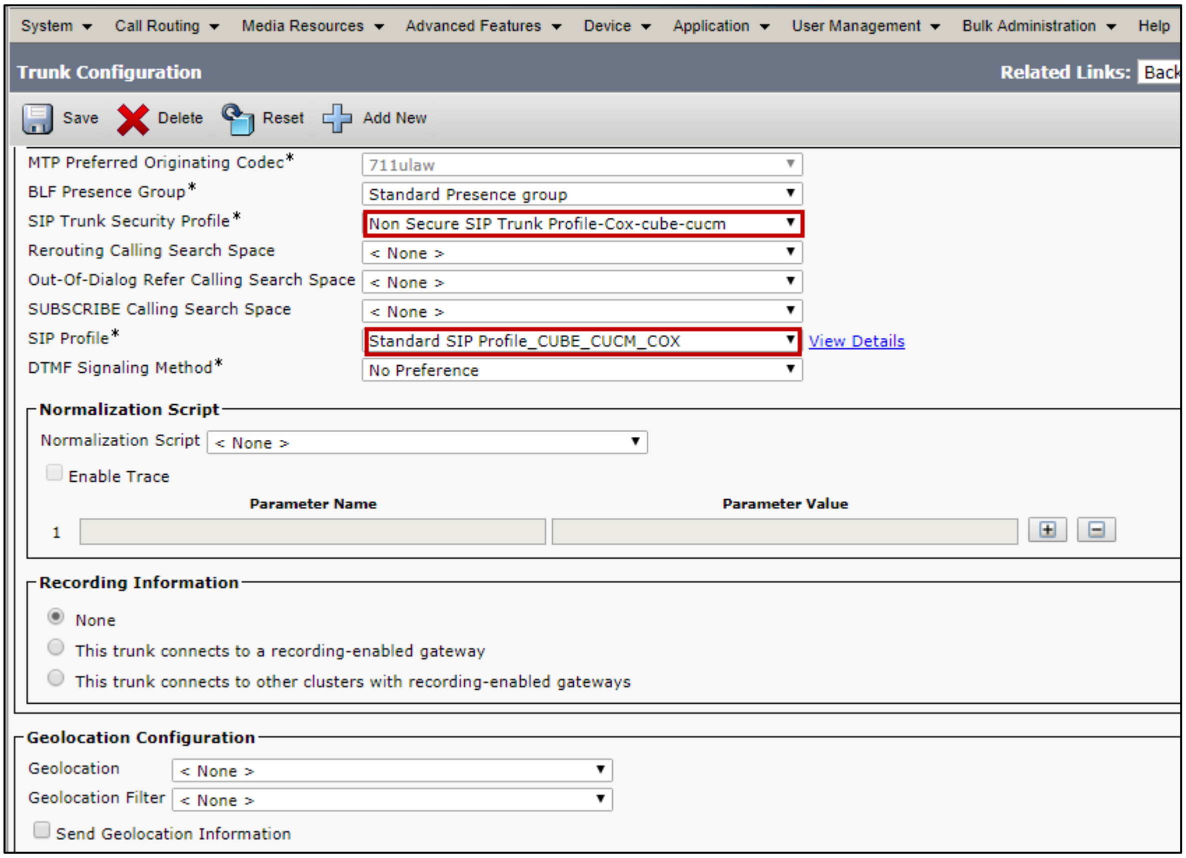


Figure 15: SIP Trunk to Cisco UBE (Cont.)

Trunk configuration from Cisco UCM to Fax Gateway:

Navigation: Devices → Trunk → Add New

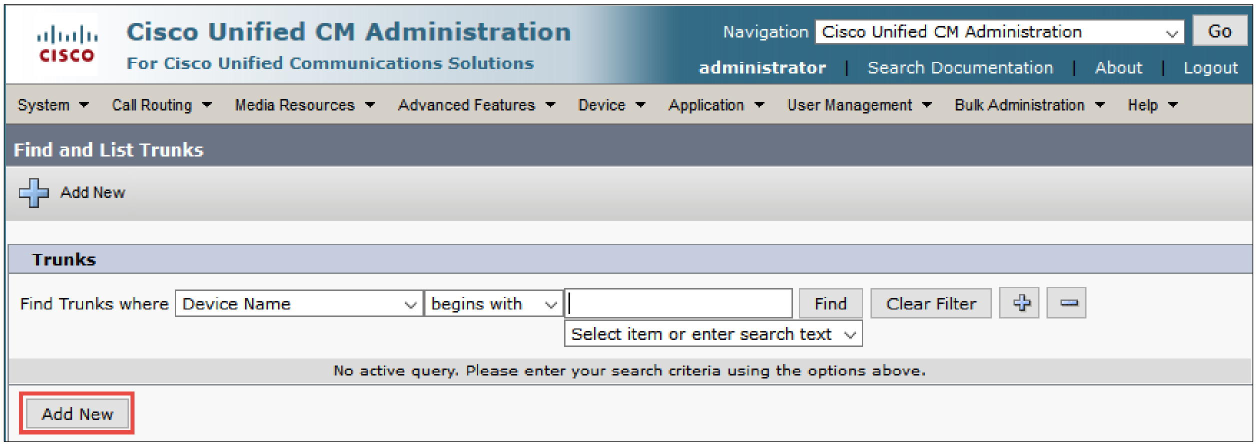


Figure 16: Add New Trunk to Fax Gateway

- Select 'Trunk Type' as **SIP Trunk** and 'Device Protocol' as **SIP** and select '**Next**' as shown below.

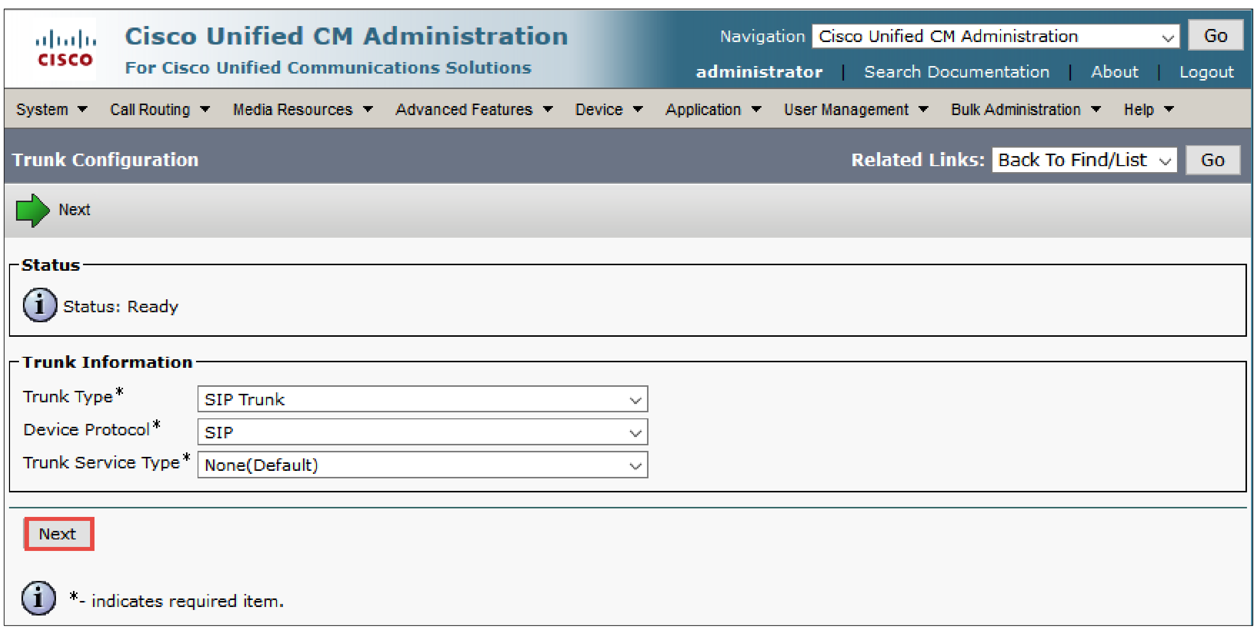


Figure 17: Add SIP Trunk Type

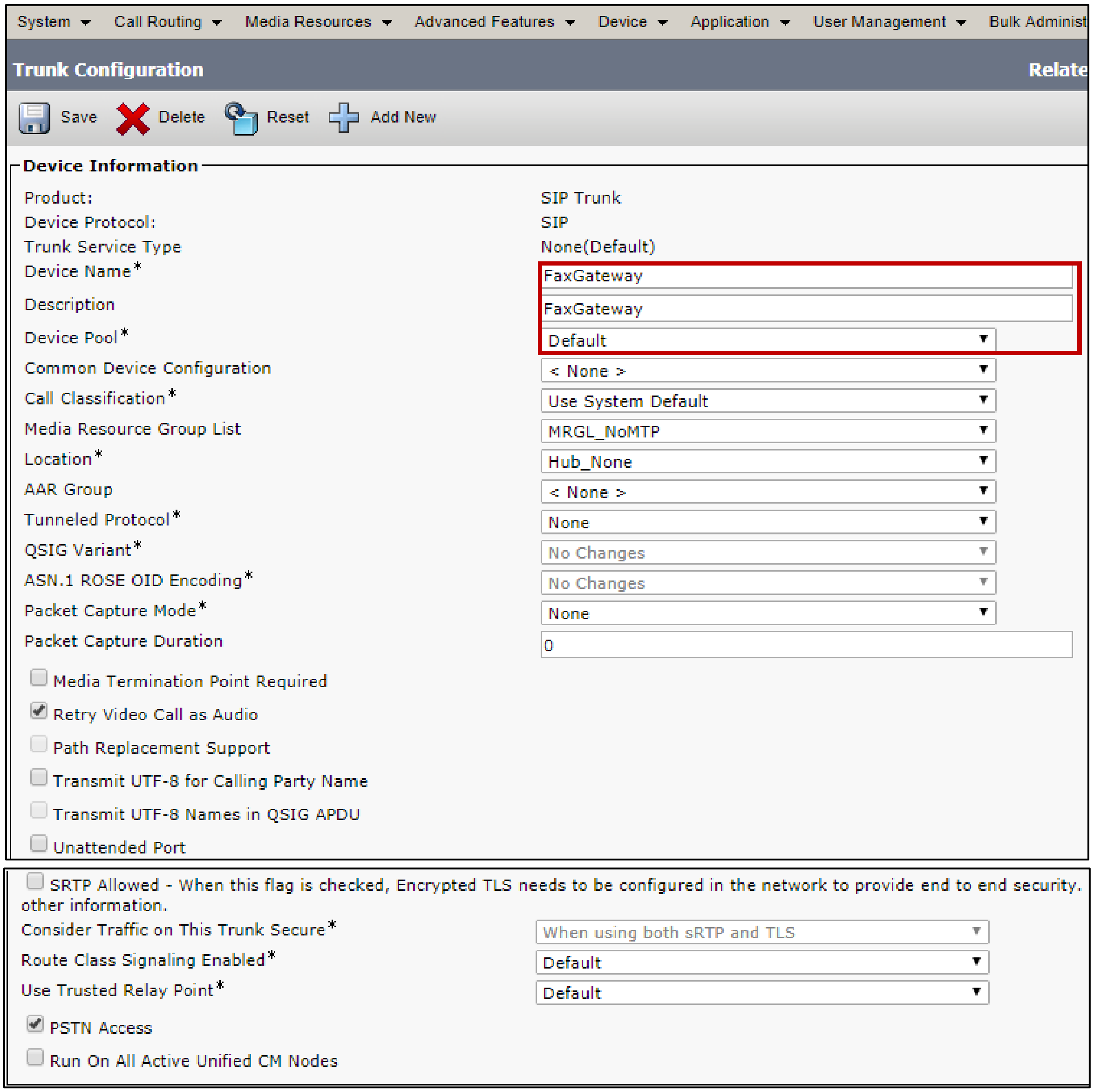
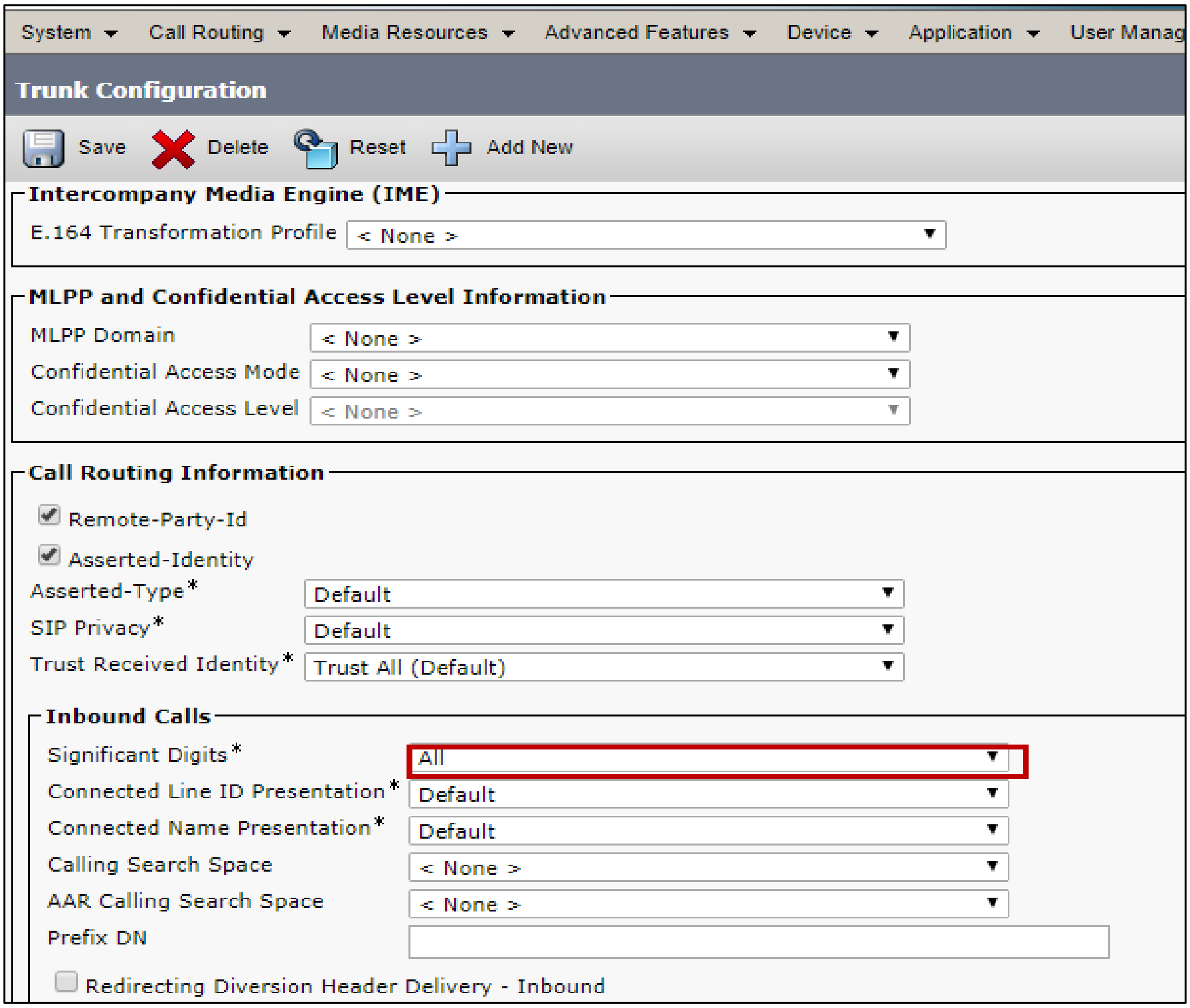


Figure 18: SIP Trunk to FAX Gateway



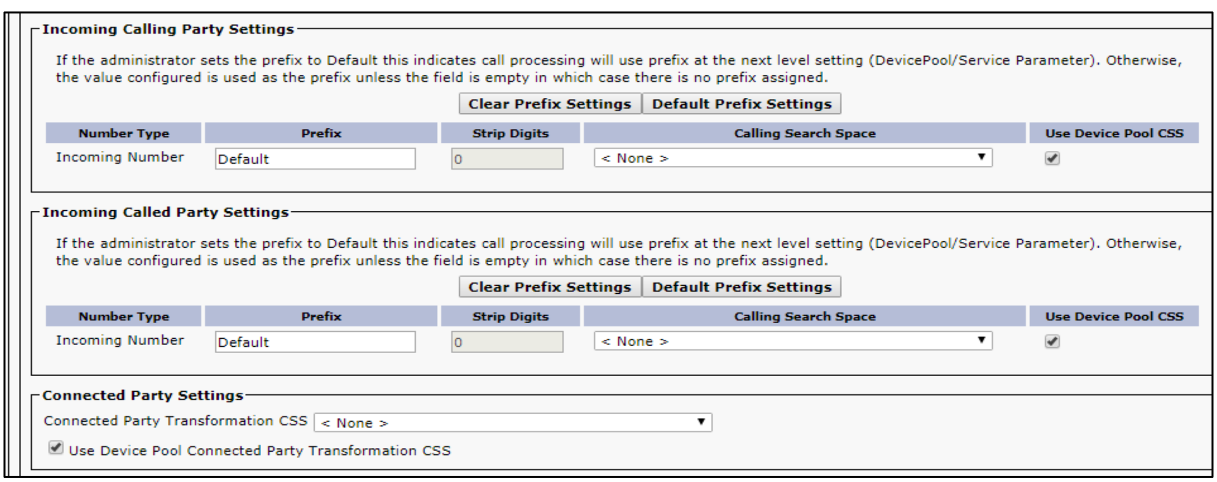


Figure 19: SIP Trunk to FAX Gateway (Cont.)

- Configure the IP address of Fax Gateway and the Destination Port

- Configure the SIP Trunk Security Profile and SIP Profile as shown below
- The rest of the configuration is all default
- Click Save and Reset after completion

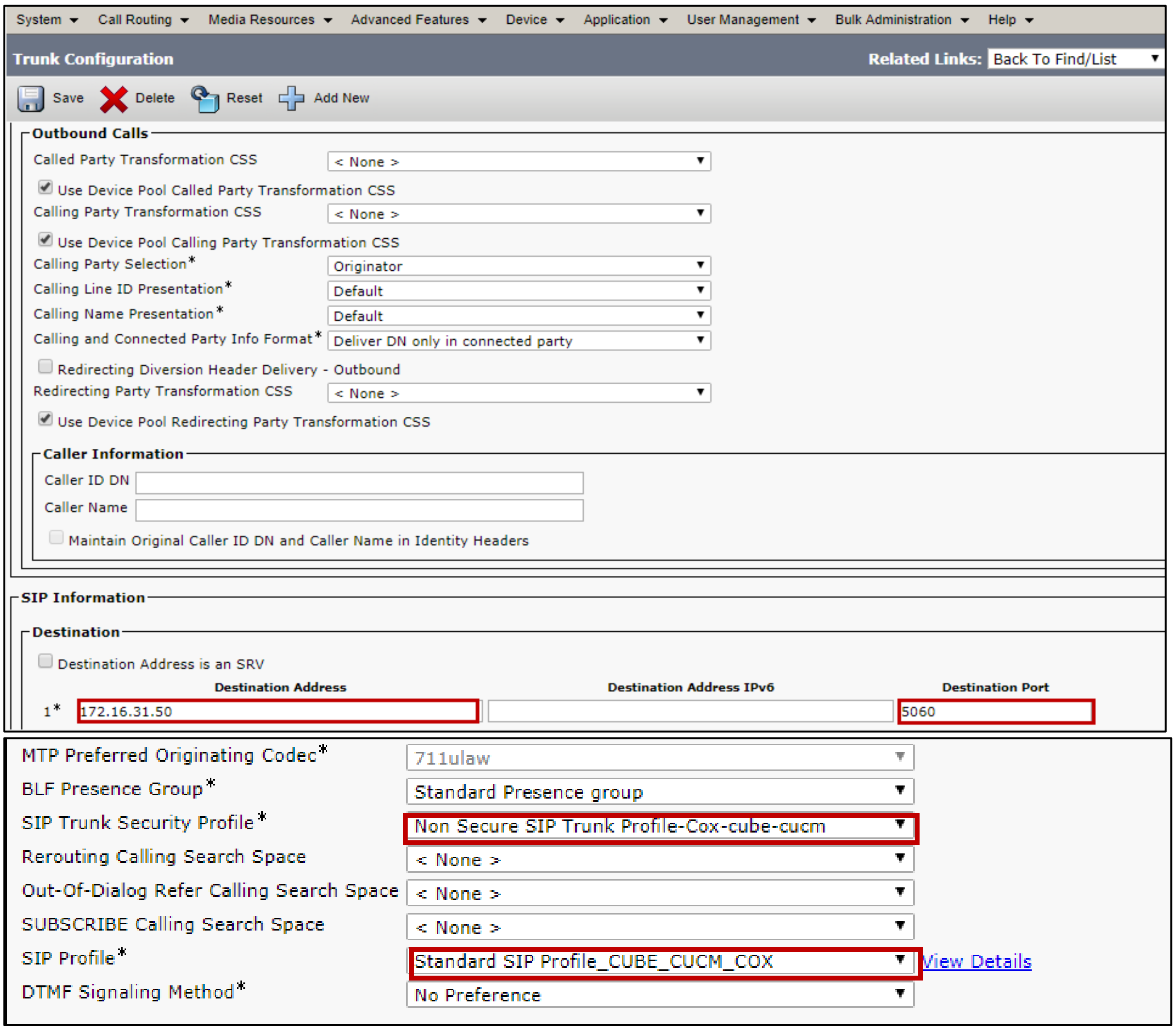


Figure 20: SIP Trunk to FAX Gateway (Cont.)

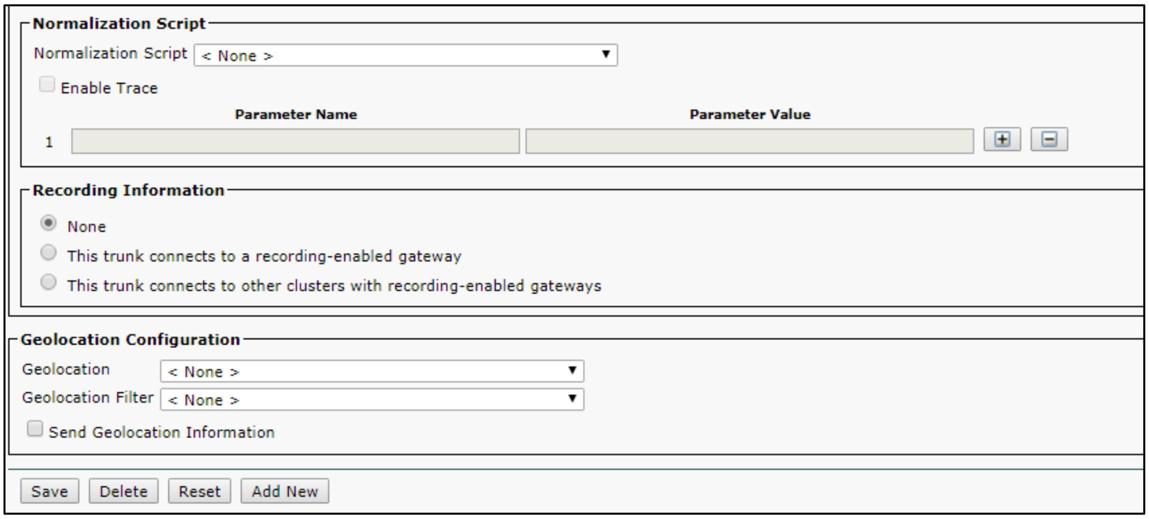


Figure 21: SIP Trunk to FAX Gateway (Cont.)

Routing configuration

Route Pattern for Cisco UBE:

Navigation: Call Routing → Route/Hunt → Route Pattern → Add New. Route patterns are configured as below.

- Cisco IP phone dial “9”+10 digits number to access PSTN via Cisco UBE.”9” is removed before sending to Cisco UBE.
- For FAX call, Access Code “9”+10 digits number is used at Cisco Fax gateway. “9” is removed at Cisco UCM. The rest of the number is sent to Cisco UBE to COX.
- Incoming fax call to 46XX will be sent to Cisco Fax gateway.
- For Anonymous call, access code “8*”+10 digits number is used. “8*” is removed at Cisco UCM. The rest of the number is sent to Cisco UBE to COX.
- Cisco IP phones dial “9” + X11 for emergency call and will send all digits to Cisco UBE to COX.

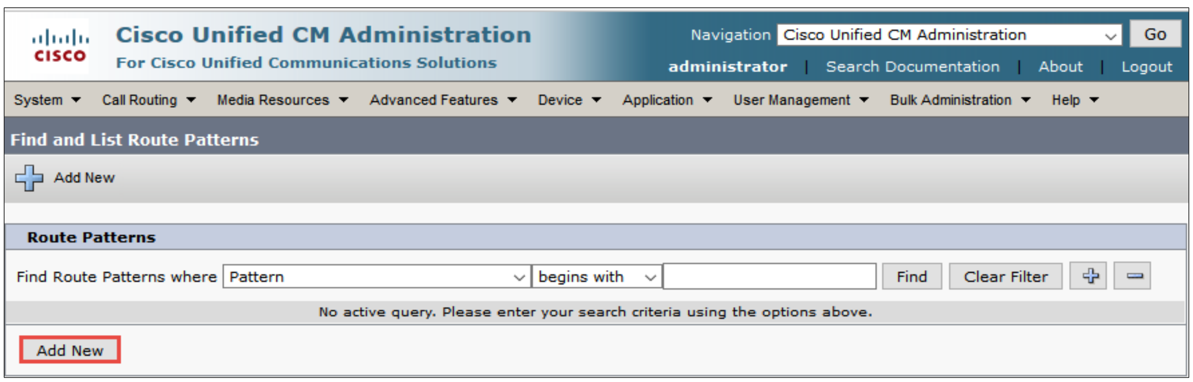


Figure 22: Add New Route Pattern for Cisco UBE

- Route Pattern: Specify appropriate **Route pattern**.
- Gateway/Route List: Select **Cox-Cube-CUCM**
- **Use Calling Party’s External Phone Number Mask** is enabled to send the 10-Digit Calling Number as DID when the call is routed via this route.

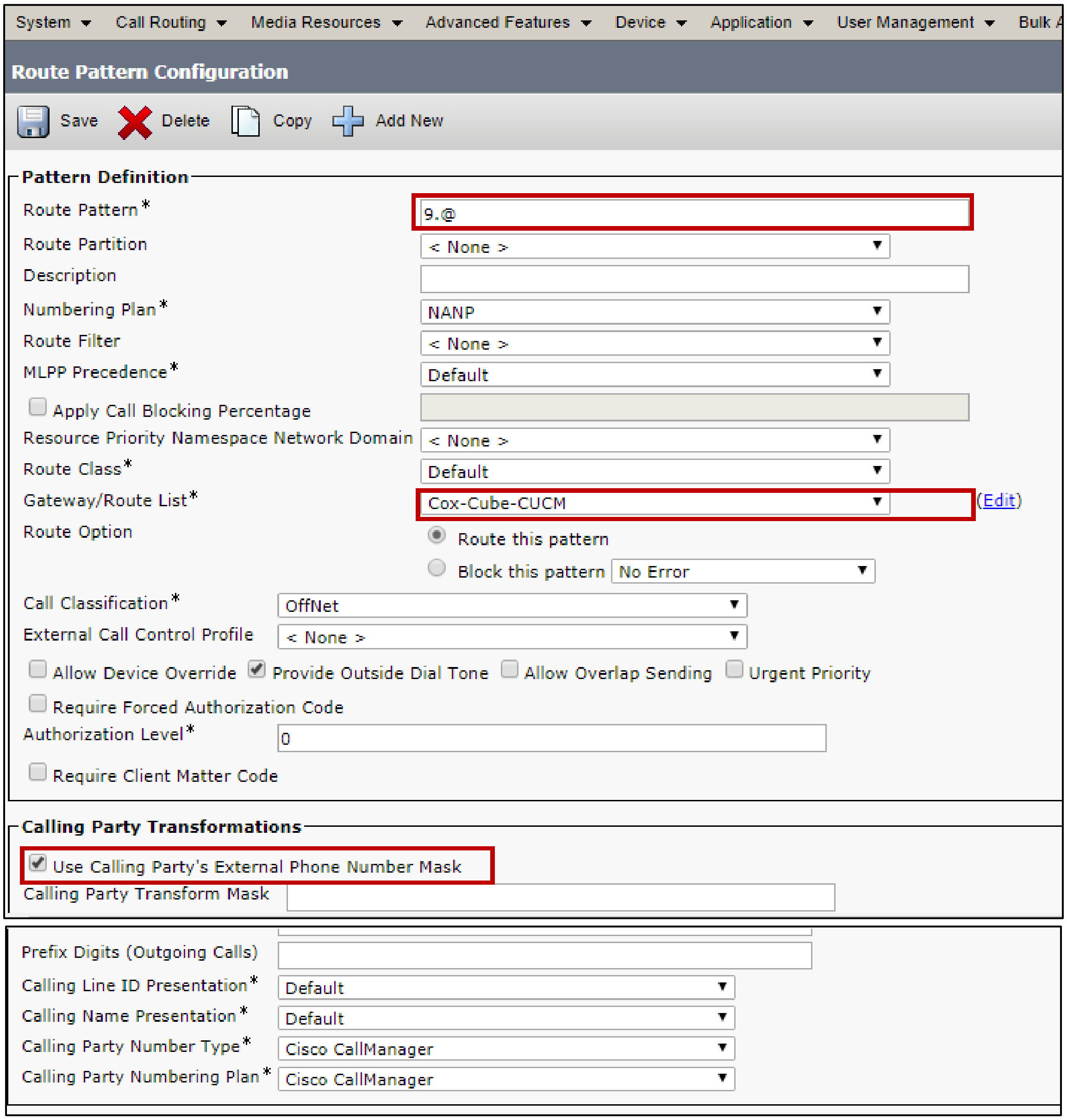


Figure 23: Route Pattern Configuration for Cisco UBE-PSTN Access

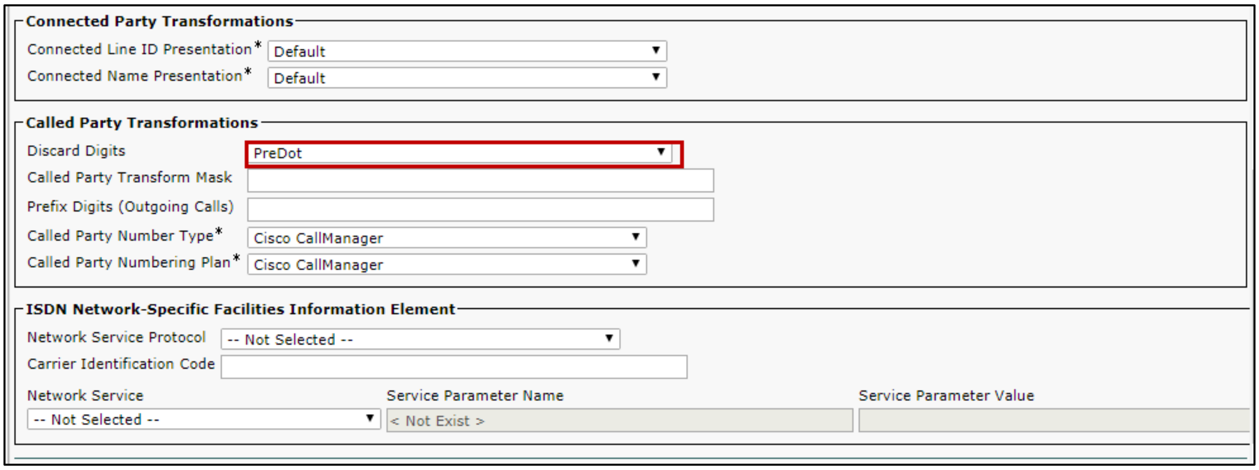


Figure 24: Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.)

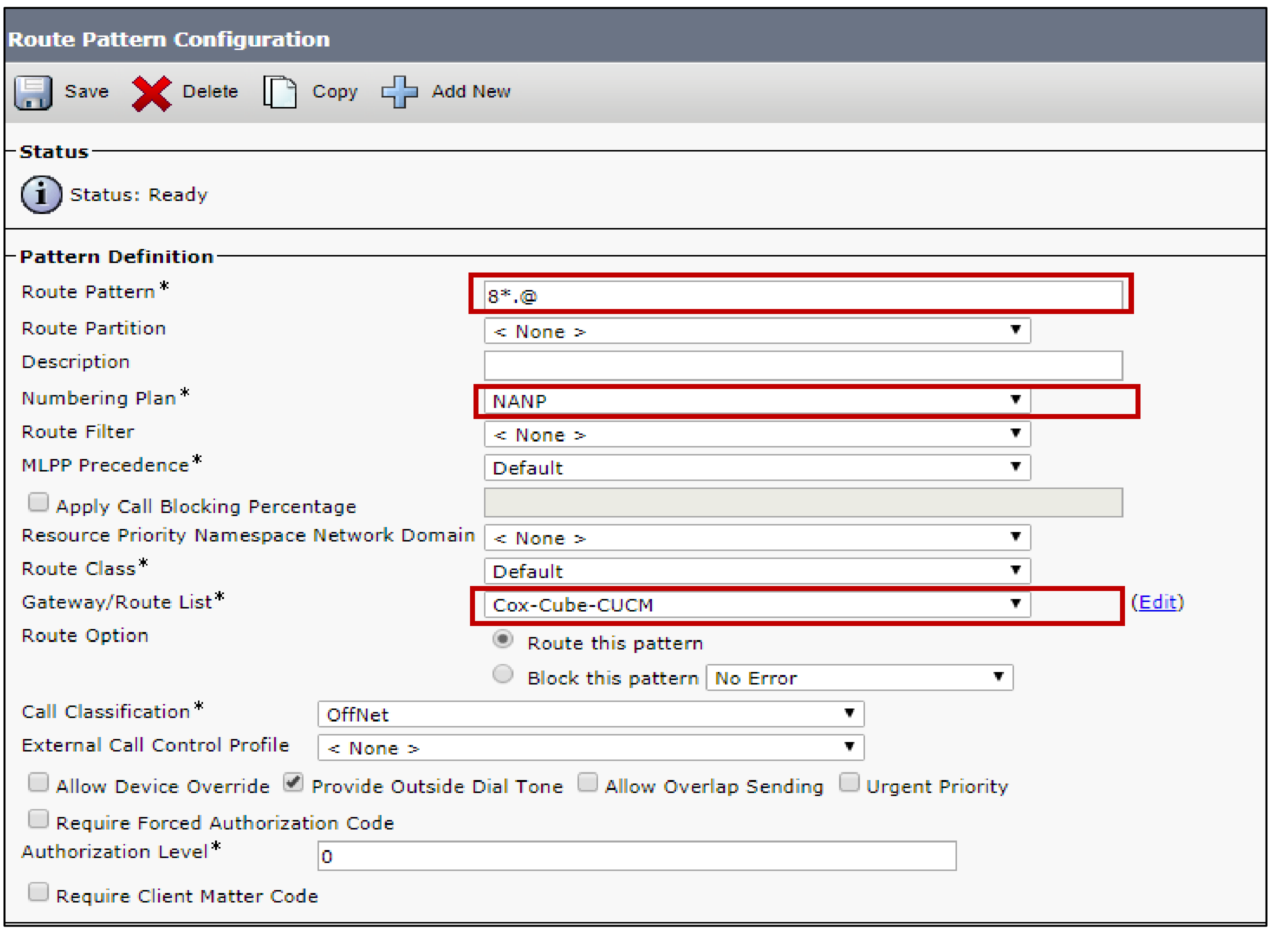


Figure 25 Route Pattern Configuration for Cisco UBE-PSTN Access National- Anonymous Call

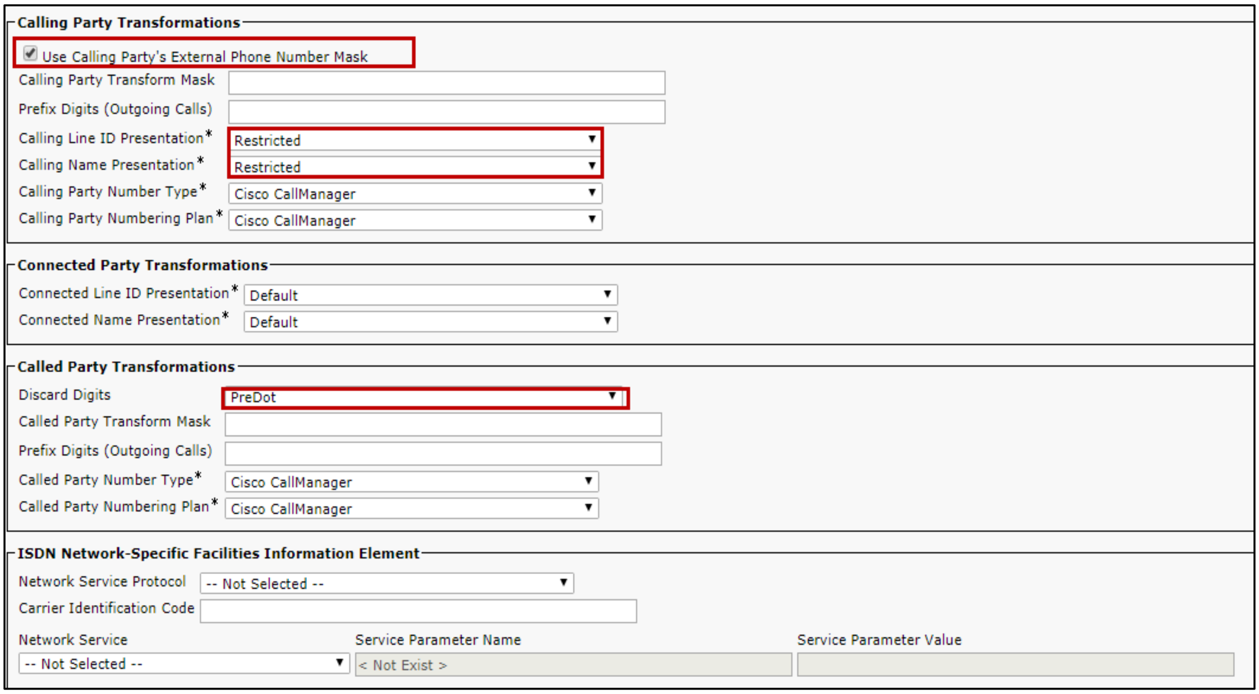


Figure 26: Route Pattern Configuration for Cisco UBE-PSTN Access National- Anonymous Call (Cont.)

Route Pattern for Fax Gateway:

- **Navigation**: Call Routing → Route/Hunt → Route Pattern → Add New

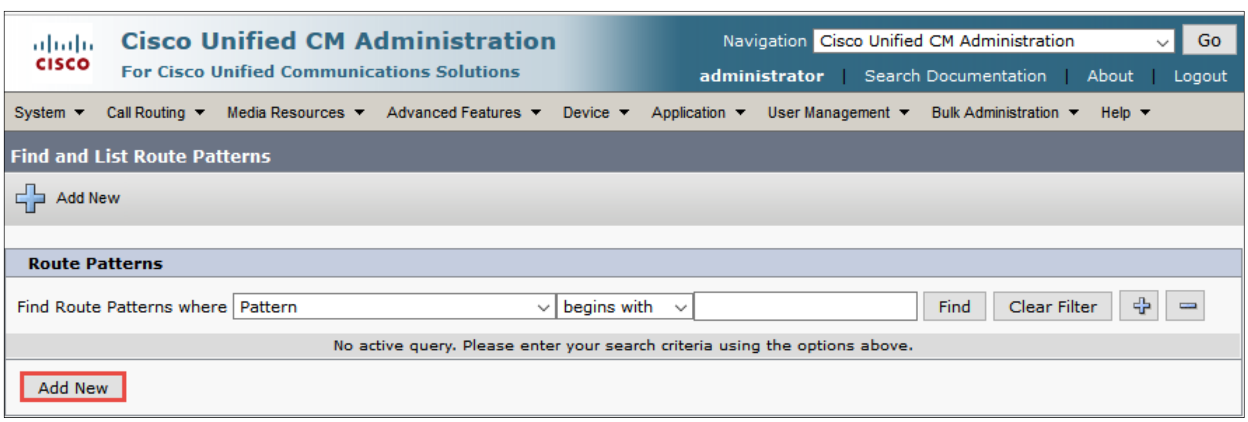


Figure 27: Add New Route patter to Fax Gateway

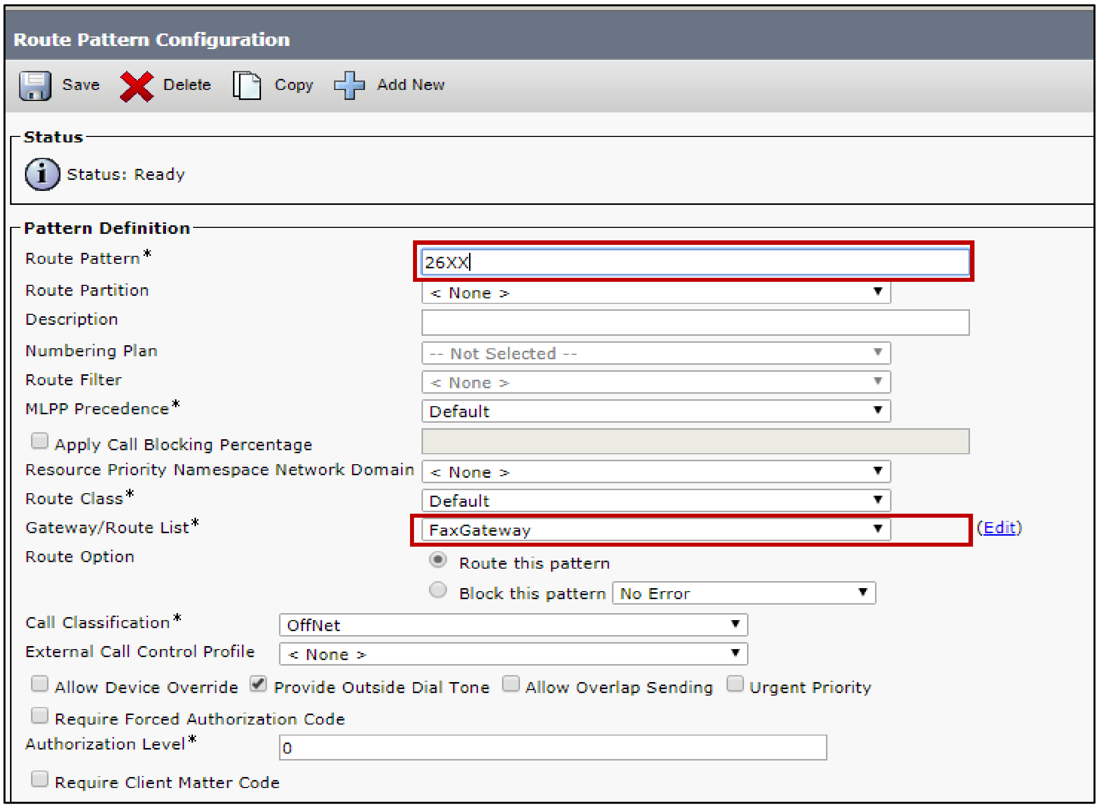


Figure 28: Route Pattern Configuration for Fax Gateway

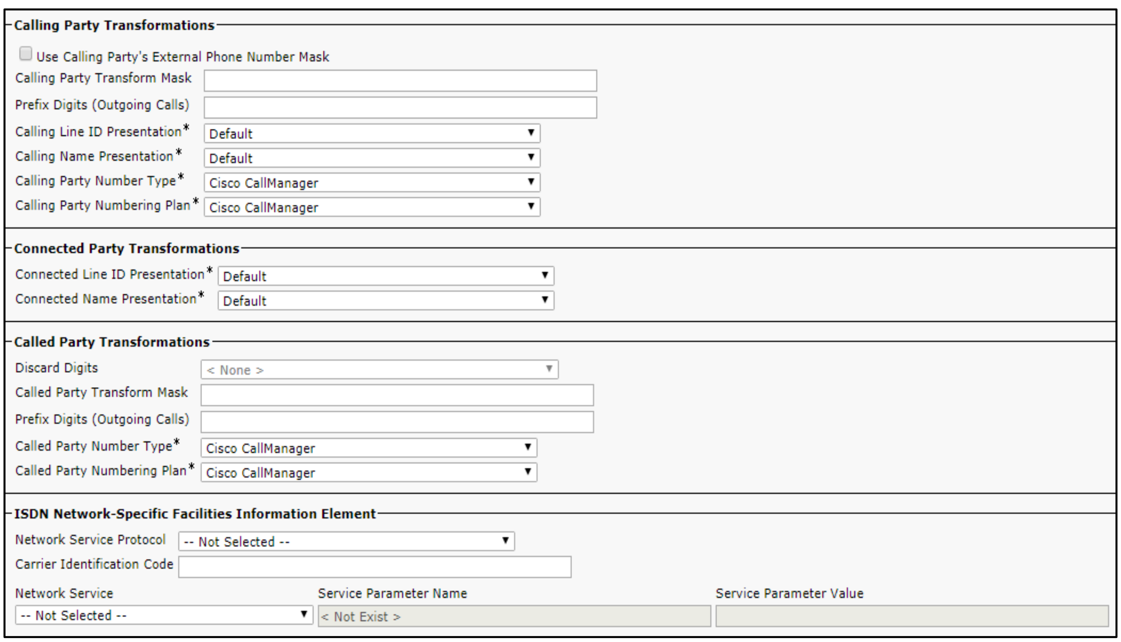


Figure 29: Route Pattern Configuration for Fax Gateway (Cont.)

Configuring Cisco Voice Gateway for Fax

The following configuration snippet contains a sample configuration of Cisco Voice gateway.

Global Settings

```
voice service voip
 allow-connections sip to sip
 no supplementary-service sip handle-replaces
 redirect ip2ip
 fax protocol pass-through g711ulaw
 no fax-relay sg3-to-g3
 sip
  rel1xx disable
  midcall-signaling passthru
  g729 annexb-all
!
```

Codecs

G711ulaw and G729 voice codecs are used for this testing. Codec preferences used to change according to the test plan description.

```
voice class codec 3
 codec preference 1 g711ulaw
!
voice class codec 2
 codec preference 1 g711alaw
 codec preference 2 g729br8
 codec preference 3 g729r8
!
voice class codec 1
 codec preference 1 g729br8
 codec preference 2 g729r8
 codec preference 3 g711ulaw!
```

Dial peer

Outbound Dial-peer to Cisco UCM:

```
dial-peer voice 11 voip
 description Gateway to CUCM for COXCUBECUCM
 service session
 destination-pattern 9T
 session protocol sipv2
 session target ipv4:10.80.10.2
 session transport udp
 voice-class codec 3
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
```

Inbound Dial-peer from Cisco UCM:

```
dial-peer voice 12 voip
 description COXCUBECUCM Inbound Fax
 service session
 session protocol sipv2
 session transport udp
 incoming called-number 4962
 voice-class codec 3
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax protocol pass-through g711ulaw
 no vad
```

POTS and Port Configuration:

Based on configured POTS destination pattern, gateway forwards the call to designated voice port.

```
dial-peer voice 10 pots
 service session
 destination-pattern 4029324962
 no digit-strip
 port 0/0/1
 forward-digits all

voice-port 0/0/1
 no echo-cancel enable
 no vad
 cptone IN
 station-id name CoxFAXDIDname
 station-id number 4029324962
 caller-id enable
!
```

Configuration example

The following configuration snippet contains a sample configuration of Cisco Voice Gateway with all parameters mentioned previously.

```
cme.in.tekvizion.com#sh run


version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname cme.in.tekvizion.com
!
boot-start-marker
boot-end-marker
!
aaa new-model
!
aaa authentication login local_auth local
!
aaa session-id common
clock timezone IST 5 30
network-clock-participate wic 2
network-clock-participate wic 3
!
dot11 syslog
ip source-route
!
ip cef
!
isdn switch-type primary-qsig
!
voice rtp send-recv
```

```
!
voice service pots
!
voice service voip
 allow-connections sip to sip
 no supplementary-service sip handle-replaces
 redirect ip2ip
 no fax-relay sg3-to-g3
 sip
  rel1xx disable
  fax protocol pass-through g711ulaw
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 3
 codec preference 1 g711ulaw
!
voice class codec 2
 codec preference 1 g711alaw
 codec preference 2 g729br8
 codec preference 3 g729r8
!
voice class codec 1
 codec preference 1 g729br8
 codec preference 2 g729r8
 codec preference 3 g711ulaw
!
voice class codec 4
 codec preference 1 g711alaw
 codec preference 2 g722-64
!
voice class codec 5
 codec preference 1 g729r8
```

```
!
voice-card 0
 dspfarm
 dsp services dspfarm
!
crypto pki token default removal timeout 0
!
license udi pid CISCO2851 sn FHK1137F4LY
!
interface GigabitEthernet0/0
 ip address 172.16.31.50 255.255.255.0
 duplex auto
 speed auto
!
ip forward-protocol nd
!
ip http server
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 172.16.31.1
ip route 0.0.0.0 0.0.0.0 172.16.29.1
!
ipv6 route ::/0 2620:96:C000:8::1
!
snmp-server community public RO
snmp-server location Chennai
!
ipv6 access-list ipv6
 permit ipv6 any any
!
control-plane
!
voice-port 0/0/0
```

```
 no echo-cancel enable
 no vad
 cptone IN
 station-id name COX user2
 station-id number 26XX
 caller-id enable
!
voice-port 0/0/1
 no echo-cancel enable
 no vad
 cptone IN
 station-id name CoxFAXDIDname
 station-id number 402XXXXXXX
 caller-id enable
!
voice-port 0/3/0:15
!
voice-port 0/2/0:15
!
voice-port 0/1/0
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 172.16.26.2
ccm-manager config
!
mgcp
mgcp call-agent 172.16.26.2 2427 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
```

```
mgcp package-capability pre-package
mgcp default-package mt-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
sccp local GigabitEthernet0/0
sccp ccm 10.71.3.10 identifier 1 version 7.0
!
sccp ccm group 6
 bind interface GigabitEthernet0/0
 associate ccm 1 priority 1
!
dial-peer voice 10 pots
 service session
 destination-pattern 402XXXXXXX
 no digit-strip
 port 0/0/1
 forward-digits all
!
dial-peer voice 11 voip
 description Gateway to CUCM for COXCUBECUCM
 service session
 destination-pattern 1T
 session protocol sipv2
 session target ipv4:10.80.10.2
 session transport udp
 voice-class codec 3
 dtmf-relay rtp-nte
```

```
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 12 voip
 description COXCUBECUCM Inbound Fax
 service session
 session protocol sipv2
 session transport udp
 incoming called-number 49XX
 voice-class codec 3
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 20 pots
 service session
 destination-pattern 26XX
 no digit-strip
 port 0/0/0
 forward-digits all
!
sip-ua
 no remote-party-id
 retry register 5
 no timers hold
 protocol mode dual-stack
!
```

Configuring Cisco Unity Connection

Cisco Unity Connection Version

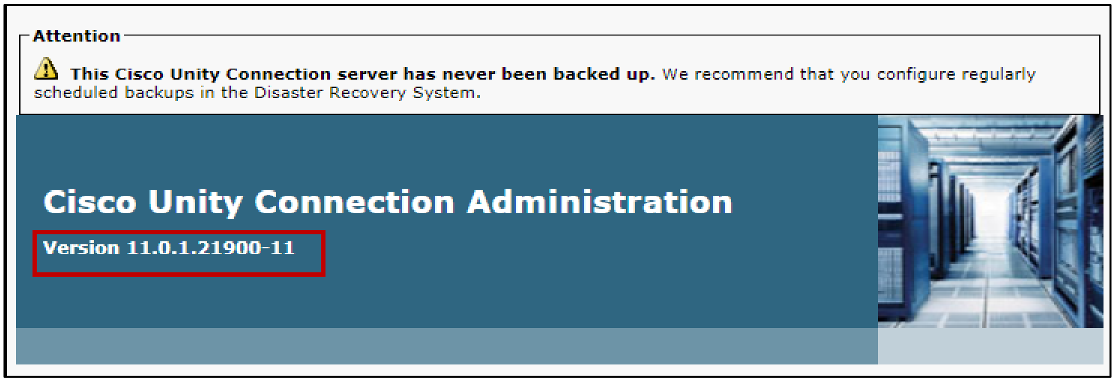


Figure 30: Version information

User Configuration

Navigation: Cisco Unity Connection → Users → Users

- Set Alias= **cox-cube-cucm** is used for this example
- Set First Name = **cox** is used to identify this User
- Set Extension = **4563** is used in this example
- Set Phone System = **cox-cube-cucm** is used in this example
- All other values are default

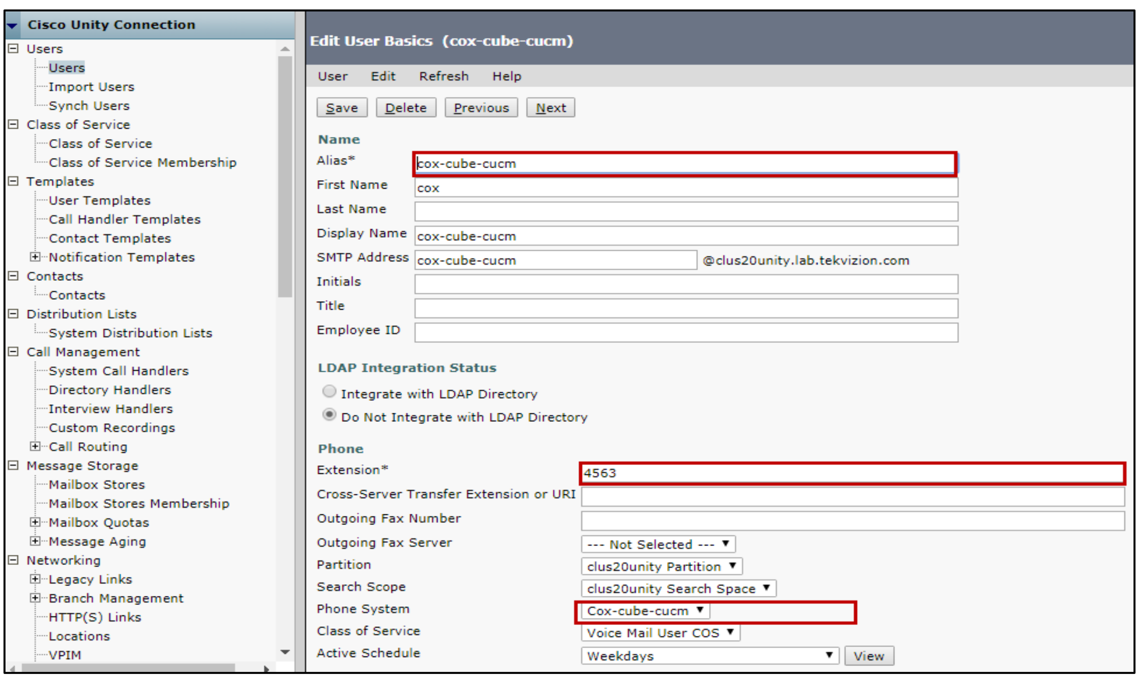


Figure 31: User Configuration

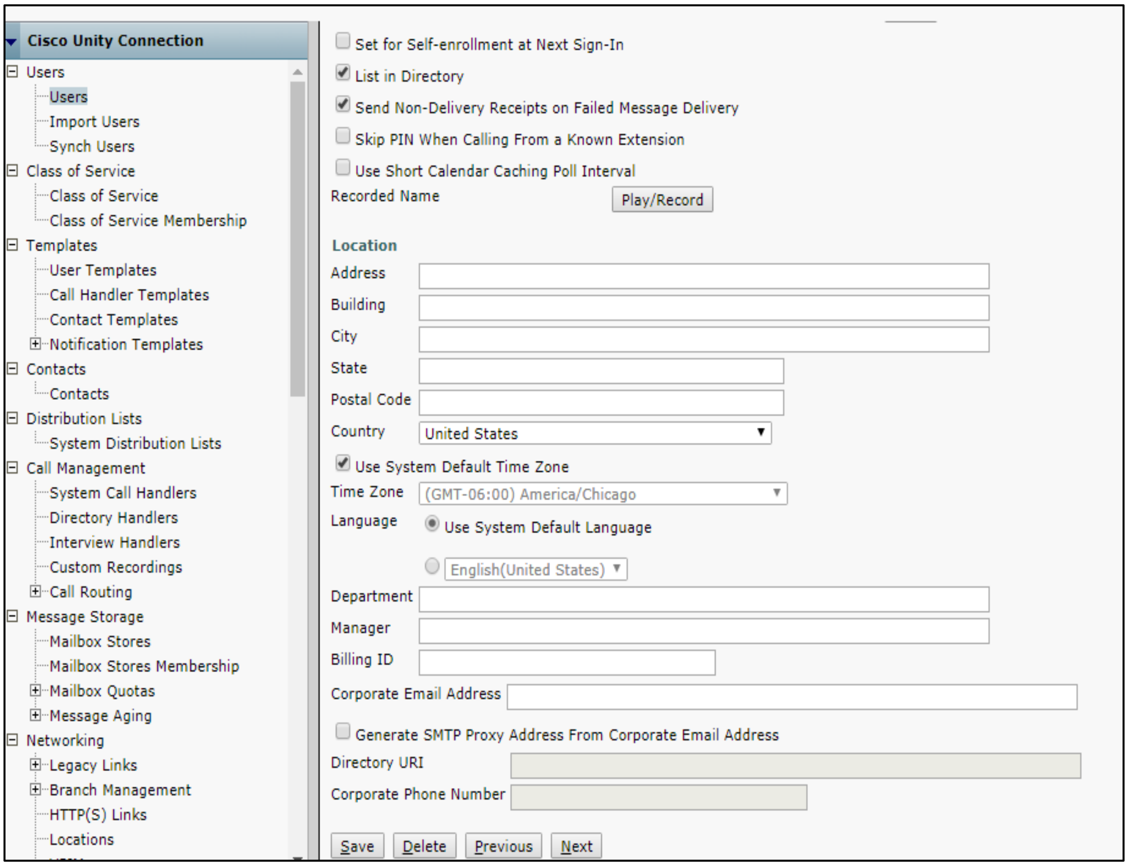


Figure 32: User Configuration (cont.)

Telephony Integration

Navigation: Telephony Integrations → Phone system

- Set Phone System Name = **Cox-cube-cucm** is used for this example

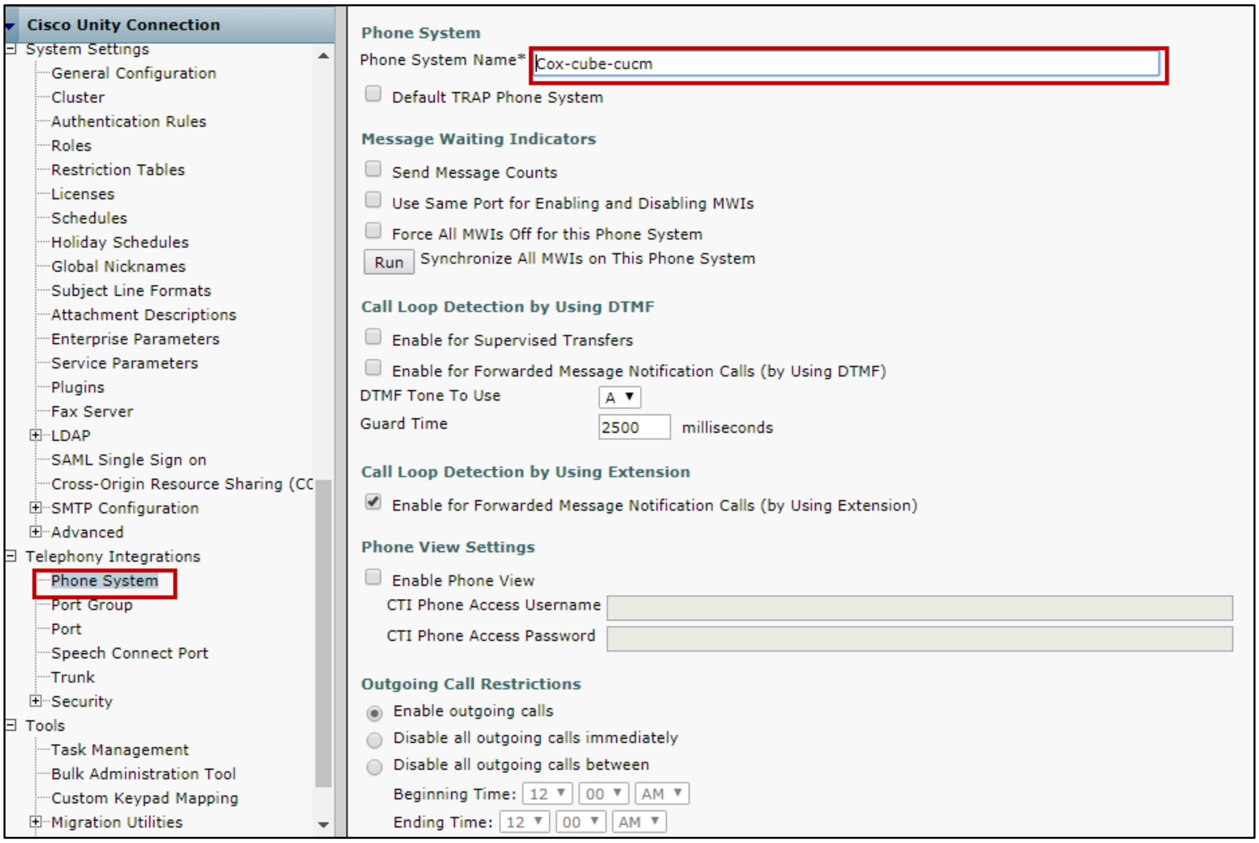


Figure 33: Telephone Integration – Phone System

Port Group

Navigation: Telephony Integration → Port Group

- Set Display Name = **COX-cube-cucm-**1 is used for this example

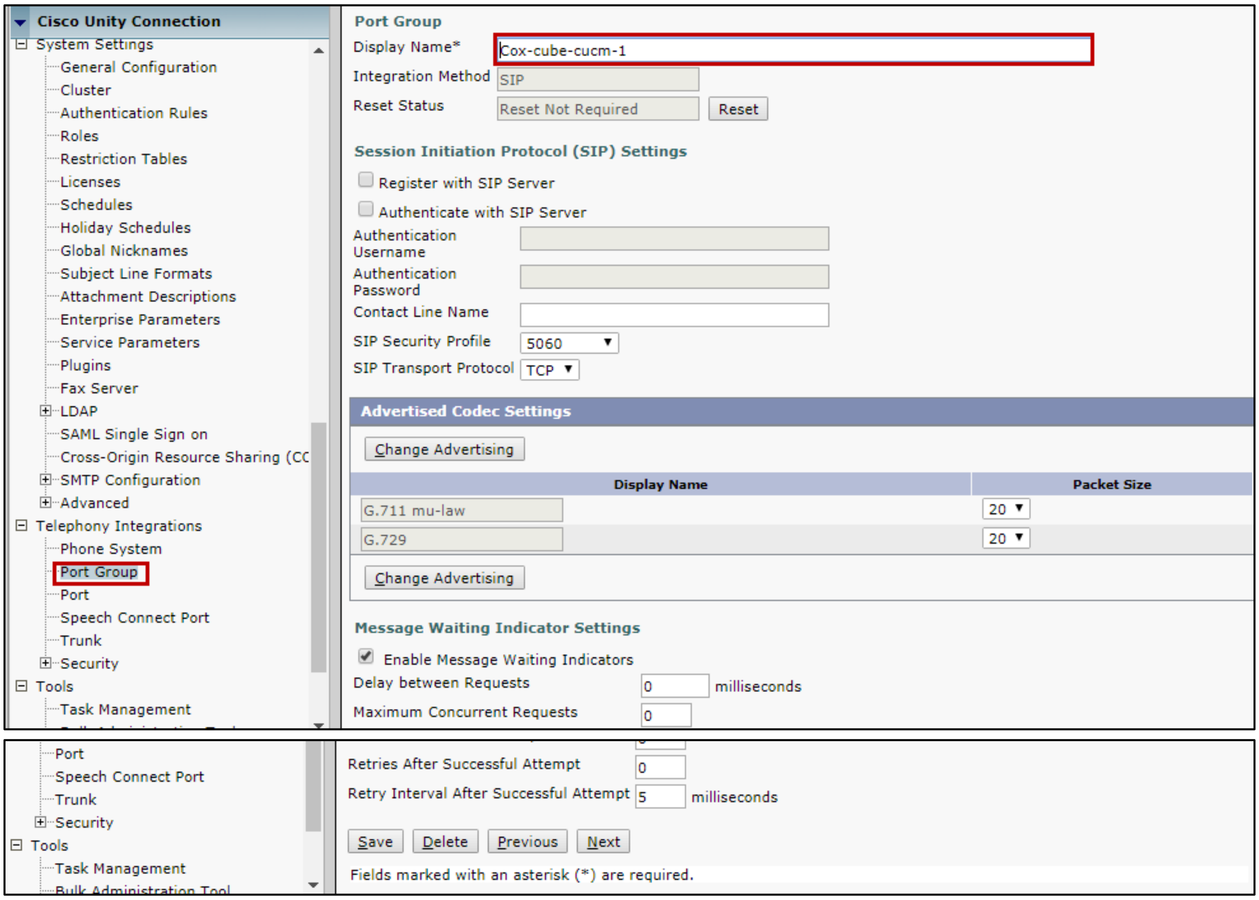


Figure 34: Port Group

Navigation: Telephony Integration→ Port Group → Edit → Servers

- Set IPV4 Address = **10.80.10.2 (Cisco UCM IP is used for this example)**

Figure 35: Port Group (cont.)

Port

Navigation: Telephony Integration → Port

- Set Port Name = **Cox-cube-cucm-1-001** is used for this example
- Set Phone System = **Cox-cube-cucm**
- Set Port Group = **Cox-cube-cucm-1**
- Set Server = **clus20unity.skypelabsj.local** is used for this example

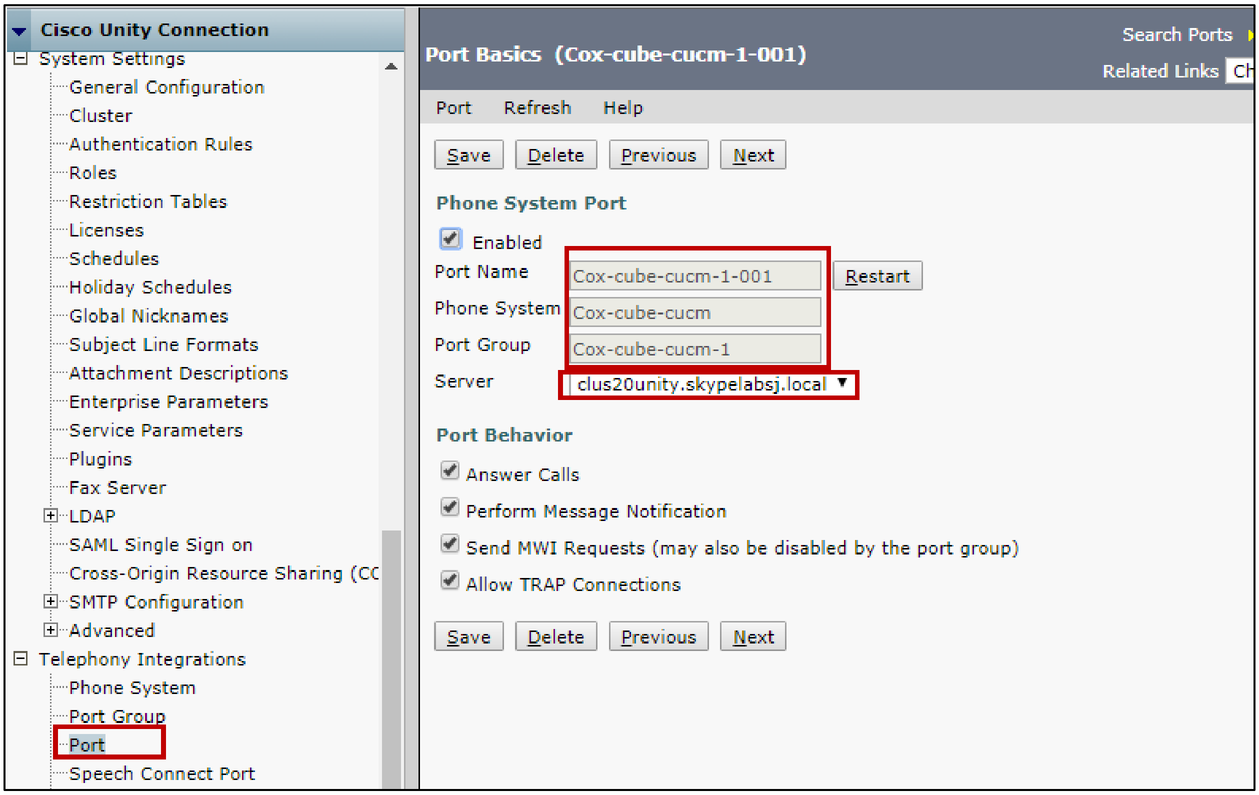


Figure 36: Port

Acronyms

Acronym	Definitions
CPE	Customer Premise Equipment
Cisco UBE	Cisco Unified Border Element
Cisco UCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
ESBC	Enterprise Session Border Controller
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol

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