Application Note

Connecting Cisco Unified Communication Manager [v11.5.1] to British Telecom Global SIP Trunking Service via Cisco Unified Border Element v12.0 [IOS-XE 16.06.01]

December 7, 2017

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Introduction

Service Providers today, such as British Telecom, 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.

British Telecom 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 British Telecom network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS-XE 16.6.1 can be used. The Cisco Unified Border Element 12.0.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to British Telecom IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco Unified Communications Manager. Only configuration settings specifically required for British Telecom 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 11.5.1, and Cisco UBE on ISR 4321/K9 [IOS-XE 16.6.1] for connectivity to British Telecom SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (British Telecom) via Cisco UBE v12.0 [IOS-XE 16.6.1]
- Testing was performed in accordance to British Telecom 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 British Telecom 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 British Telecom SIP Trunking network.

Network Topology

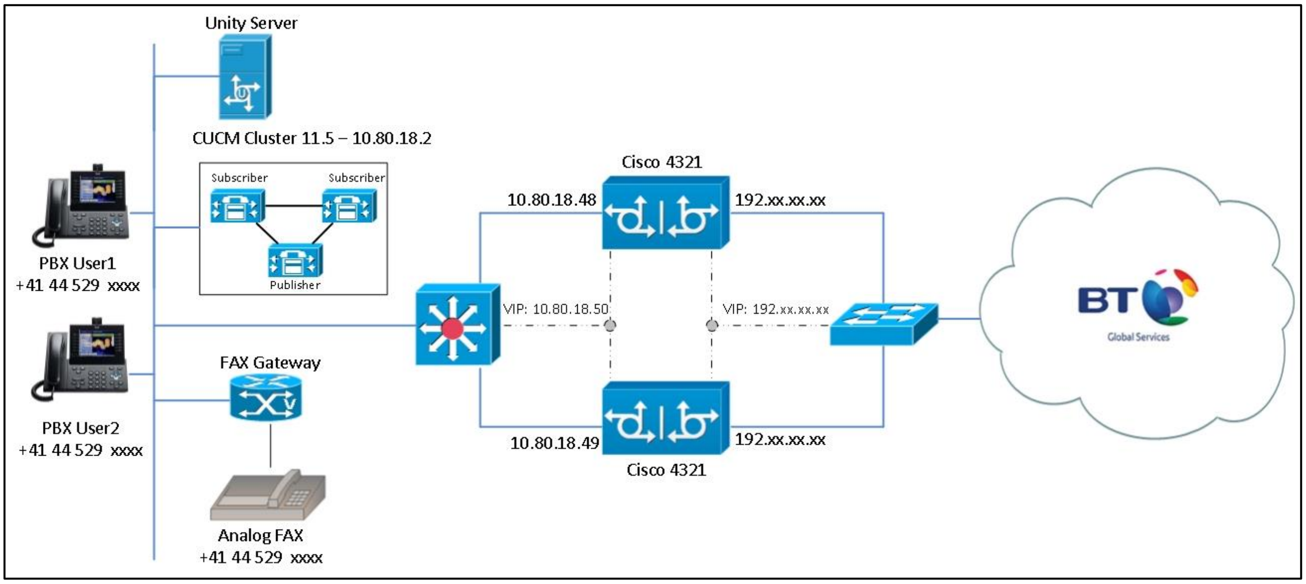


Figure 1 Network Topology

- 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. British Telecom 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 British Telecom 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 British Telecom	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 4321 router
- CUCM cluster on UCS C240, 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 11.5.1.12900-21 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
- British Telecom does not support Fax at Super G3 Speed

Caveats

- Caller ID is not 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.
- British Telecom does not support SG3 Fax. Fax test cases are executed only on G3. Maximum bit rate supported by British Telecom for T38 fax is 9600bps.
- Loopback fax over G711 Pass-Through was unsuccessful. Re-Invite for pass-through from the originating fax gateway is dropped in the BT network.
- British Telecom supports only E164 format for International calls.

Configuration

Configuring 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.

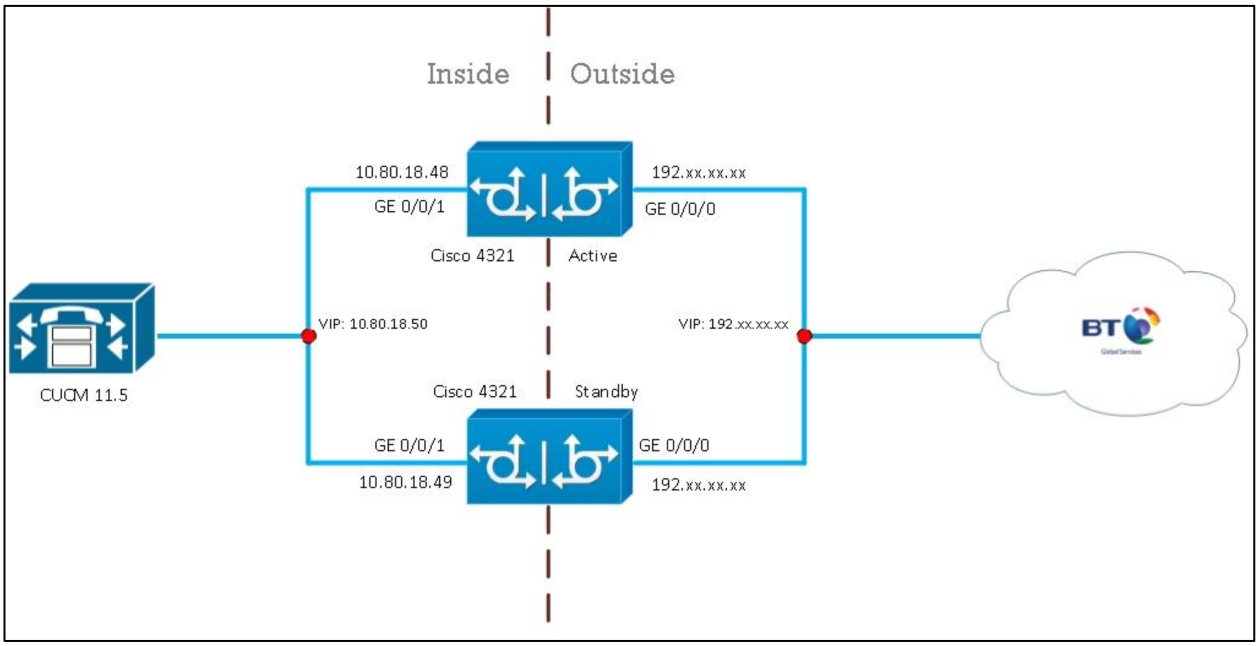


Figure 2 High Availability topology

Cisco UBE 1:

```
interface GigabitEthernet0/0/0
 description BT WAN Interface
 ip address 192.XX.XX.XX 255.255.255.128
 media-type rj45
 negotiation auto
 redundancy rii 15
 redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
 description BT LAN Interface
 ip address 10.80.18.48 255.255.255.0
 negotiation auto
 redundancy rii 16
 redundancy group 2 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
 description BT CubeHA Interface
 ip address 10.70.50.100 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
```

Cisco UBE 2:

```
interface GigabitEthernet0/0/0
 description BT WAN Interface
 ip address 192.XX.XX.XX 255.255.255.128
 media-type rj45
 negotiation auto
 redundancy rii 15
 redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
 description BT LAN Interface
 ip address 10.80.18.49 255.255.255.0
 negotiation auto
 redundancy rii 16
 redundancy group 2 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
 description BT CubeHA Interface
 ip address 10.70.50.110 255.255.255.0
 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 2
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  session refresh
  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 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
!
voice class codec 2
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 g711alaw
!
```

Dial peer

Outbound Dial-peer to British Telecom:

```
dial-peer voice 500 voip
 description Outgoing Call to BT - LAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number .T
 voice-class codec 1
 voice-class sip asserted-id pai
 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
 fax-relay ecm disable
 fax rate 14400
```

```
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 510 voip
 description Outgoing call to BT - WAN facing
 translation-profile outgoing E164dialing
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 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
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
```

Inbound Dial-peer from British Telecom:

```
dial-peer voice 600 voip
 description Incoming call to IP-PBX - WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number +4144529....
 incoming uri via PSTN
 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
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 610 voip
 description Incoming call to PBX - LAN facing
 huntstop
 destination-pattern +4144529....
 session protocol sipv2
 session target ipv4:10.80.18.2
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 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
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
```

Configuration example

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

Active Cisco UBE:

```
Current configuration : 6222 bytes
!
! Last configuration change at 09:44:23 UTC Mon Nov 6 2017
!
version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname BTCube1
!
boot-start-marker
boot system flash isr4300-universalk9.16.06.01.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
```

```
enable secret 5 xxxxxxx
!
no aaa new-model
!
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 2
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
voice class uri PSTN sip
 host ipv4:213.XX.XX.XX
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
```

```
!
voice class codec 2
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 g711alaw
!
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>"
"<sip:+4144529\1@\2>"
!
voice translation-rule 1
 rule 1 /\(^...........$\)/ /+\1/
 rule 2 /\(^............$\)/ /+\1/
!
voice translation-profile E164dialing
 translate called 1
!

license udi pid ISR4321/K9 sn FDO19220MQ8
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco password 0 ******
!

redundancy
 mode none
 application redundancy
  group 2
   name b2bhaBT
```

```
  priority 100 failover threshold 75
  timers delay 30 reload 60
  control GigabitEthernet0/1/0 protocol 1
  data GigabitEthernet0/1/0
  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 BT WAN Interface
 ip address 192.XX.XX.XX 255.255.255.128
 media-type rj45
 negotiation auto
 redundancy rii 15
 redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
 description BT LAN Interface
 ip address 10.80.18.48 255.255.255.0
 negotiation auto
 redundancy rii 16
 redundancy group 2 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
 description BT CubeHA Interface
```

```
 ip address 10.70.50.100 255.255.255.0
 negotiation auto
!
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.XX.XX.XX
ip route 10.70.50.0 255.255.255.0 10.80.18.1
ip route 172.16.0.0 255.255.0.0 10.80.18.1
!
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 500 voip
 description Outgoing Call to BT - LAN facing
```

```
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number .T
 voice-class codec 1
 voice-class sip asserted-id pai
 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
 fax-relay ecm disable
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 510 voip
 description Outgoing call to BT - WAN facing
 translation-profile outgoing E164dialing
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 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
```

```
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 600 voip
 description Incoming call to IP-PBX - WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number +4144529....
 incoming uri via PSTN
 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
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 610 voip
 description Incoming call to PBX - LAN facing
 huntstop
 destination-pattern +4144529....
```

```
 session protocol sipv2
 session target ipv4:10.80.18.2
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 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
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 710 voip
 description IP-PBX to IP-PBX loopbackcall
 translation-profile outgoing E164dialing
 huntstop
 destination-pattern 4144529....
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 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
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
!



sip-ua
 keepalive target ipv4:213.XX.XX.XX:5060
 timers keepalive active 180
 sip-server ipv4:213.XX.XX.XX
!
line con 0
 exec-timeout 0 0
 login local
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 login local
 transport input telnet
!
end
```

Standby Cisco UBE:

```
Current configuration : 6580 bytes
!
! Last configuration change at 09:53:15 UTC Mon Nov 6 2017
!
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 BTCube2
!
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 $1$aEKr$TYXs2kVsHhvpeqnj4bdYn.
!
no aaa new-model
```

```
!
subscriber templating
!
!
multilink bundle-name authenticated
!
!
crypto pki trustpoint TP-self-signed-2548443246
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2548443246
 revocation-check none
 rsakeypair TP-self-signed-2548443246
!
!
crypto pki certificate chain TP-self-signed-2548443246
!
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 2
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
```

```
  midcall-signaling passthru
  g729 annexb-all
!
voice class uri PSTN sip
 host ipv4:213.XX.XX.XX
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
!
voice class codec 2
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 g711alaw
!
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:+4144529\1@\2>"
!
voice translation-rule 1
 rule 1 /\(^..........$\)/ /+\1/
 rule 2 /\(^...........$\)/ /+\1/
!
!
voice translation-profile E164dialing
 translate called 1
!
!
license udi pid ISR4321/K9 sn FDO19220MW3
license boot level appxk9
```

```
license boot level uck9
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 131112193D5D1E7B7B2A
!
redundancy
 mode none
 application redundancy
  group 2
   name b2bhaBT
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/1/0 protocol 1
   data GigabitEthernet0/1/0
   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 BT WAN Interface
 ip address 192.XX.XX.XX 255.255.255.128
 media-type rj45
 negotiation auto
```

```
 redundancy rii 15
 redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
 description BT LAN Interface
 ip address 10.80.18.49 255.255.255.0
 negotiation auto
 redundancy rii 16
 redundancy group 2 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
 description BT CubeHA Interface
 ip address 10.70.50.110 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.XX.XX.XX
ip route 10.70.50.0 255.255.255.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
```

```
!
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 500 voip
 description Outgoing Call to BT - LAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number .T
 voice-class codec 1
 voice-class sip asserted-id pai
 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
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
```

```
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 510 voip
 description Outgoing call to BT - WAN facing
 translation-profile outgoing E164dialing
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 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
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 600 voip
 description Incoming call to IP-PBX - WAN facing
 huntstop
 session protocol sipv2
 session transport udp
```

```
 incoming called-number +4144529....
 incoming uri via PSTN
 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
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 610 voip
 description Incoming call to PBX - LAN facing
 huntstop
 destination-pattern +4144529....
 session protocol sipv2
 session target ipv4:10.80.18.2
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 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
 fax-relay ecm disable
```

```
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 710 voip
 description IP-PBX to IP-PBX loopbackcall
 translation-profile outgoing E164dialing
 huntstop
 destination-pattern 4144529....
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 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
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
sip-ua
 keepalive target ipv4:213.XX.XX.XX:5060
 timers keepalive active 180
```

```
 sip-server ipv4:213.XX.XX.XX
!
!
line con 0
 exec-timeout 0 0
 login local
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 login local
 transport input telnet
!
End
```

Configuring Cisco UCM 11.5 Cluster

Cisco UCM Version

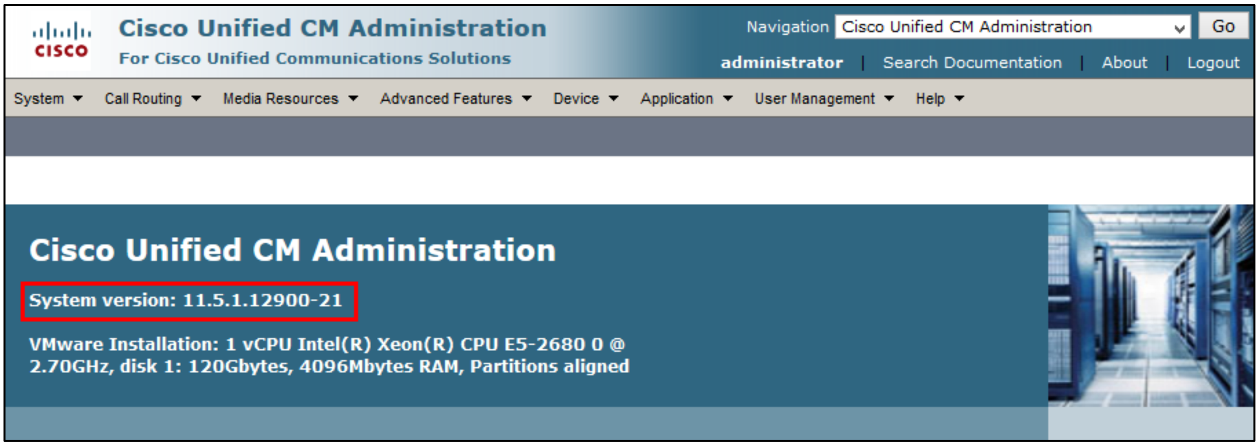


Figure 3: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

- Select Server* = Clus28pub--CUCM Voice/Video (Active)
- Select Service*= Cisco CallManager (Active)
- Duplex Streaming Enabled* = True
- All other fields are set to default values

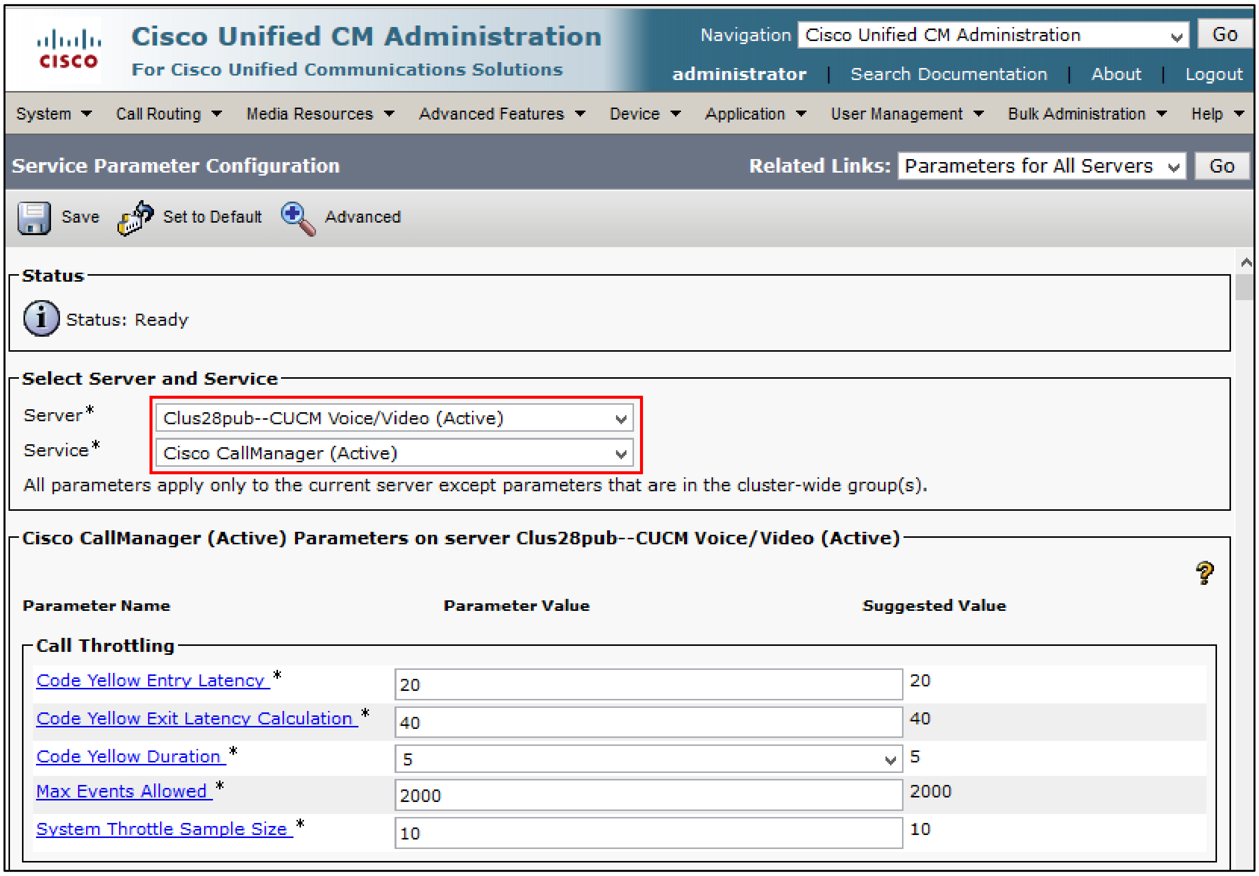


Figure 4: Service Parameters

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 for BT** 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**

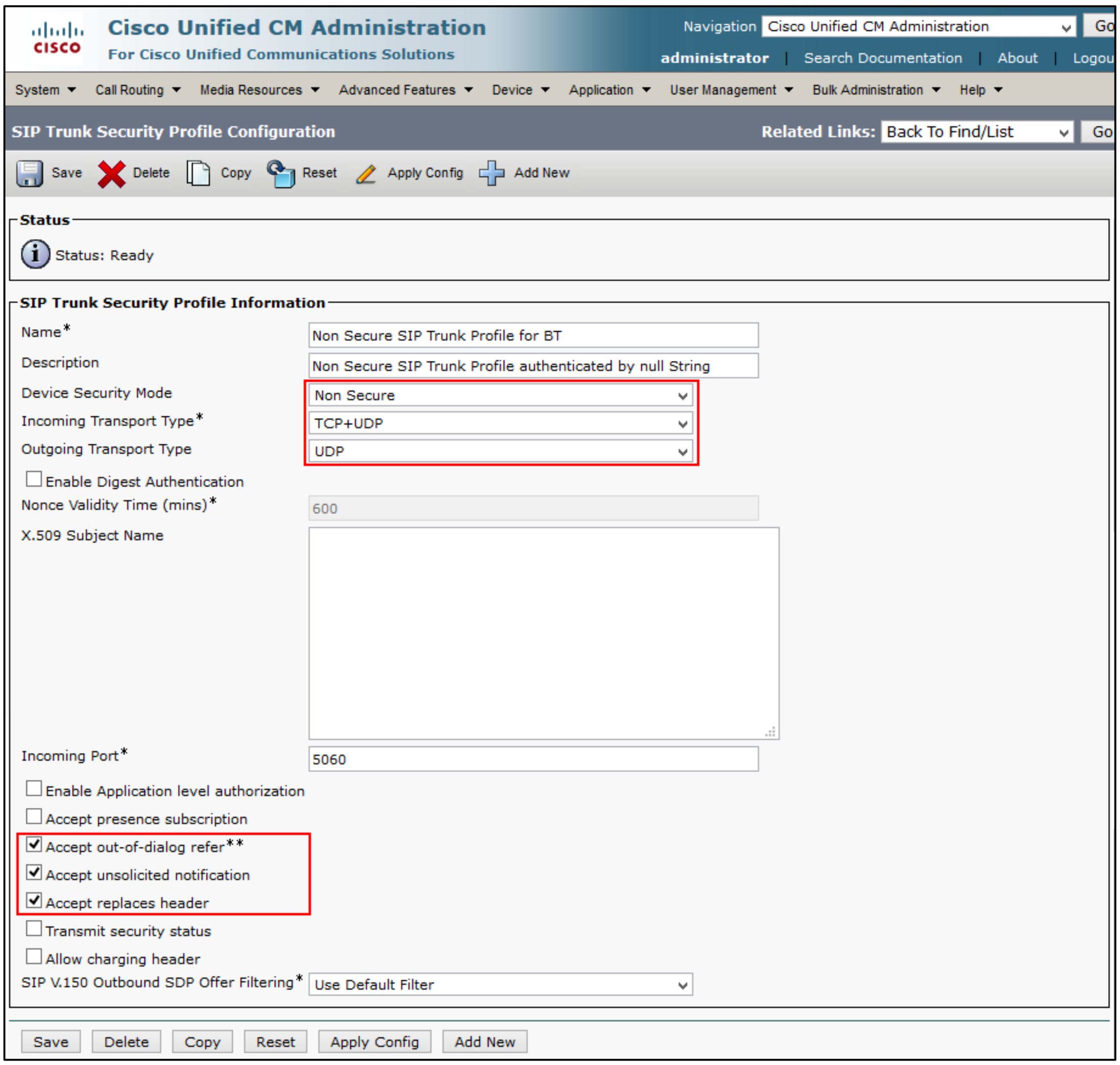


Figure 6: SIP Trunk Security Profile

SIP Profile

Navigation: Device → Device Settings → SIP Profile

- Name*= **Standard SIP Profile for BT** is used as an example
- Description = **Standard SIP Profile for BT** is used as an example

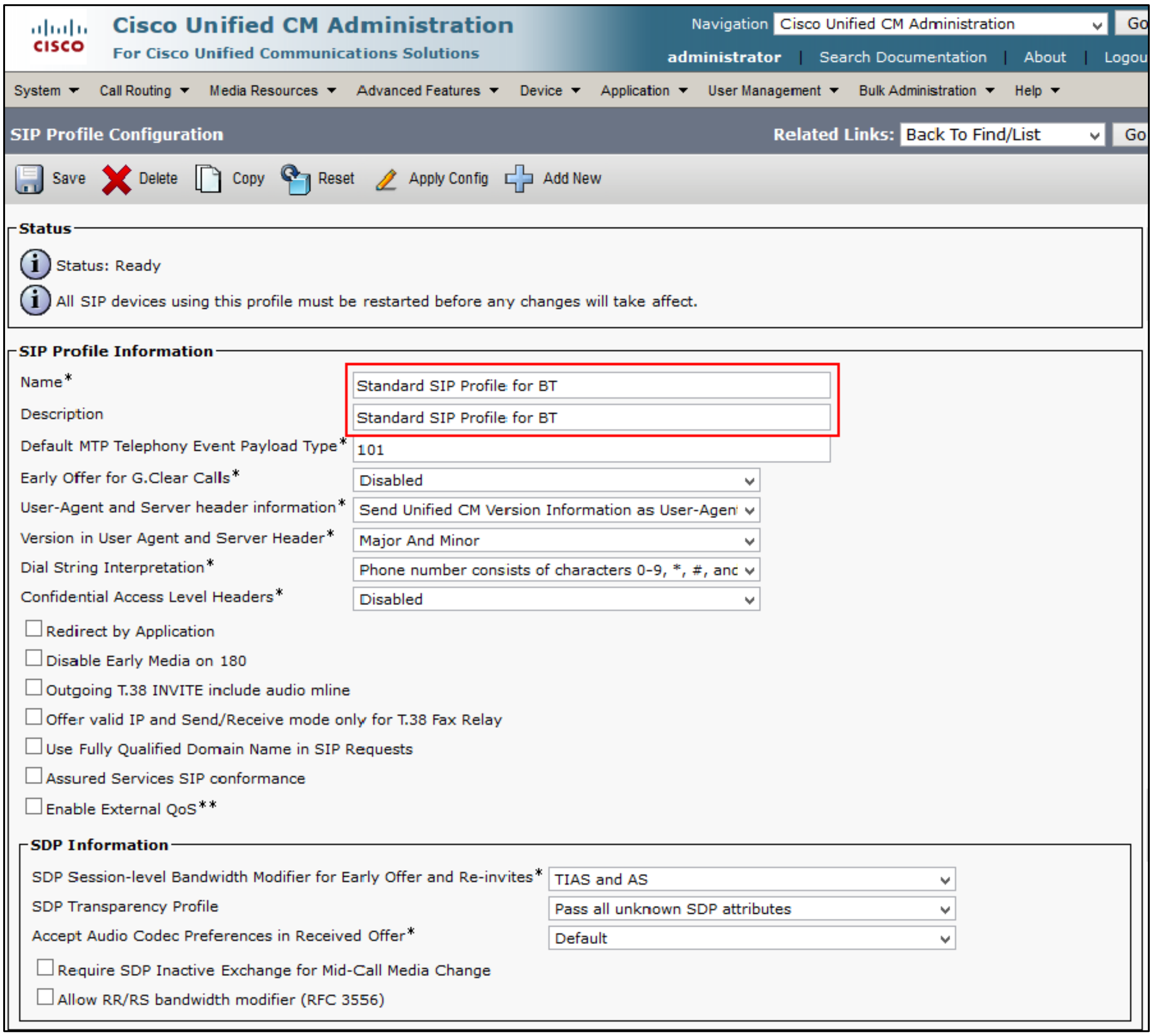
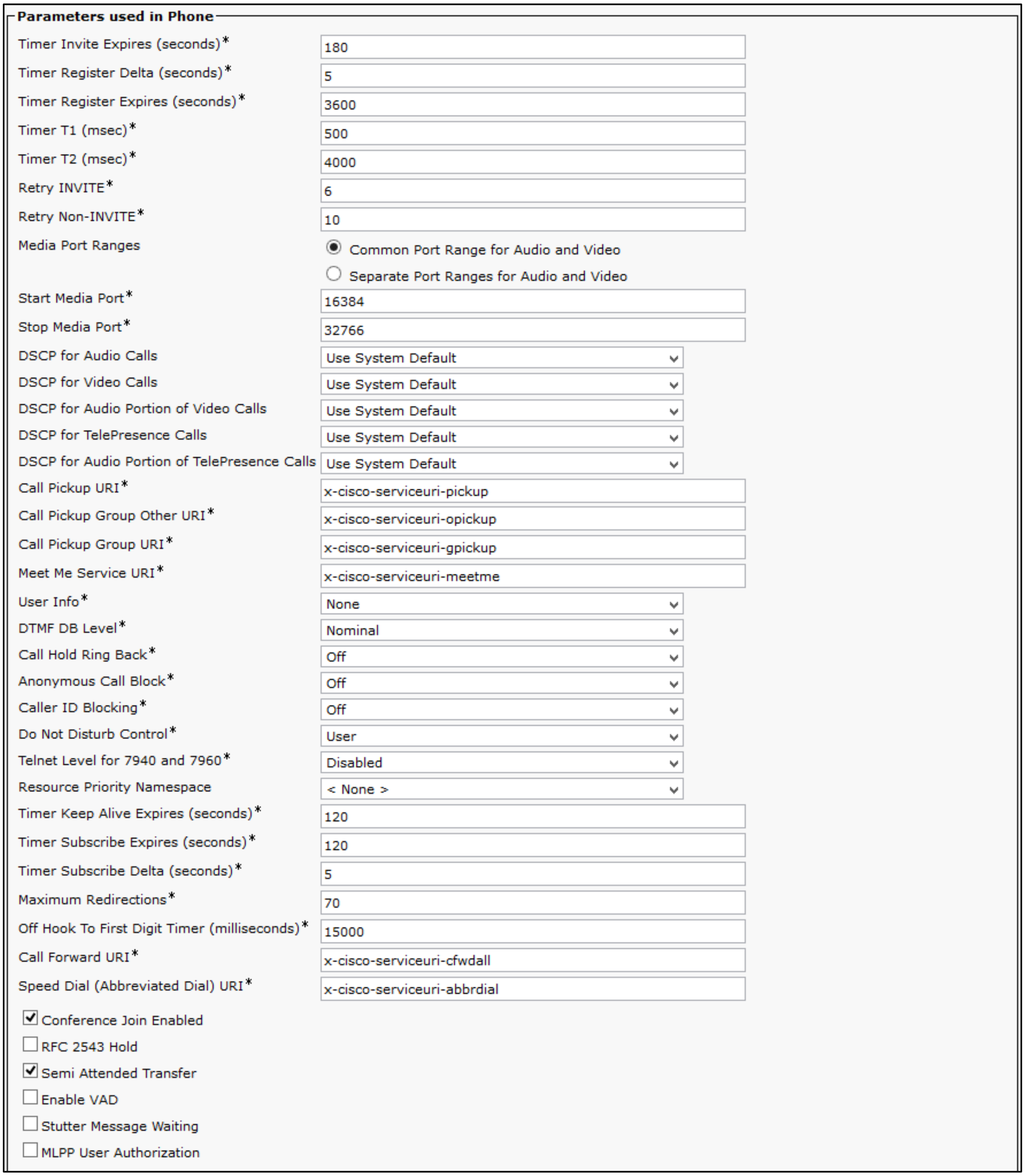


Figure 7: SIP Profile



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.)

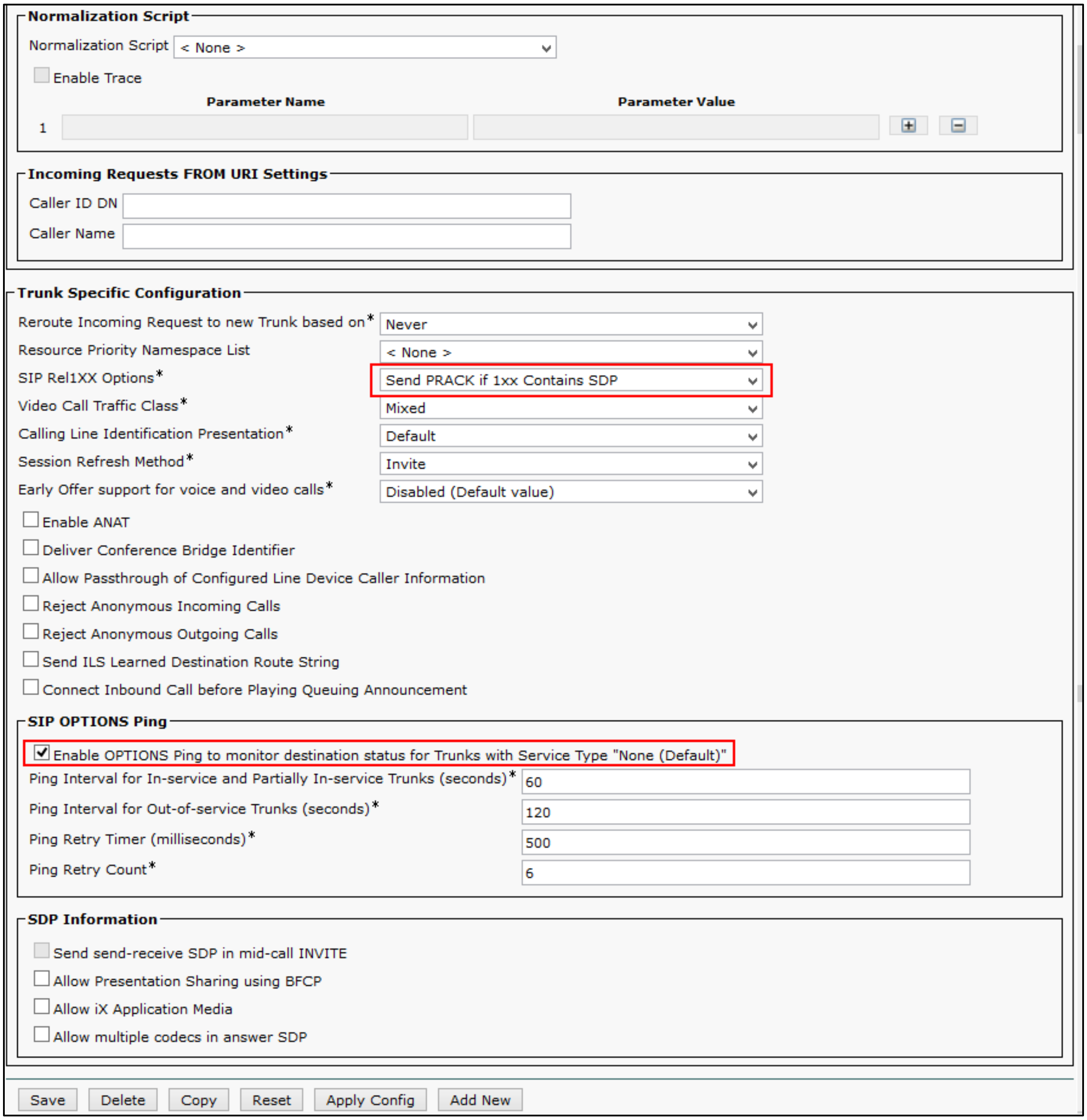


Figure 9: SIP Profile (Cont.)

Trunk configuration

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

Navigation: Device → Trunk → Add New

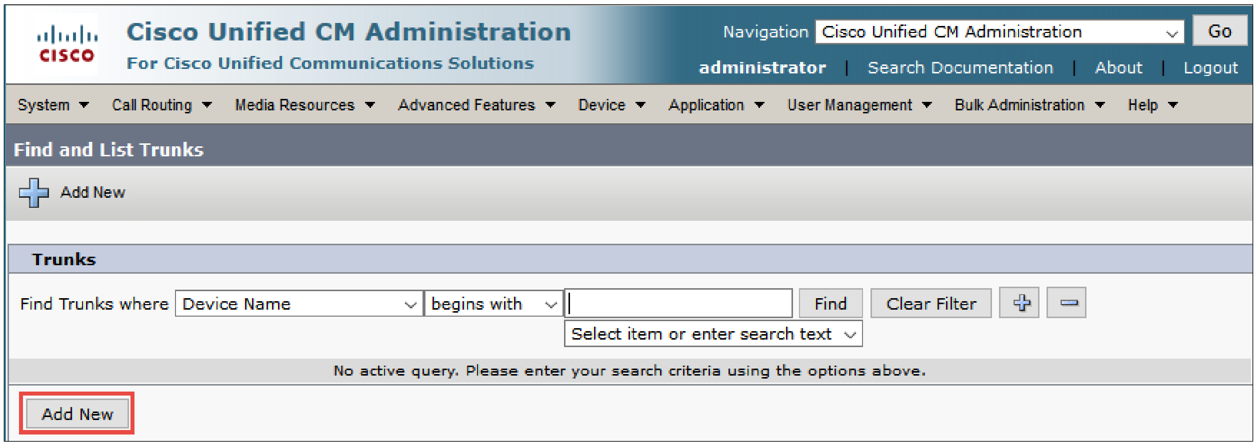


Figure 10: Add New Trunk to Cisco UBE

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

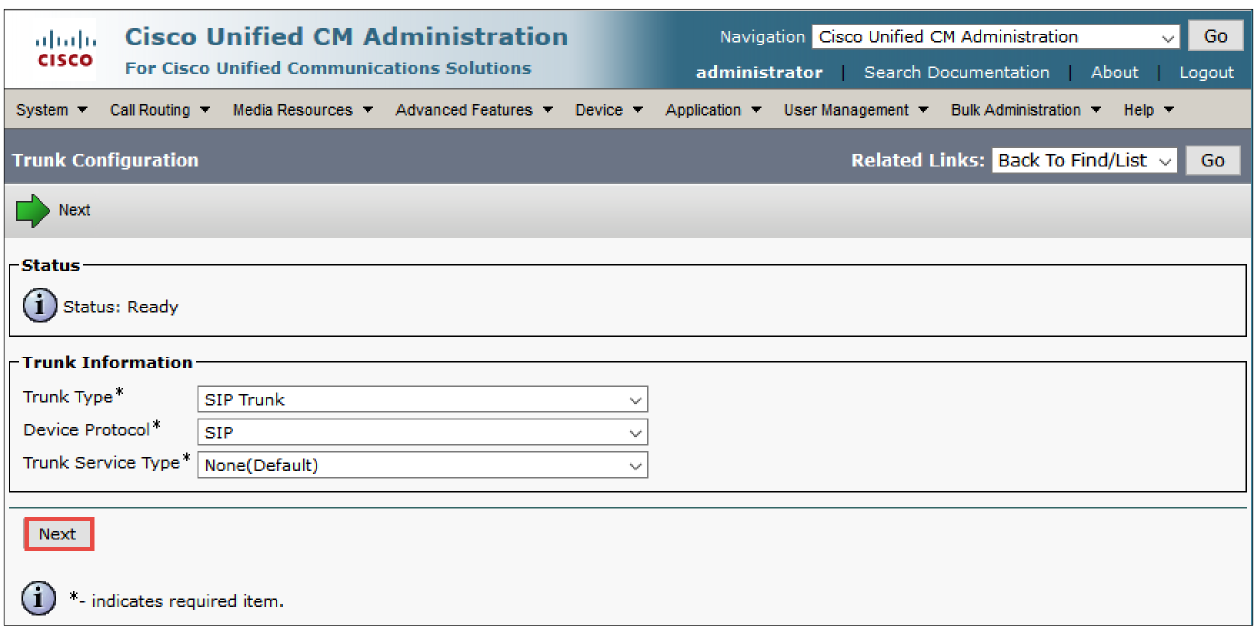


Figure 11: Add SIP Trunk Type

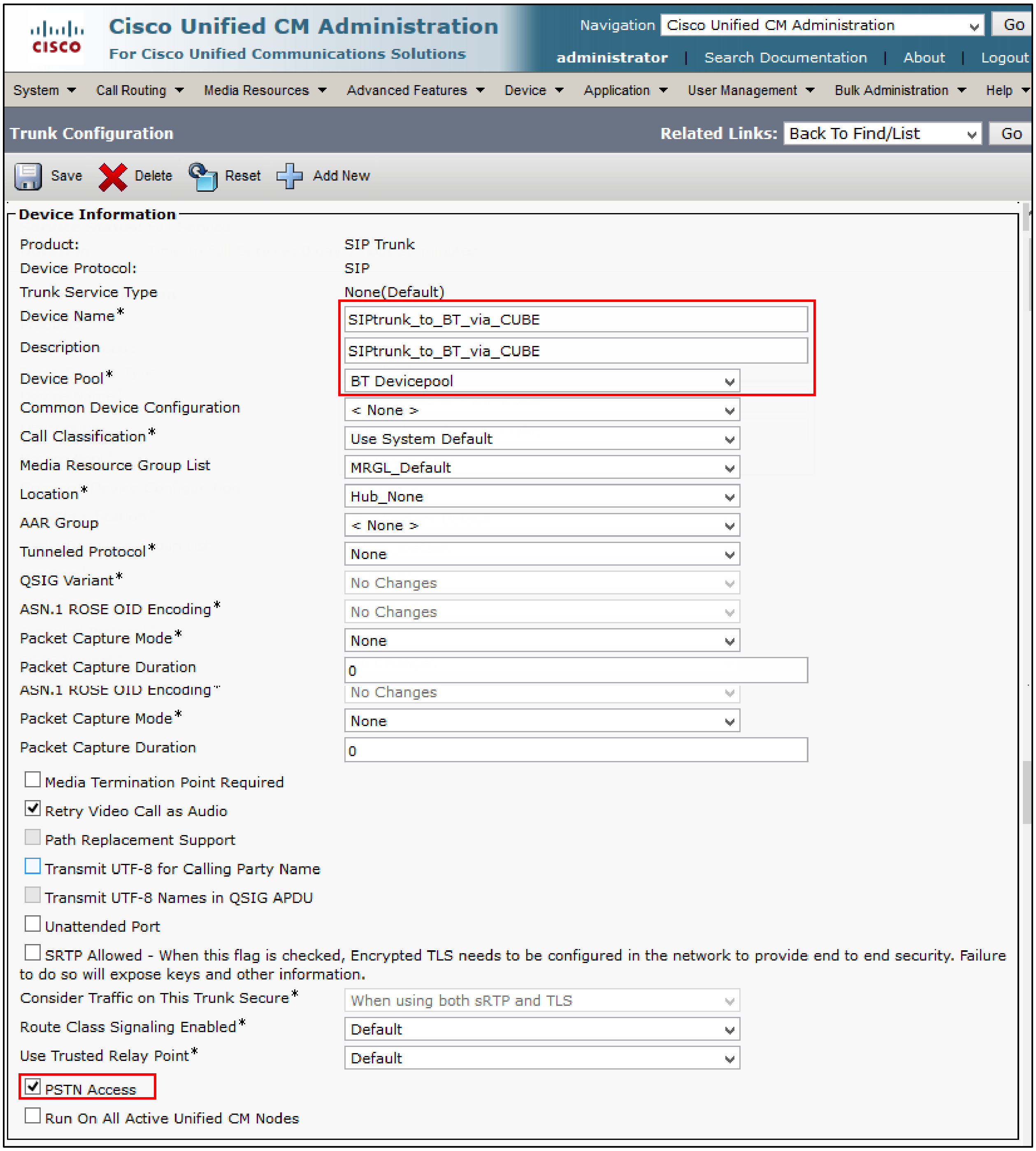


Figure 12: SIP Trunk to Cisco UBE

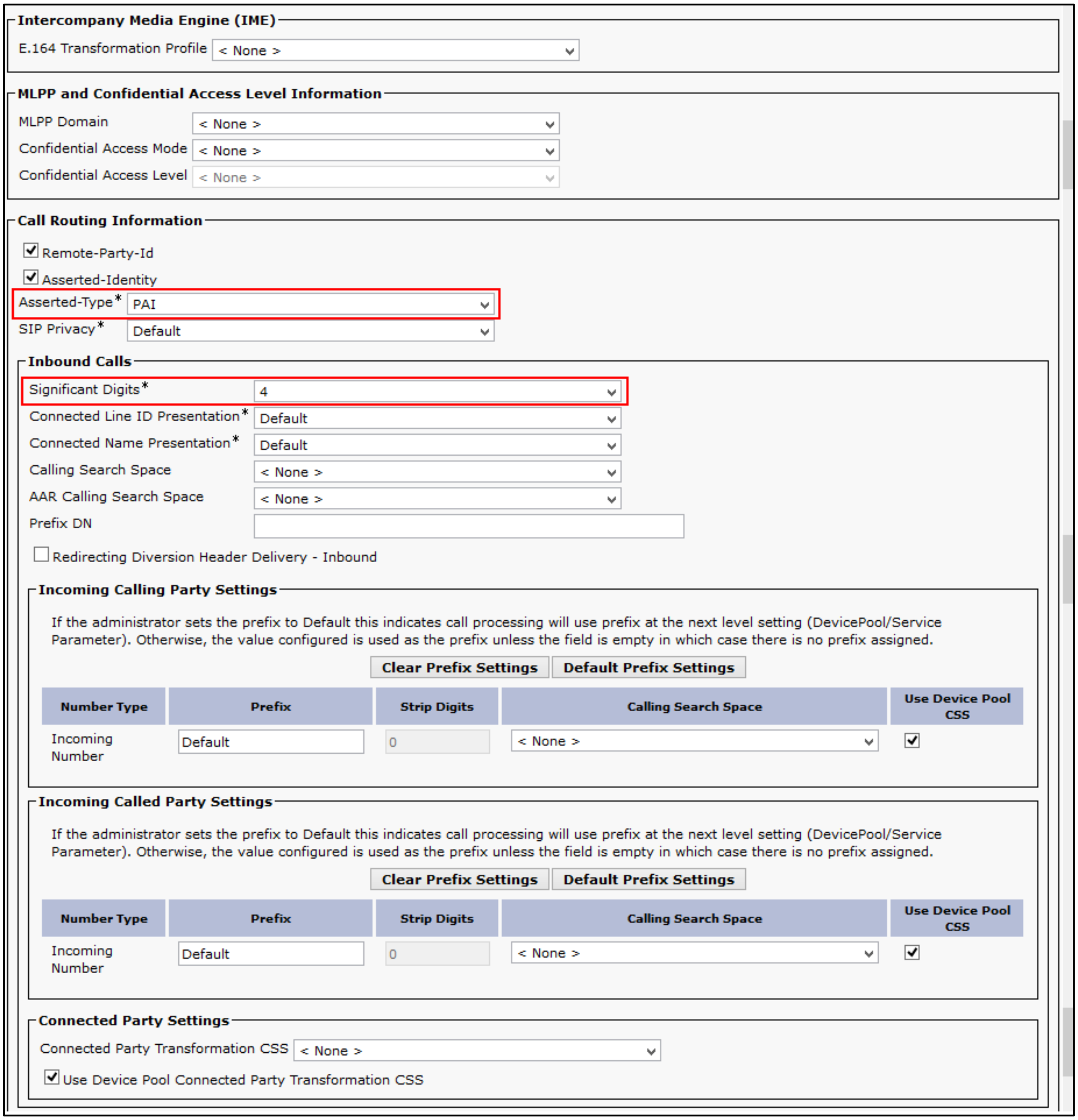


Figure 13: SIP Trunk to Cisco UBE (Cont.)

- 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
- The rest of the configuration is all default
- Click Save and Reset after completion

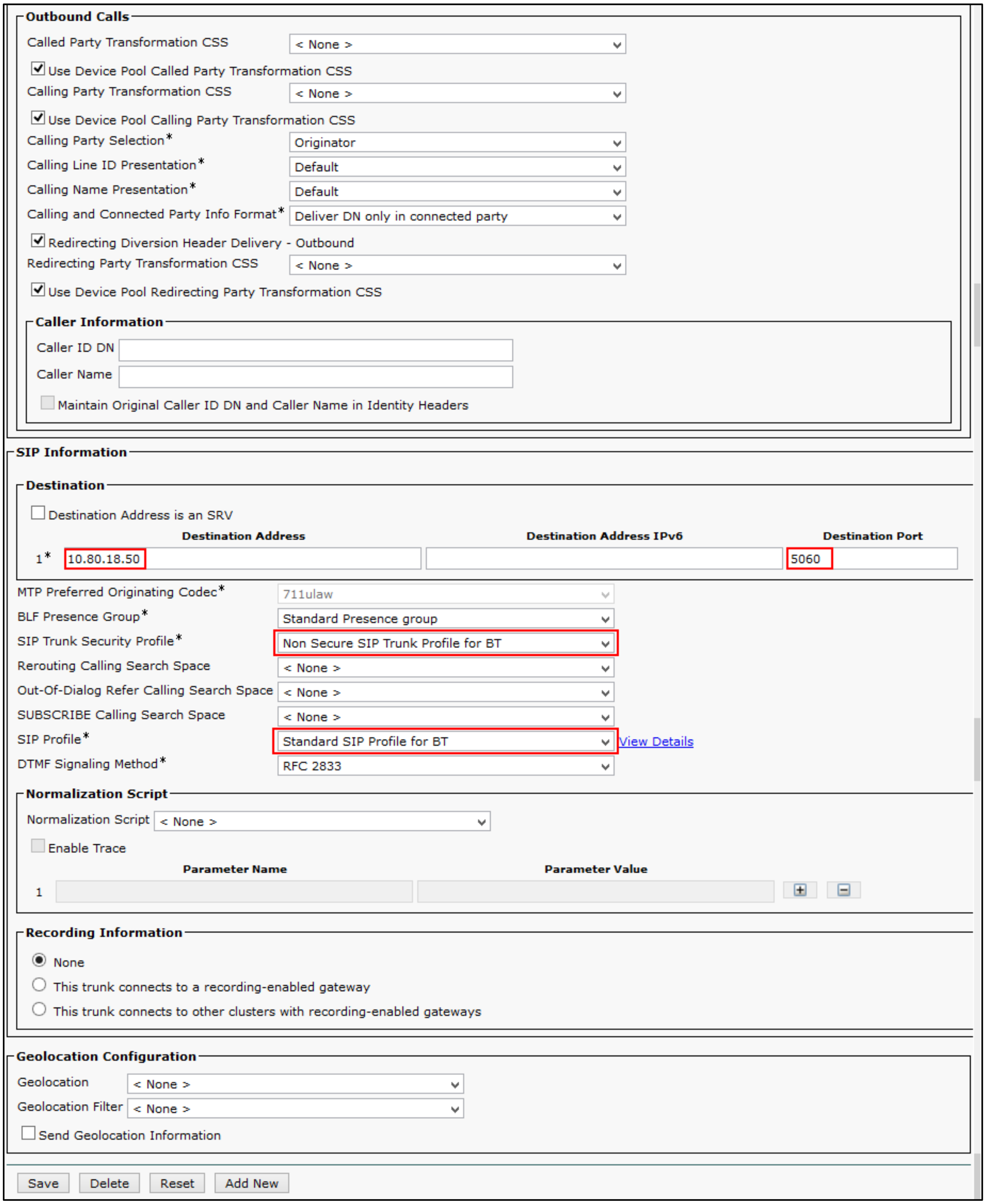


Figure 14: SIP Trunk to Cisco UBE (Cont.)

Trunk configuration from Cisco UCM to Fax Gateway:

Navigation: Devices → Trunk → Add New

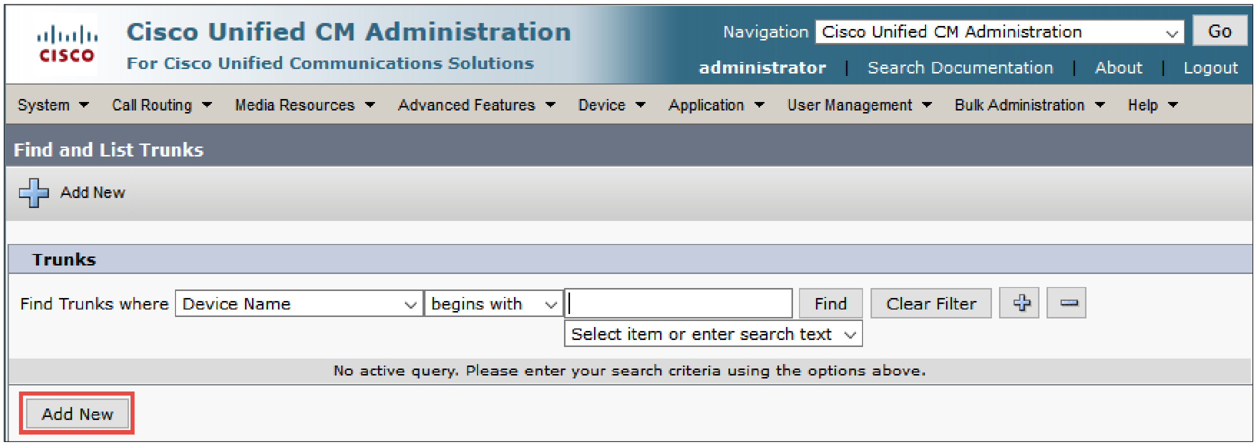


Figure 15: Add New Trunk to Fax Gateway

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

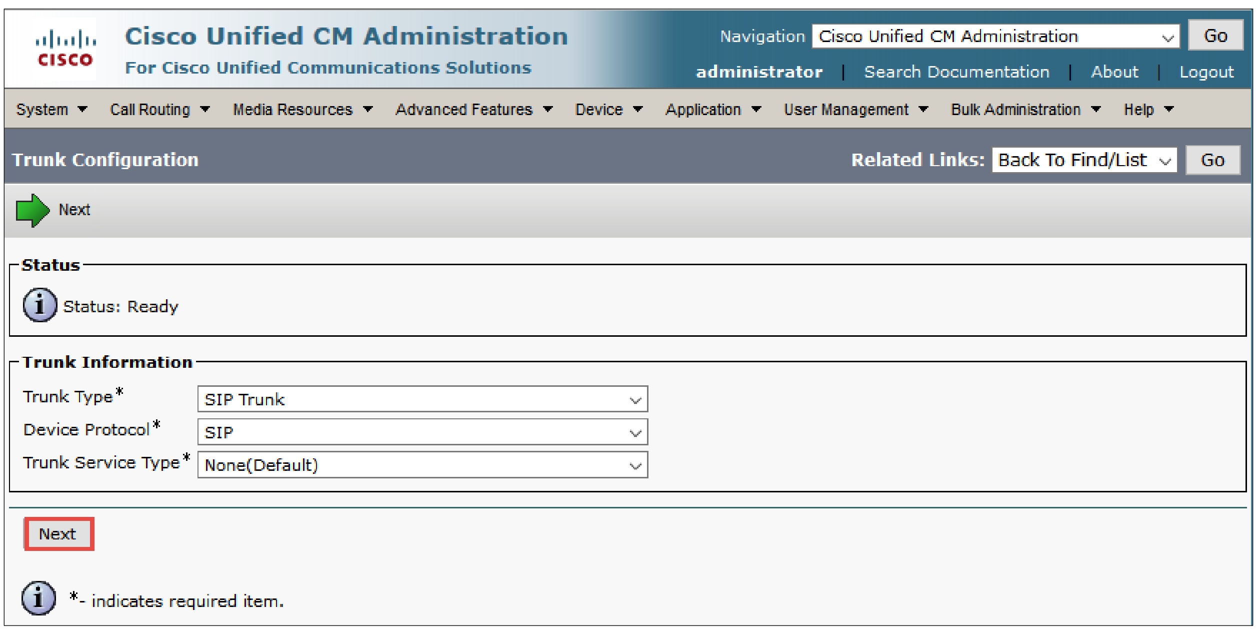


Figure 16: Add SIP Trunk Type

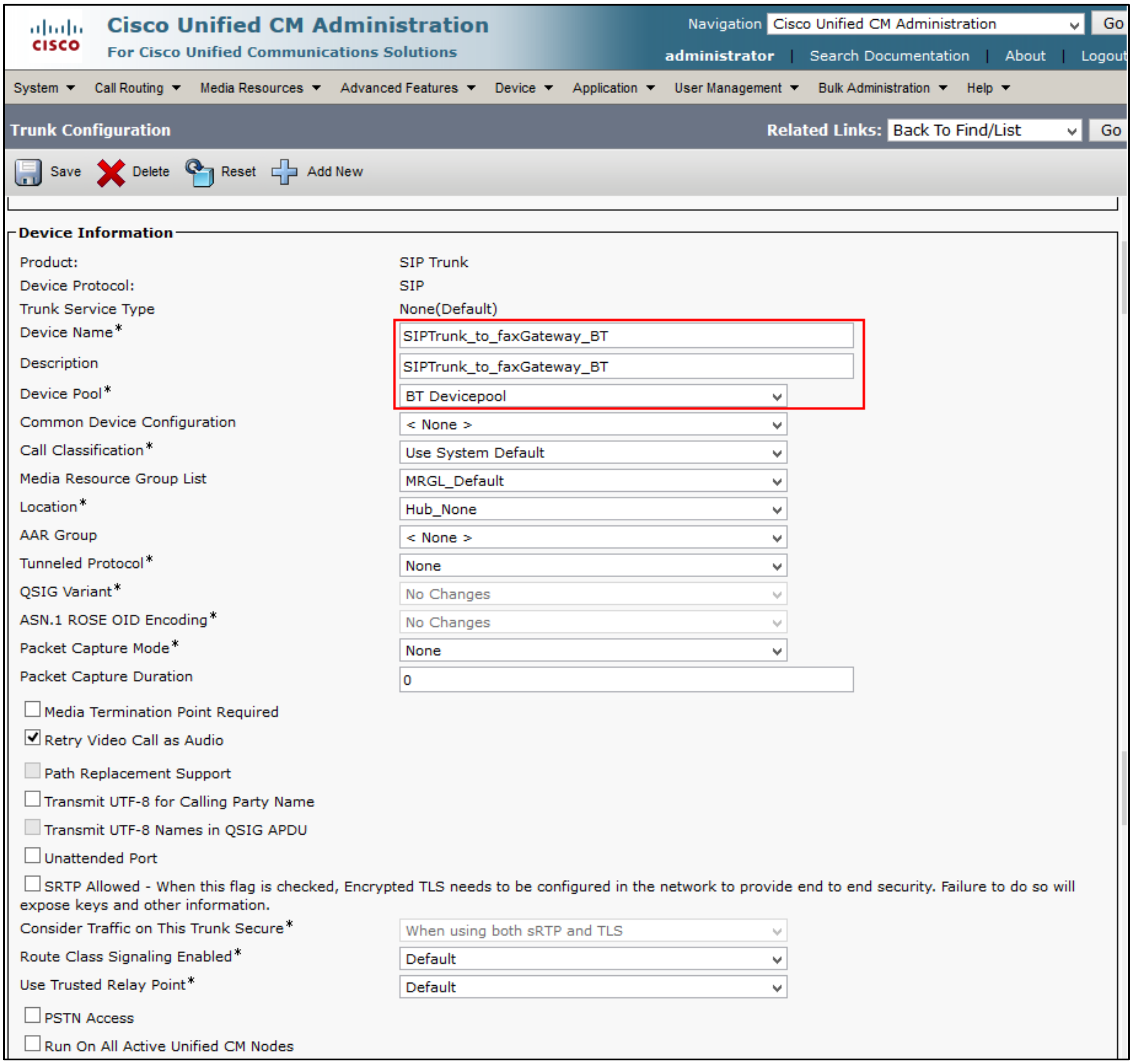


Figure 17: SIP Trunk to FAX Gateway

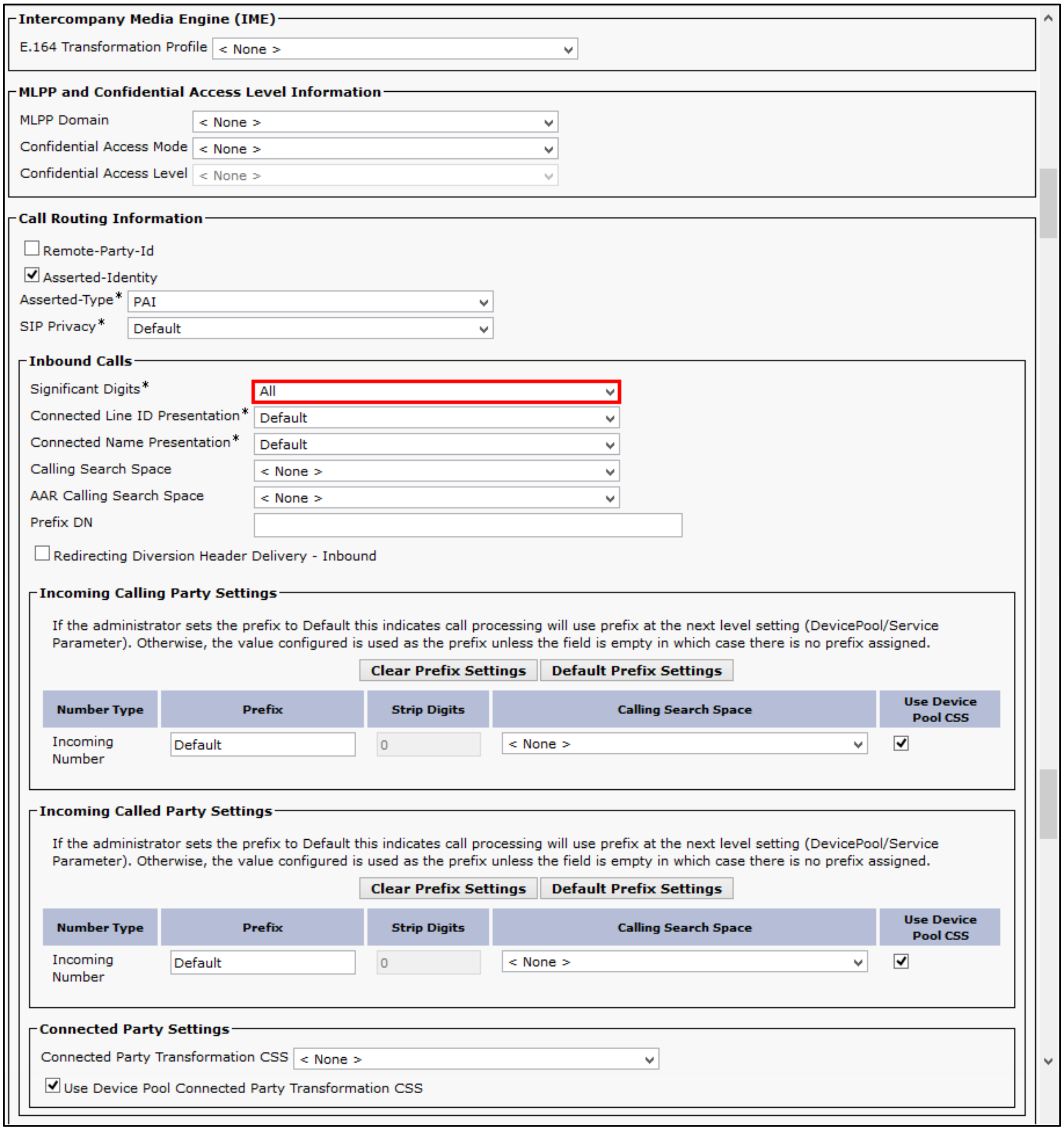


Figure 18: SIP Trunk to FAX Gateway (Cont.)

- 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

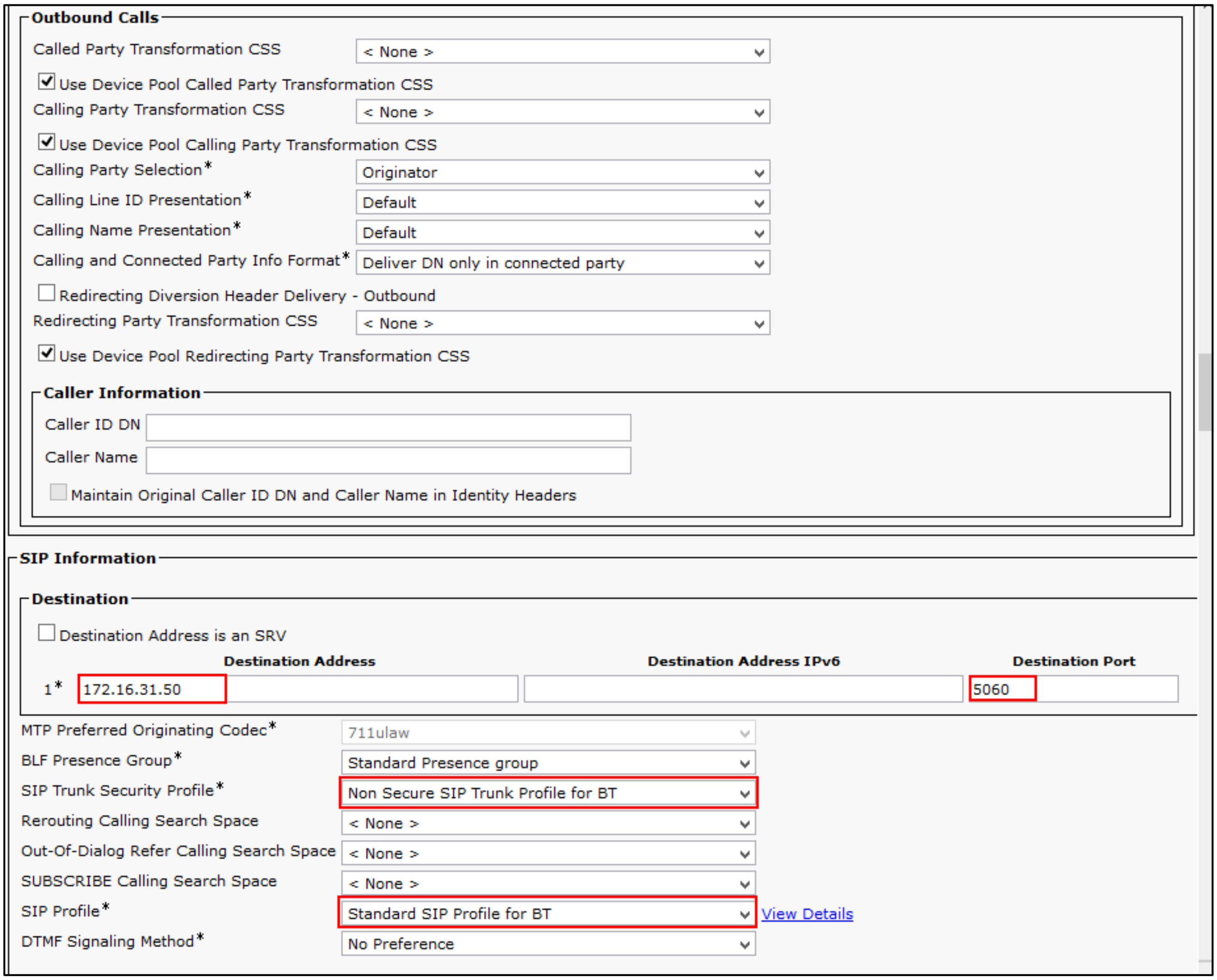


Figure 19: SIP Trunk to FAX Gateway (Cont.)

Routing configuration

Route Pattern for Cisco UBE:

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

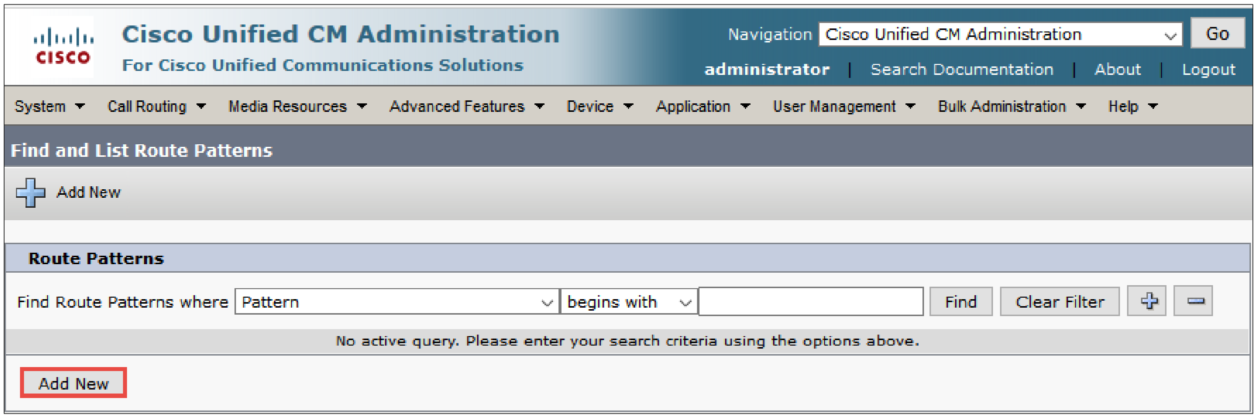


Figure 20: Add New Route Pattern for Cisco UBE

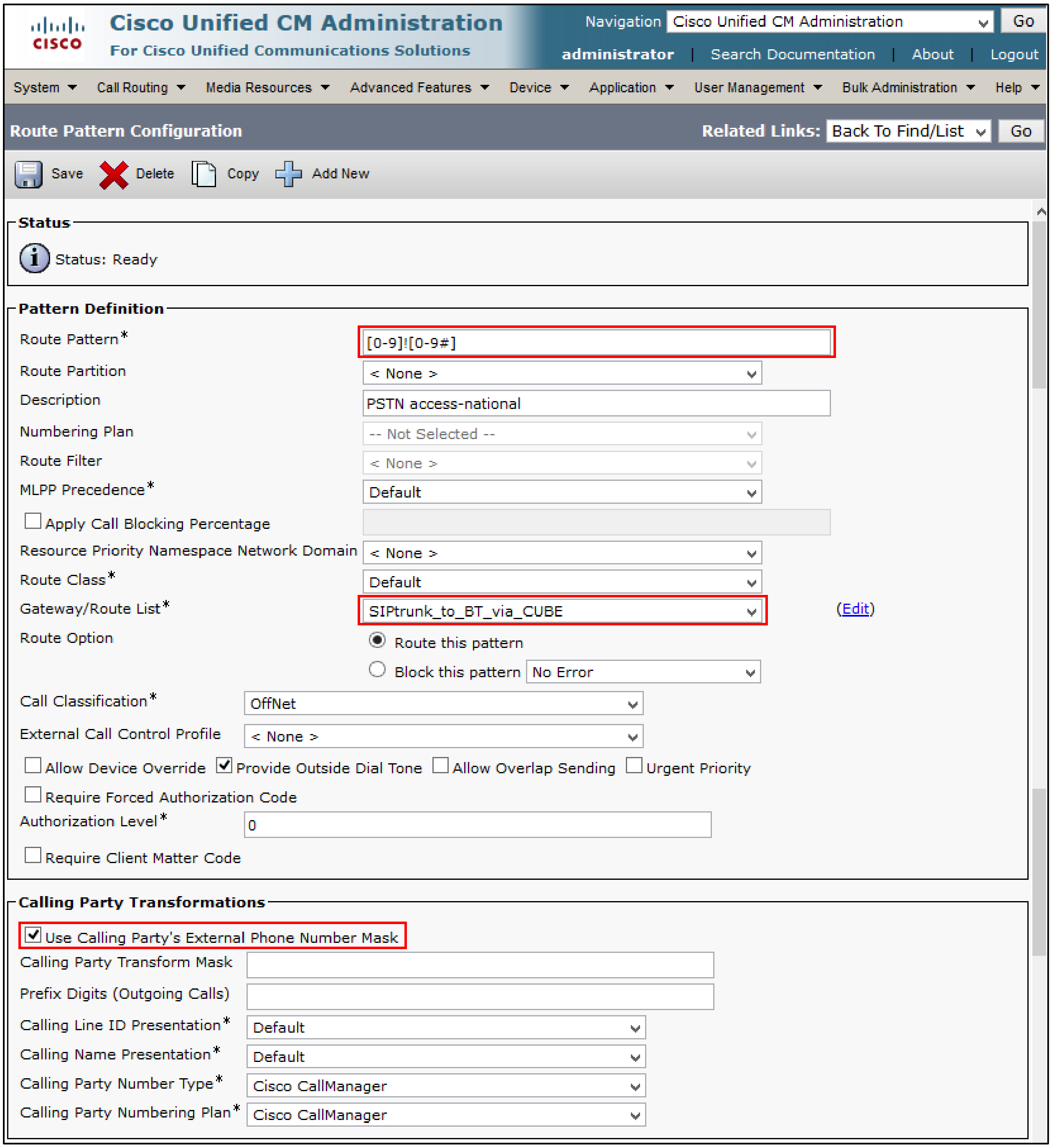


Figure 21: Route Pattern Configuration for Cisco UBE-PSTN Access

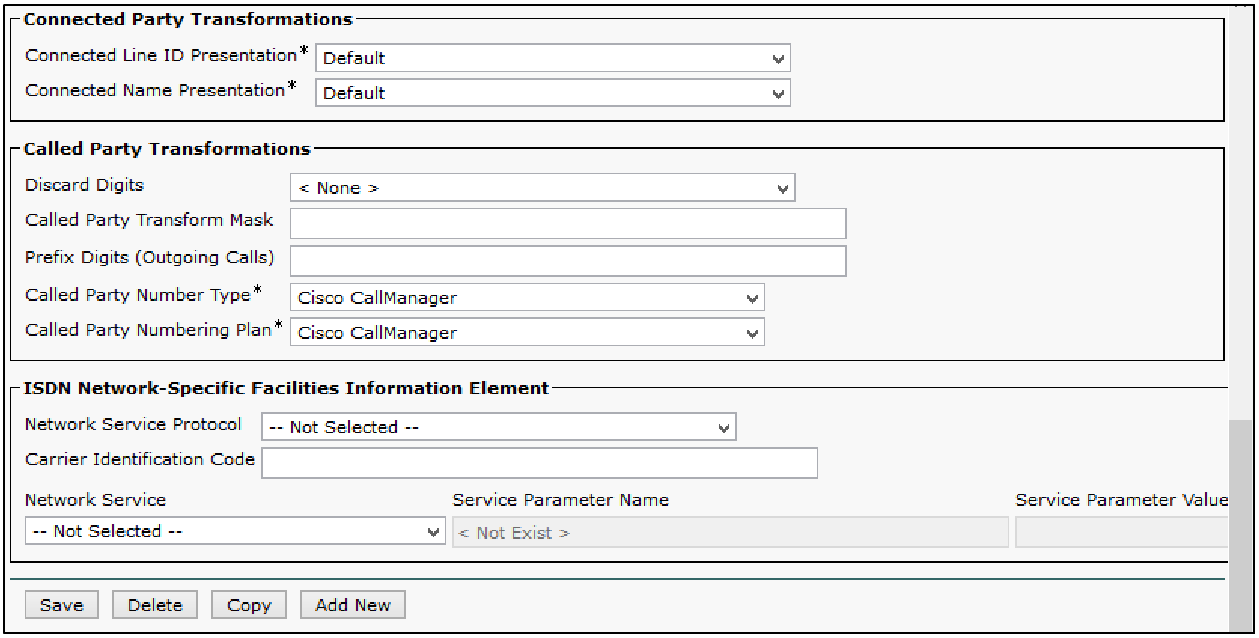


Figure 22: Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.)

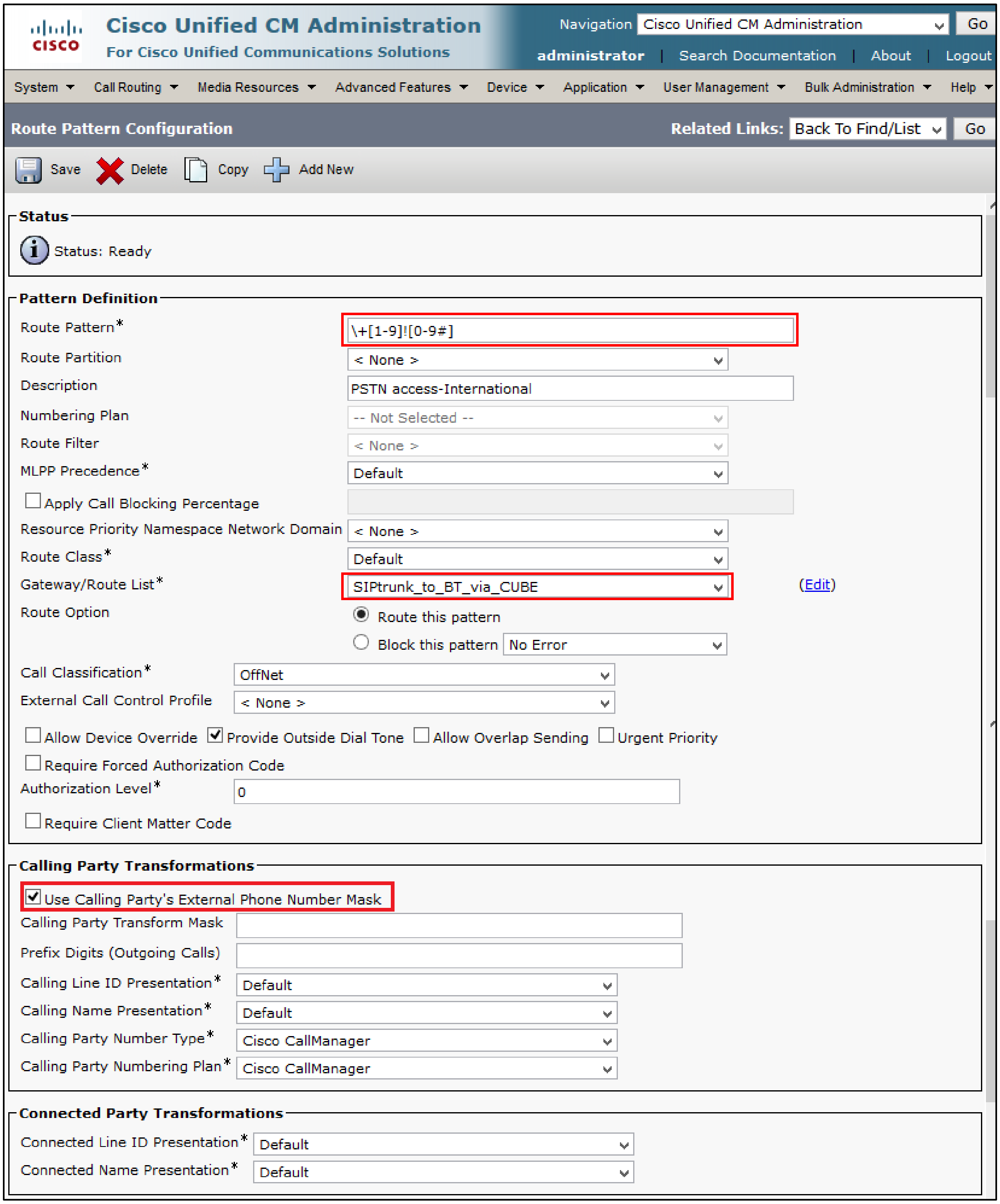


Figure 23: Route Pattern Configuration for Cisco UBE-PSTN Access-International

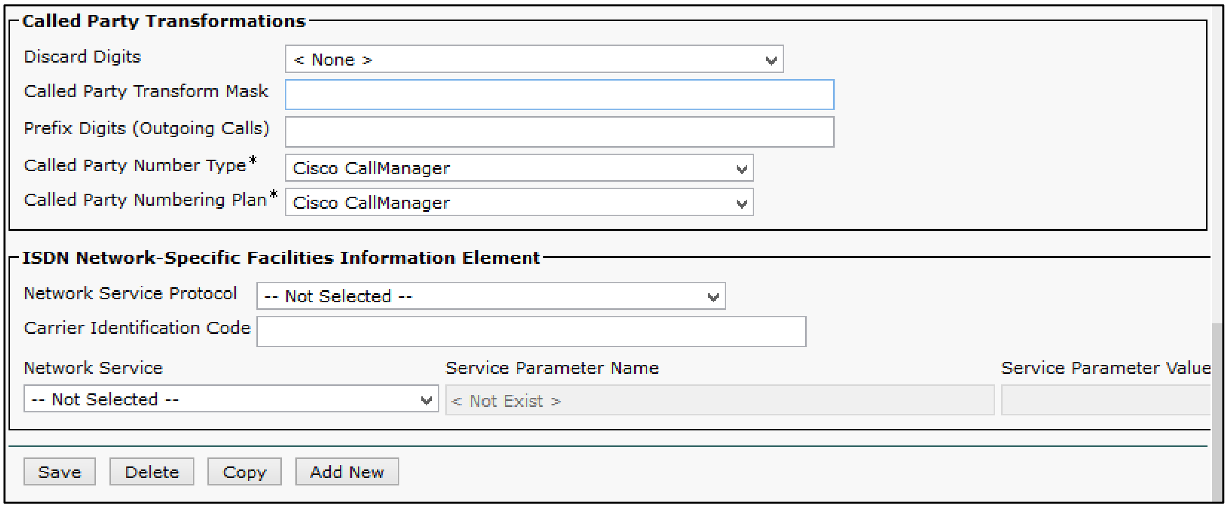


Figure 24: Route Pattern Configuration for Cisco UBE-PSTN Access-International (Cont.)

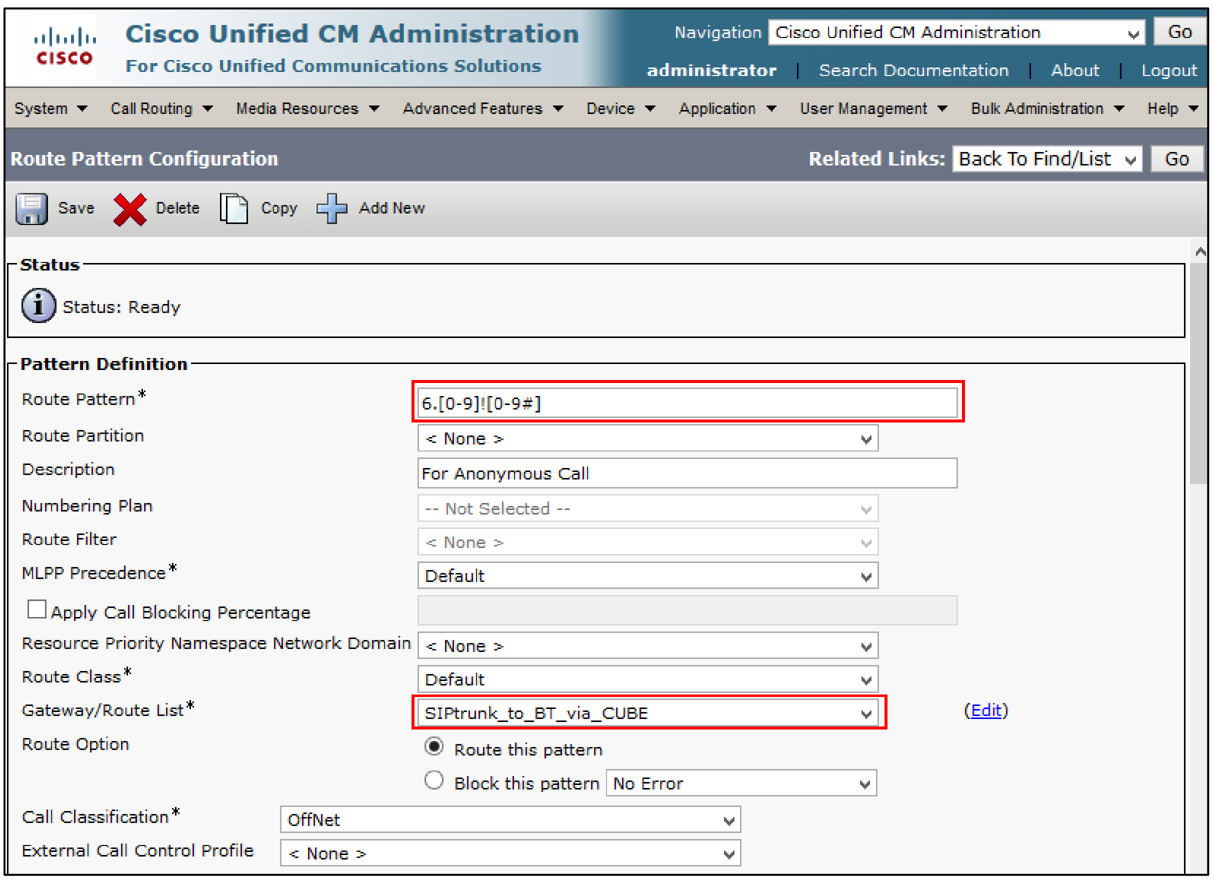


Figure 25 Route Pattern Configuration for Cisco UBE-PSTN Access National- Anonymous Call

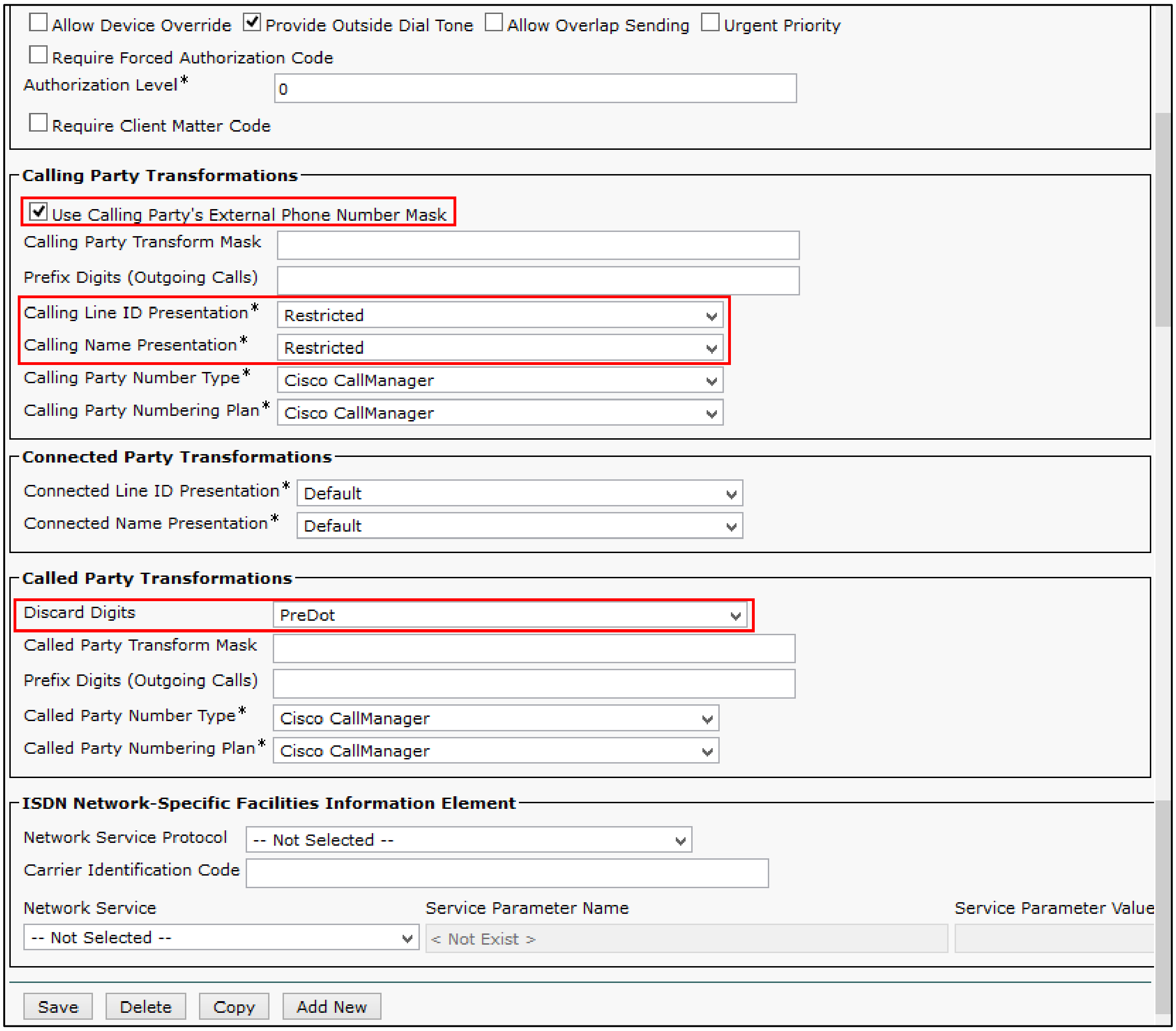


Figure 26: Route Pattern Configuration for Cisco UBE-PSTN Access National- Anonymous Call (Cont.)

Route Pattern for Fax Gateway:

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

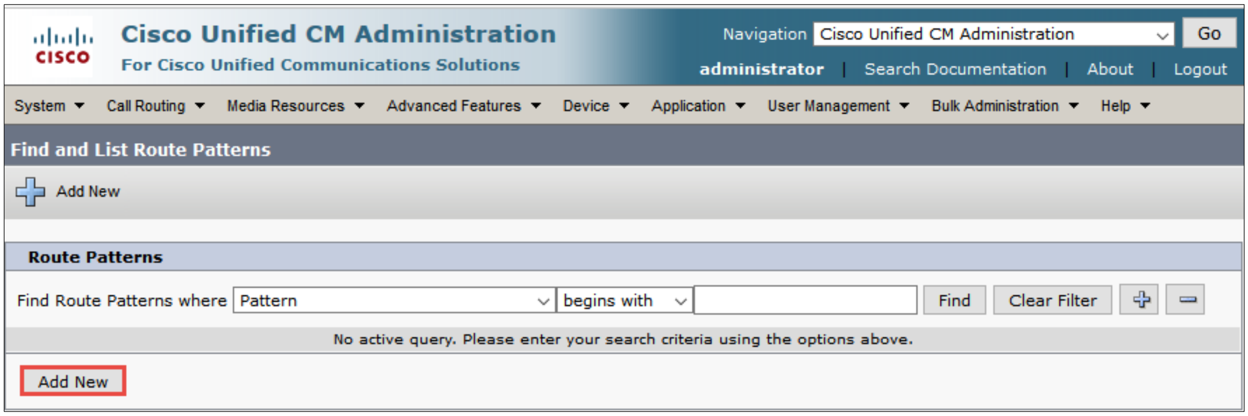


Figure 27: Add New Route patter to Fax Gateway

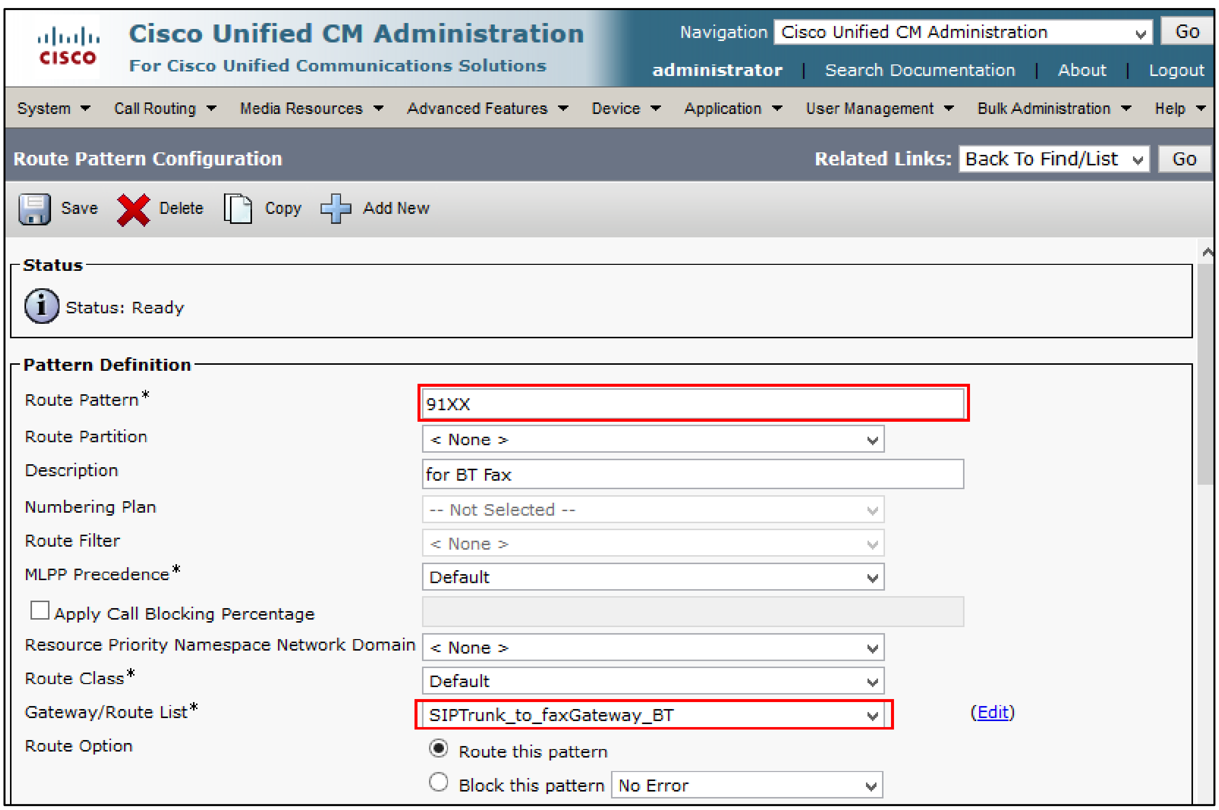


Figure 28: Route Pattern Configuration for Fax Gateway

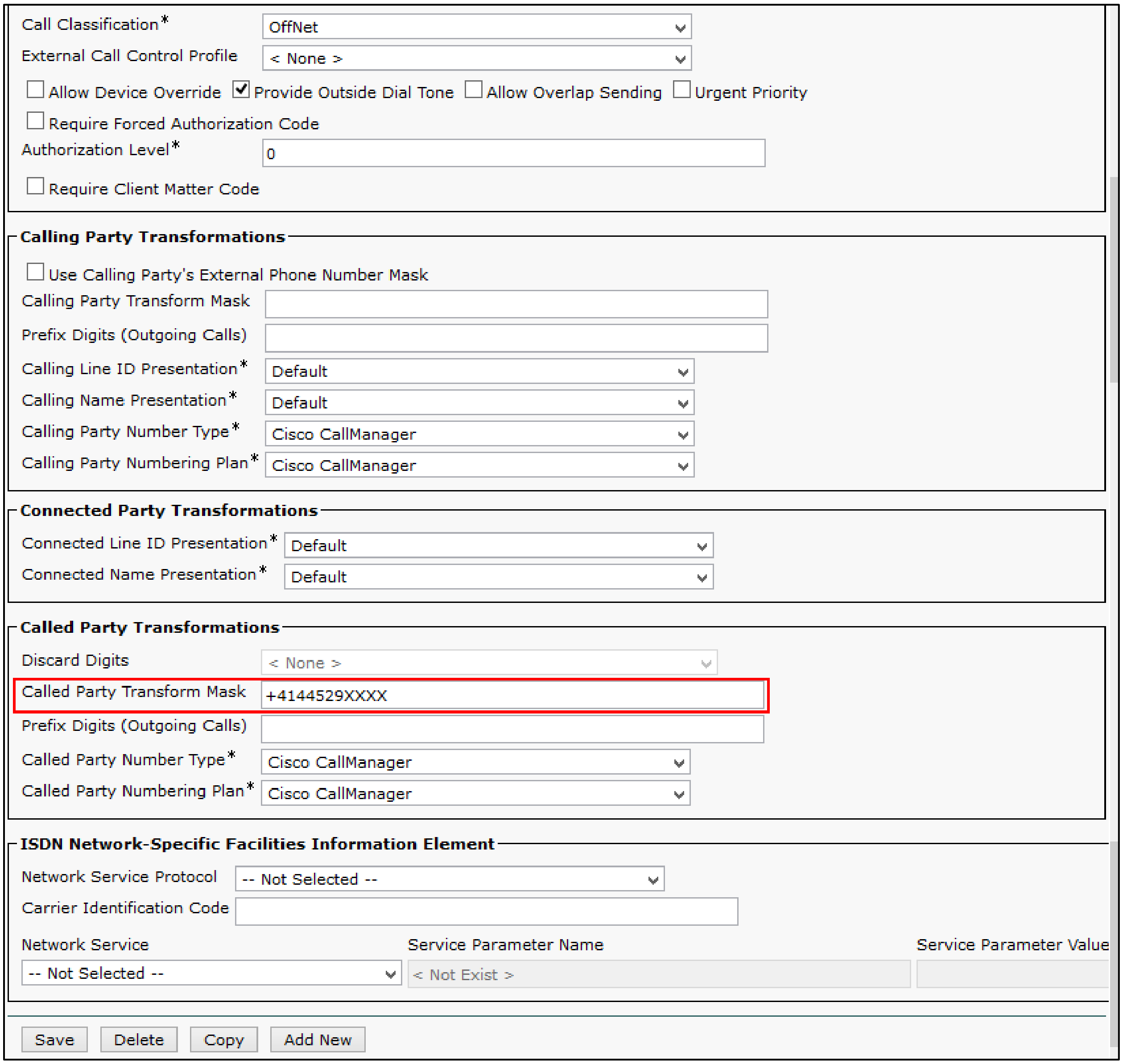


Figure 29: Route Pattern Configuration for Fax Gateway (Cont.)

Configuring Cisco Voice Gateway for Fax

Global Settings

```
voice service voip
 allow-connections sip to sip
 no supplementary-service sip handle-replaces
 redirect ip2ip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 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 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
!
voice class codec 2
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 g711alaw
!
```

Dial peer

Outbound Dial-peer to Cisco UCM:

```
dial-peer voice 4000 voip
 description BT Outbound FAX
 translation-profile outgoing E164dialing
 destination-pattern 1972265....
 session protocol sipv2
 session target ipv4:10.80.18.2
 session transport udp
 voice-class codec 1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
```

Inbound Dial-peer from Cisco UCM:

```
dial-peer voice 5000 voip
 description BT Inbound FAX
 service session
 session protocol sipv2
 session transport udp
 incoming called-number +4144529....
 voice-class codec 1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
```

POTS and Port Configuration:

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

```
dial-peer voice 45 pots
 huntstop
 service session
 destination-pattern +4144529XXXX
 no digit-strip
 port 0/0/0
 forward-digits all

voice-port 0/0/0
 no echo-cancel enable
 no vad
 cptone IN
 station-id name BT Fax user1
 station-id number +4144529XXXX
 caller-id enable
!
```

Configuration example

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

```
Current configuration : 13111 bytes
!
! Last configuration change at 16:30:57 IST Mon Nov 20 2017 by cisco
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
!
!
enable password ********
!
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
!
multilink bundle-name authenticated
!
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
```

```
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no fax-relay sg3-to-g3
 sip
  rel1xx disable
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
!
voice class codec 2
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 g711alaw
!
voice translation-rule 1
 rule 1 /\(^...........$\)/ /+\1/
!
voice translation-profile E164dialing
 translate called 1
!
voice-card 0
 dspfarm
 dsp services dspfarm
!
crypto pki token default removal timeout 0
!
license udi pid CISCO2851 sn FHK1137F4LY
username cisco password 0 ******
!
```

```
controller E1 0/2/0
 shutdown
 pri-group timeslots 1-31 service mgcp
!
controller E1 0/3/0
 clock source internal
 pri-group timeslots 1-31
!
interface GigabitEthernet0/0
 ip address 172.16.31.50 255.255.255.0
 duplex auto
 speed auto
!
interface Service-Engine0/0
 no ip address
 shutdown
!
interface GigabitEthernet0/1
 no ip address
 ip nat outside
 ip virtual-reassembly in
 shutdown
 duplex auto
 speed auto
!
interface Serial0/2/0:15
 no ip address
 !
interface Serial0/3/0:15
 no ip address
 !
interface Service-Engine1/0
 no ip address
```

```
 shutdown
!
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
!
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 BT Fax user1
 station-id number +4144529XXXX
 caller-id enable
!
voice-port 0/0/1
!
voice-port 0/3/0:15
!
voice-port 0/2/0:15
```

```
!
voice-port 0/1/0
 no echo-cancel enable
 no vad
 cptone IN
 station-id name BT Fax user2
 station-id number +4144529XXXX
 caller-id enable
!
voice-port 0/1/1
 !
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config
!
mgcp
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 5000 voip
 description BT Inbound FAX
 service session
 session protocol sipv2
 session transport udp
 incoming called-number +4144529....
 voice-class codec 1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 4000 voip
 description BT Outbound FAX
 translation-profile outgoing E164dialing
 destination-pattern 1972265....
 session protocol sipv2
 session target ipv4:10.80.18.2
 session transport udp
 voice-class codec 1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 45 pots
```

```
 huntstop
 service session
 destination-pattern +4144529XXXX
 no digit-strip
 port 0/0/0
 forward-digits all
!
dial-peer voice 4001 voip
 description BT Outbound FAX loopback
 destination-pattern 4144529....
 session protocol sipv2
 session target ipv4:10.80.18.2
 session transport udp
 voice-class codec 1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 46 pots
 huntstop
 service session
 destination-pattern +4144529XXXX
 no digit-strip
 port 0/1/0
 forward-digits all
!
!
telephony-service
 max-ephones 50
 max-dn 60
```

```
 ip source-address 172.16.31.50 port 2000
 service phone sshAccess 0
 cnf-file perphone
 max-conferences 8 gain -6
 web admin system name user password ******
 transfer-system full-consult
 create cnf-files version-stamp 7960 Nov 22 2013 19:05:58
!
banner login ^CC
!
line con 0
line aux 0
line 66
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
line 194
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output all
line vty 0 4
 session-timeout 180
 exec-timeout 0 0
 password *******
 login authentication local_auth
 transport input all
!
scheduler allocate 20000 1000
ntp server 103.6.16.254
```

```
end
cme.in.tekvizion.com#
```

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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