Application Note

Connecting Cisco Unified Communication Manager [v11.5.1] to JT Global SIP Trunks via Cisco Unified Border Element v12.0.0 [IOS-XE 16.06.01]

May 14, 2018

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Table of Contents

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Table of Figures

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Introduction

Service Providers today, such as JT Global, 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.

JT Global 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 JT Global network, Cisco Unified Border Element (Cisco UBE) ISR 4331/K9 running IOS 16.06.01 can be used. The Cisco Unified Border Element 16.06.01 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to JT Global 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 JT Global 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 4331/K9 [IOS – 16.06.01] for connectivity to JT Global SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (JT Global) via Cisco UBE v12.0 [IOS-XE] 16.06.01.
- Testing was performed in accordance to JT Global 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 JT Global 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 JT Global SIP Trunking network.

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Network Topology

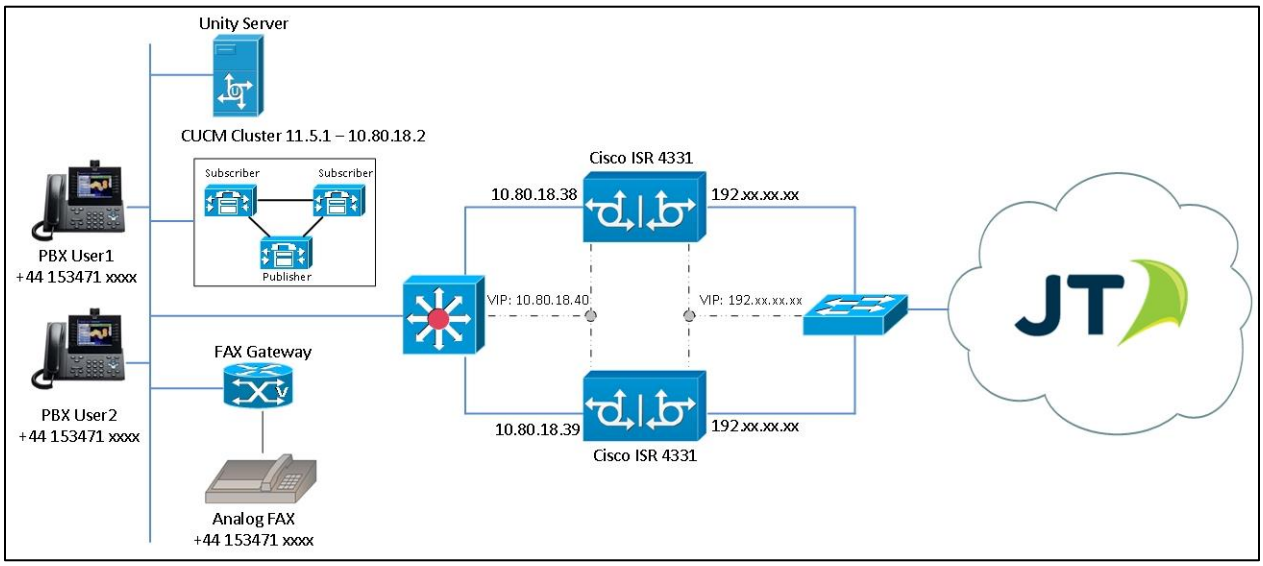


Figure 1 Network Topology

- 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. JT Global 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 UBE and Cisco UCM is TCP and to JT Global is UDP.

Cisco UCM and Cisco UBE Settings:

Setting	Value
Transport from Cisco UBE to Cisco UCM	TCP with RTP
Transport from Cisco UBE to JT Global	UDP with RTP
Voice Mail Support	YES
Session Refresh	YES
Early Media support with PRACK	NO

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

System Components

Hardware Requirements

- Cisco UBE on Cisco ISR 4331/K9 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: 12.0.0 running IOS-XE 16.06.01
- CUCM UCOS 11.5.1.12900-21 for 1 Publisher and 2 Subscriber
- Cisco IOS v15.1(4)M5 for the fax gateway

Features

Features Supported

- Incoming and outgoing calls using G711alaw 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 (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
- JT Global does not support Fax at Super G3 Speed
- JT Global does not support T.38 Fax

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

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.
- JT Global does not support SG3 Fax. Fax test cases are executed only on G3.
- Transcoder has been enabled on Cisco UBE to support mid call codec renegotiation for Conference and transfer scenarios. Please refer the Cisco UBE configuration section for more details.
- For anonymous call test case, JT Global suggested to prefix +44153414100 with DID for the anonymous call
- While doing conference and transfer scenario, Mid-call-codec-renegotiation was not happening. Call established with PSTN 1 with G729 and then PBX initiate conference to PSTN 2 with G729. When complete the conference on PBX. Cisco UCM sends re-INVITE with G711alaw only on SDP for codec re negotiation but JT Global respond back with 200 OK/SDP contains only G729 due to this condition the CUBE disconnected the call
- For PRACK test case, SBC of JT Global strips the '100rel' from the 183 Session Progress message.
- While JT Global was sending CANCEL message towards Cisco UCM, Request URI of CANCEL contains "CANCEL sip:+JESIPSRVXXXXX@192.XX.XX.XX:5060 SIP/2.0" for that Cisco UBE has disconnected the call. To resolve this issue the SIP profile has been used. Please refer the Cisco UBE configuration section for more details.
- Tested with E.164 format for all calls.

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

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.

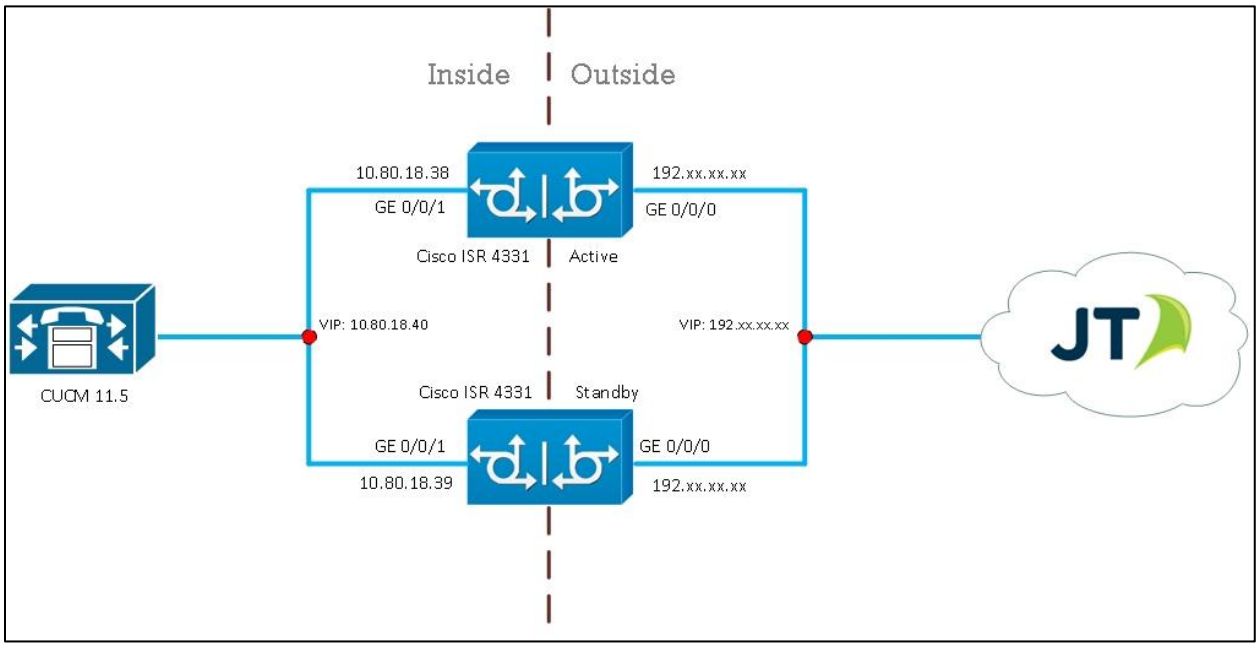


Figure 2 High Availability topology

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Cisco UBE 1:

```
interface GigabitEthernet0/0/0
 ip address 192.XX.XX.XX 255.255.255.224
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
 ip address 10.80.18.38 255.255.255.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.40 exclusive
!
interface GigabitEthernet0/0/2
 ip address 1.1.1.5 255.255.255.252
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Cisco UBE 2:

```
interface GigabitEthernet0/0/0
 ip address 192.XX.XX.XX 255.255.255.224
 speed 1000
 no negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
 ip address 10.80.18.39 255.255.255.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.40 exclusive
!
interface GigabitEthernet0/0/2
 ip address 1.1.1.6 255.255.255.252
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

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
 mode border-element license capacity 200
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711alaw
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  rel1xx supported "rel100"
  session refresh
  header-passing
  asserted-id pai
  privacy pstn
  midcall-signaling passthru media-change
  early-offer forced
  privacy-policy passthru
  g729 annexb-all
  sip-profiles inbound
  sip-profiles 222 inbound
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

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
`midcall-signaling passthru media-change`	Pass-through media change method optimizes or consumes mid-call, media-related signaling within the call
`sip-profiles inbound`	Enables inbound SIP profiles feature
`sip-profiles 222 inbound`	SIP profile can be configured with rules to add, remove, copy, or modify the SIP, Session Description Protocol (SDP), and peer headers that enter or leave CUBE.
`g729 annexb-all`	Specifies that the G.729br8 codec is treated as a superset of G.729r8 and G.729br8 codecs to communicate with Cisco UCM.

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Codecs

G711alaw 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 g711alaw
 !
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711alaw
 codec preference 3 g711ulaw
!
voice class codec 3
 codec preference 1 g711alaw
!
```

Dial peer

Outbound Dial-peer to JT Global:

```
dial-peer voice 10 voip
 description Incoming from CUCM
 huntstop
 session protocol sipv2
 incoming called-number 1T
 voice-class codec 3
 voice-class sip profiles 1
 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
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 20 voip
 description Outgoing to JT_Global
 translation-profile outgoing e164
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target ipv4:212.9.11.194
 voice-class codec 3
 voice-class sip options-ping 60
 voice-class sip profiles 1
 no voice-class sip pass-thru headers
 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
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
```

Inbound Dial-peer from JT Global:

```
dial-peer voice 30 voip
 description Incoming from JT_Global
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 huntstop
 session protocol sipv2
 incoming called-number +44
 voice-class codec 3
 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
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 40 voip
 description Outgoing to CUCM
 huntstop
 destination-pattern +44
 session protocol sipv2
 session target ipv4:10.80.18.2
 voice-class codec 3
 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
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Configuration example

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

Active Cisco UBE:

```
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 JTCertCube10
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no aaa new-model
no ip domain lookup
!
subscriber templating
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-2930804041
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2930804041
 revocation-check none
 rsakeypair TP-self-signed-2930804041
!
!
crypto pki certificate chain TP-self-signed-2930804041
!
!
!
!
voice service voip
 no ip address trusted authenticate
 mode border-element license capacity 200
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711alaw
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  rel1xx supported "rel100"
  session refresh
  header-passing
  asserted-id pai
  privacy pstn
  midcall-signaling passthru media-change
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
  early-offer forced
  privacy-policy passthru
  g729 annexb-all
  sip-profiles inbound
  sip-profiles 222 inbound
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711alaw
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711alaw
 codec preference 3 g711ulaw
!
voice class codec 3
 codec preference 1 g711alaw
!
!
voice class sip-profiles 222
 request CANCEL sip-header To copy "sip:(.*)@" u01
 request CANCEL sip-header SIP-Req-URI modify ".*@(.*)" "CANCEL sip:\u01@\1"
 request ACK sip-header To copy "sip:(.*)@" u03
 request ACK sip-header SIP-Req-URI modify ".*@(.*)" "ACK sip:\u03@\1"
!
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>"
"<sip:+44153471\1@\2>"
!
!
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
!
!
!
voice translation-rule 10
 rule 1 /\(^1..........$\)/ /+\1/
 rule 2 /\(^..........$\)/ /+44\1/
 rule 3 /\(^0..........$\)/ /\1/
 rule 4 /\(^441534141001..........$\)/ /+\1/
!
!
voice translation-profile e164
 translate called 10
!
voice-card 0/4
 dsp services dspfarm
 no watchdog
!
license udi pid ISR4331/K9 sn FDO2876767
diagnostic bootup level minimal
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   timers delay 30 reload 60
   control GigabitEthernet0/0/2 protocol 1
   data GigabitEthernet0/0/2
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/1 line-protocol
track 2 interface GigabitEthernet0/0/0 line-protocol
!
interface GigabitEthernet0/0/0
 ip address 192.XX.XX.XX 255.255.255.224
 speed 1000
 no negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
 ip address 10.80.18.39 255.255.255.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.40 exclusive
!
interface GigabitEthernet0/0/2
 ip address 1.1.1.6 255.255.255.252
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 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 tftp source-interface GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 0.0.0.0 0.0.0.0 192.XX.XX.XX
ip route 10.64.0.0 255.255.0.0 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
!
dspfarm profile 111 transcode
 codec g729abr8
 codec g729ar8
 codec g711alaw
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 codec g711ulaw
 codec g729r8
 codec g729br8
 maximum sessions 10
 associate application CUBE
!
dial-peer voice 10 voip
 description Incoming from CUCM
 huntstop
 session protocol sipv2
 incoming called-number 1T
 voice-class codec 3
 voice-class sip profiles 1
 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
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 30 voip
 description Incoming from JT_Global
 huntstop
 session protocol sipv2
 incoming called-number +44
 voice-class codec 3
 voice-class sip bind control source-interface GigabitEthernet0/0/0
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 40 voip
 description Outgoing to CUCM
 huntstop
 destination-pattern +44
 session protocol sipv2
 session target ipv4:10.80.18.2
 voice-class codec 3
 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
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 118 voip
 description Operator number
 destination-pattern 118534
 session protocol sipv2
 session target ipv4:212.XX.XX.XXX
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 voice-class codec 3
 no voice-class sip pass-thru headers
 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 999 voip
 description Emergency
 huntstop
 session protocol sipv2
 incoming called-number 999
 voice-class codec 3
 no voice-class sip pass-thru headers
 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 20 voip
 description Outgoing to JT_Global
 translation-profile outgoing e164
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target ipv4:212.XX.XX.XXX
 voice-class codec 3
 voice-class sip options-ping 60
 voice-class sip profiles 1
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 no voice-class sip pass-thru headers
 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
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
!
sip-ua
 credentials username JESIPSRV00487 password 7 XXXXXXXXX realm 212.XX.XX.XXX
 authentication username JESIPSRV00487 password 7 XXXXXXXXX
 registrar ipv4:212.XX.XX.XXX expires 3600
!
!
line con 0
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 password 7 15060E07327B3175783D
 login local
 transport input telnet
!
end
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Standby Cisco UBE:

```
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 JTCertCube9
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no aaa new-model
no ip domain lookup
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-3793611302
 enrollment selfsigned
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 subject-name cn=IOS-Self-Signed-Certificate-3793611302
 revocation-check none
 rsakeypair TP-self-signed-3793611302
!
crypto pki certificate chain TP-self-signed-3793611302
!
voice service voip
 no ip address trusted authenticate
 mode border-element license capacity 200
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711alaw
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  rel1xx supported "rel100"
  session refresh
  header-passing
  asserted-id pai
  privacy pstn
  midcall-signaling passthru media-change
  early-offer forced
  privacy-policy passthru
  g729 annexb-all
  sip-profiles inbound
  sip-profiles 222 inbound
!
voice class codec 1
 codec preference 1 g729r8
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 codec preference 2 g711alaw
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711alaw
 codec preference 3 g711ulaw
!
voice class codec 3
 codec preference 1 g711alaw
 !
!
voice class sip-profiles 222
 request CANCEL sip-header To copy "sip:(.*)@" u01
 request CANCEL sip-header SIP-Req-URI modify ".*@(.*)" "CANCEL sip:\u01@\1"
 request ACK sip-header To copy "sip:(.*)@" u03
 request ACK sip-header SIP-Req-URI modify ".*@(.*)" "ACK sip:\u03@\1"
!
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>"
"<sip:+44153471\1@\2>"
!
!
voice translation-rule 10
 rule 1 /\(^1..........$\)/ /+\1/
 rule 2 /\(^..........$\)/ /+44\1/
 rule 3 /\(^0..........$\)/ /\1/
 rule 4 /\(^441534141001..........$\)/ /+\1/
!
voice translation-profile e164
 translate called 10
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
voice-card 0/4
 dsp services dspfarm
 no watchdog
!
license udi pid ISR4331/K9 sn FDO213777G
diagnostic bootup level minimal
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   timers delay 30 reload 60
   control GigabitEthernet0/0/2 protocol 1
   data GigabitEthernet0/0/2
   track 1 shutdown
   track 2 shutdown
!
!
!
track 1 interface GigabitEthernet0/0/1 line-protocol
track 2 interface GigabitEthernet0/0/0 line-protocol
!
interface GigabitEthernet0/0/0
 ip address 192.XX.XX.XX 255.255.255.224
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.79.92 exclusive
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
!
interface GigabitEthernet0/0/1
 ip address 10.80.18.38 255.255.255.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.40 exclusive
!
interface GigabitEthernet0/0/2
 ip address 1.1.1.5 255.255.255.252
 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 0.0.0.0 192.XX.XX.XX
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.80.0.0 255.255.0.0 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
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
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
!
dspfarm profile 111 transcode
 codec g729abr8
 codec g729ar8
 codec g711alaw
 codec g711ulaw
 codec g729r8
 codec g729br8
 maximum sessions 10
 associate application CUBE
!
dial-peer voice 10 voip
 description Incoming from CUCM
 huntstop
 session protocol sipv2
 incoming called-number 1T
 voice-class codec 3
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 20 voip
 description Outgoing to JT_Global
 translation-profile outgoing e164
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target ipv4:212.XX.XX.XXX
 voice-class codec 3
 voice-class sip options-ping 60
 voice-class sip profiles 1
 no voice-class sip pass-thru headers
 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
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 30 voip
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 description Incoming from JT_Global
 huntstop
 session protocol sipv2
 incoming called-number +44
 voice-class codec 3
 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
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 40 voip
 description Outgoing to CUCM
 huntstop
 destination-pattern +44
 session protocol sipv2
 session target ipv4:10.80.18.2
 voice-class codec 3
 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
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
!
dial-peer voice 118 voip
 description Operator number
 destination-pattern 118534
 session protocol sipv2
 session target ipv4:212.XX.XX.XXX
 voice-class codec 3
 no voice-class sip pass-thru headers
 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 999 voip
 description Emergency
 huntstop
 session protocol sipv2
 incoming called-number 999
 voice-class codec 3
 no voice-class sip pass-thru headers
 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
!
!
sip-ua
 credentials username JESIPSRV00487 password 7 XXXXXXXXX realm 212.XX.XX.XXX
 authentication username JESIPSRV00487 password 7 XXXXXXXXX
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 registrar ipv4:212.XX.XX.XXX expires 60
!
!
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 05000000
 login local
 transport input telnet
!
ntp server 34.202.215.187
ntp server pool.ntp.org
!
end
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Configuring Cisco UCM 11.5 Cluster

Cisco UCM Version

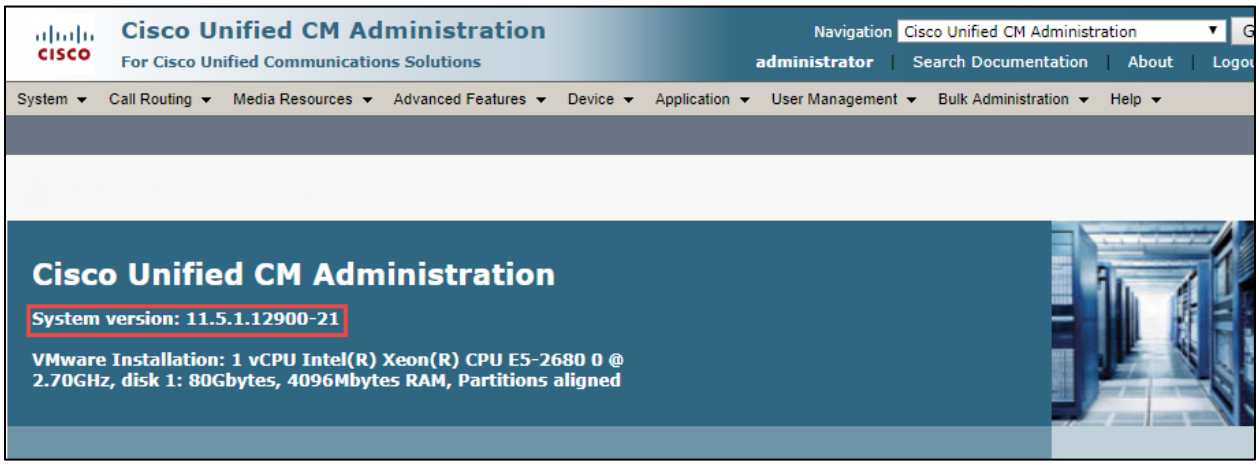


Figure 3: Cisco UCM Version

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

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

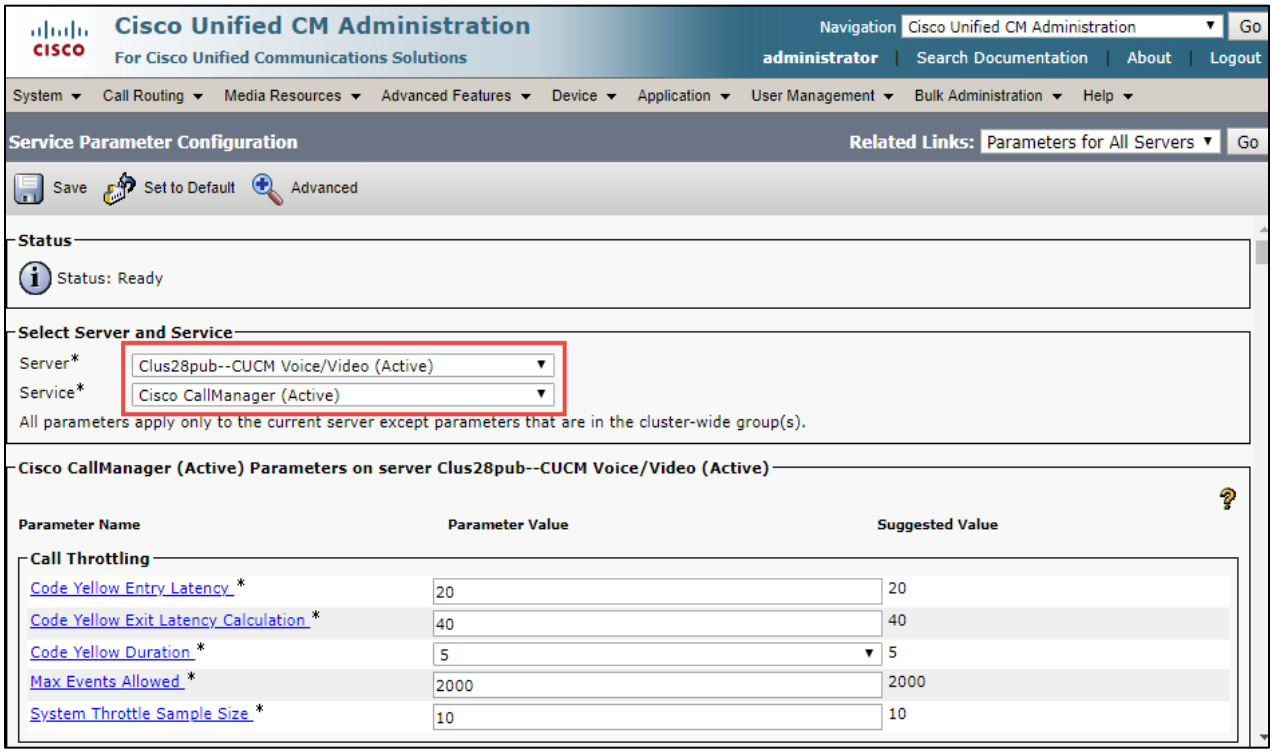
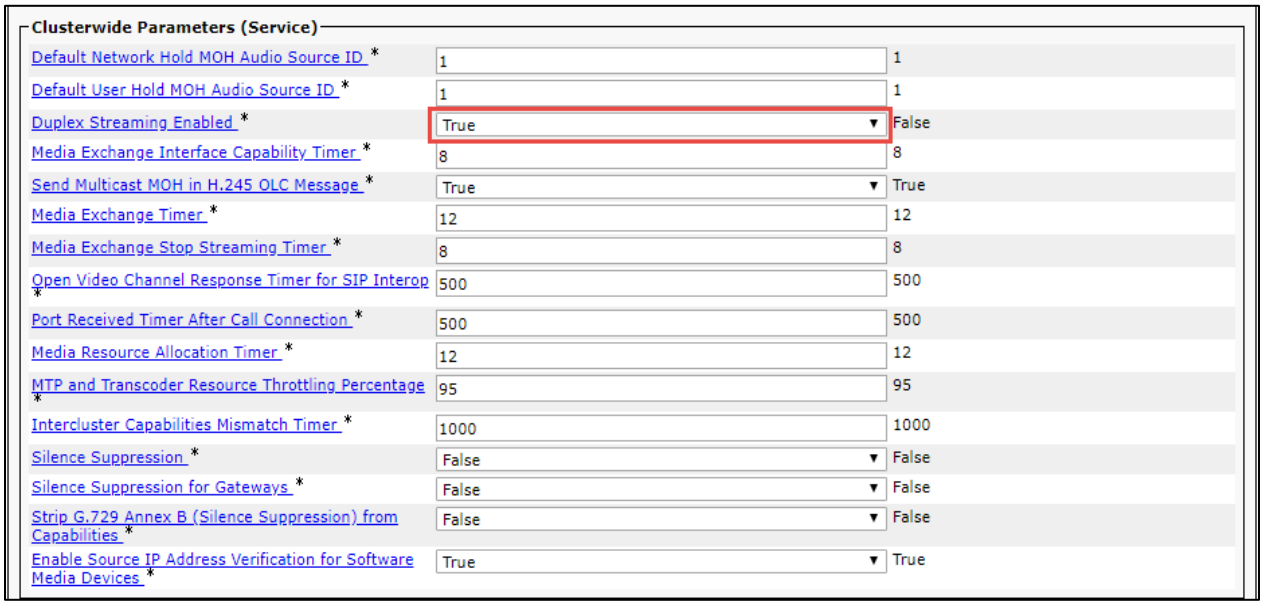


Figure 4: Service Parameters

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com



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 *	False	False
Silence Suppression for Gateways *	False	False
Strip G.729 Annex B (Silence Suppression) from Capabilities *	False	False
Enable Source IP Address Verification for Software Media Devices *	True	True

Figure 5: Service Parameters (Cont.)

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

- Name*= **Non Secure SIP Trunk Profile for JT** 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 = **TCP**

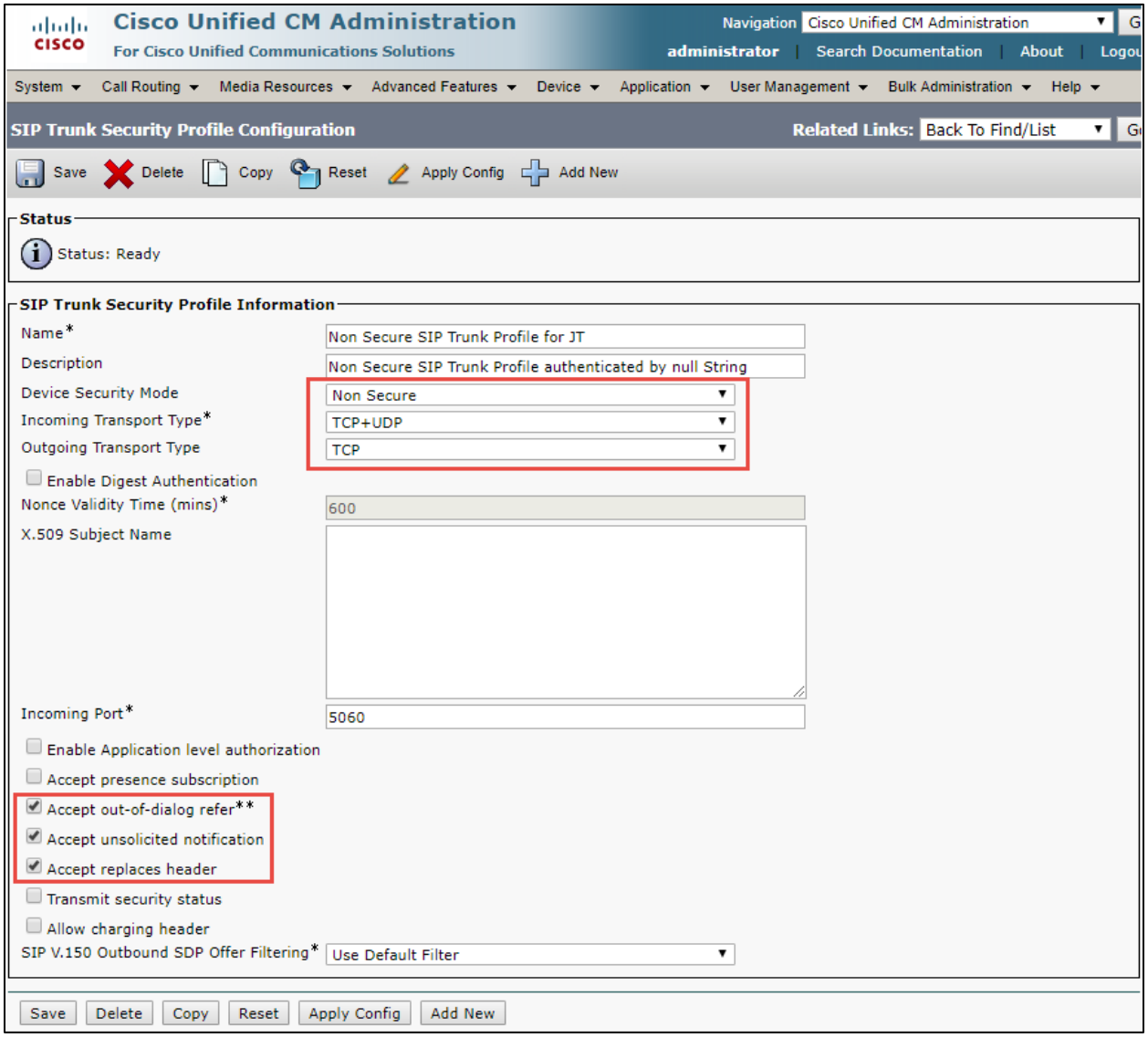


Figure 6: SIP Trunk Security Profile

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

SIP Profile

Navigation: Device → Device Settings → SIP Profile

- Name*= **JT Global SIP Profile** is used as an example
- Description = **JT Global SIP Profile** is used as an example

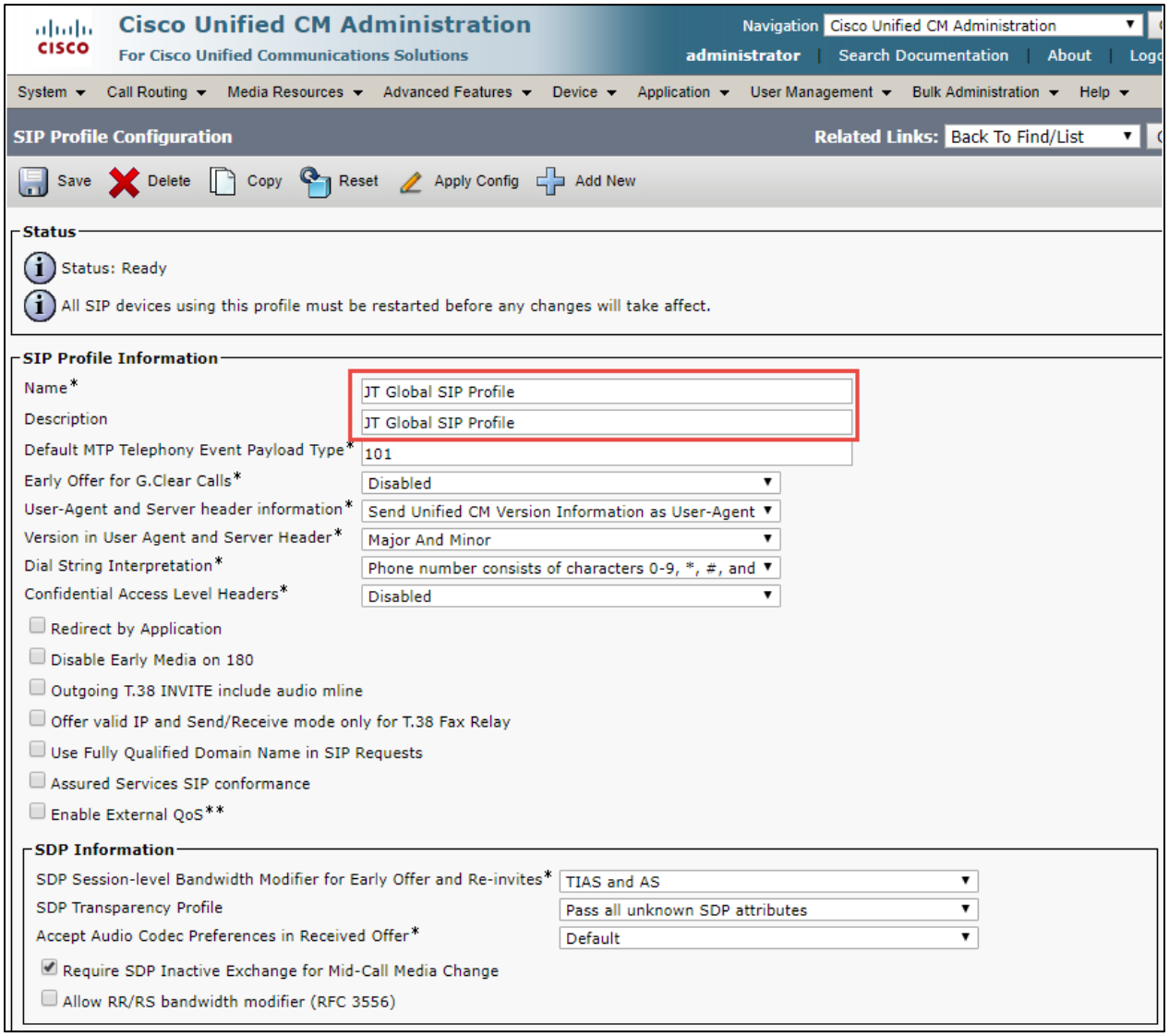


Figure 7: SIP Profile

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

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.)

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

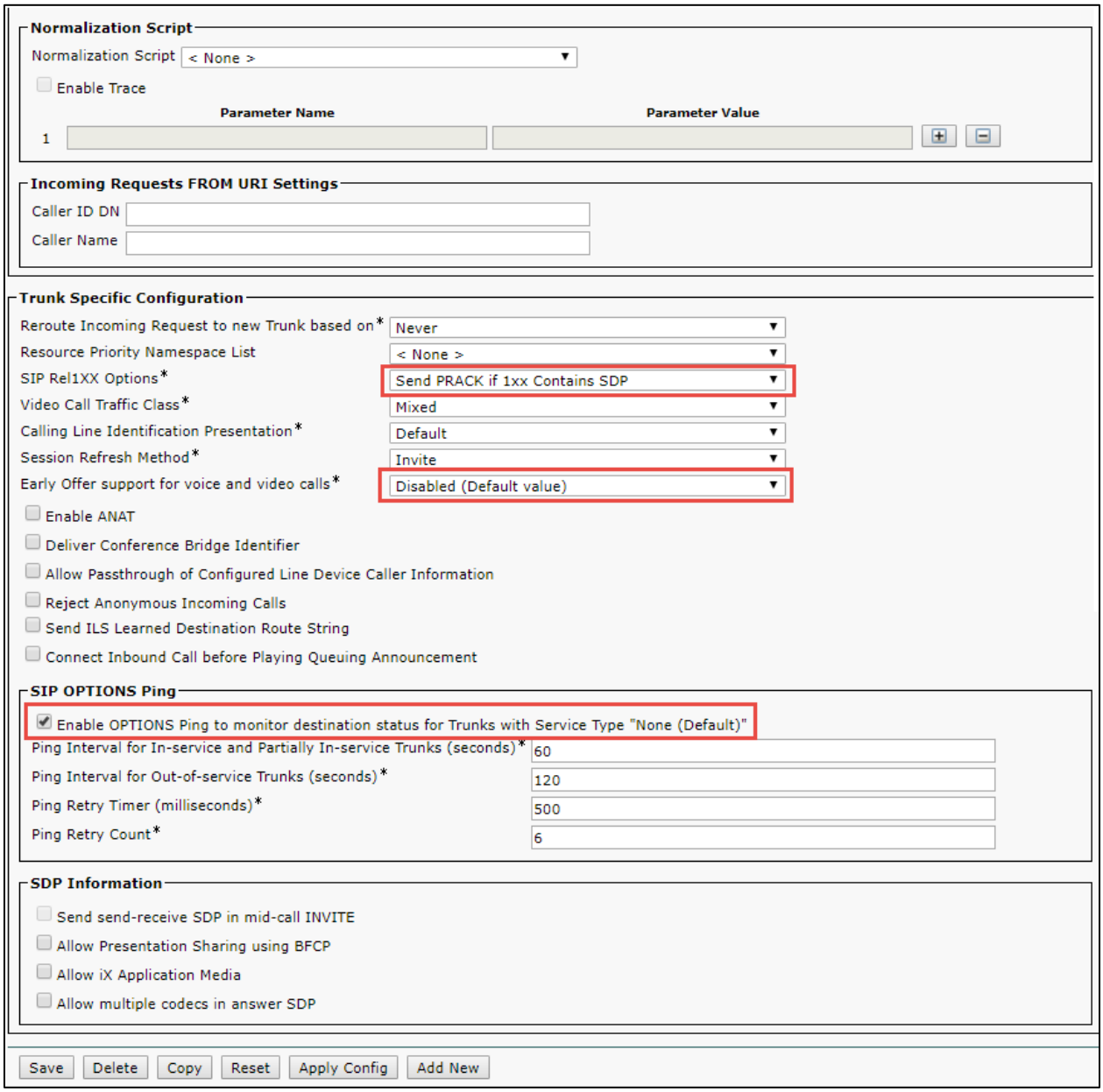


Figure 9: SIP Profile (Cont.)

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Trunk configuration

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

Navigation: Device → Trunk → Add New

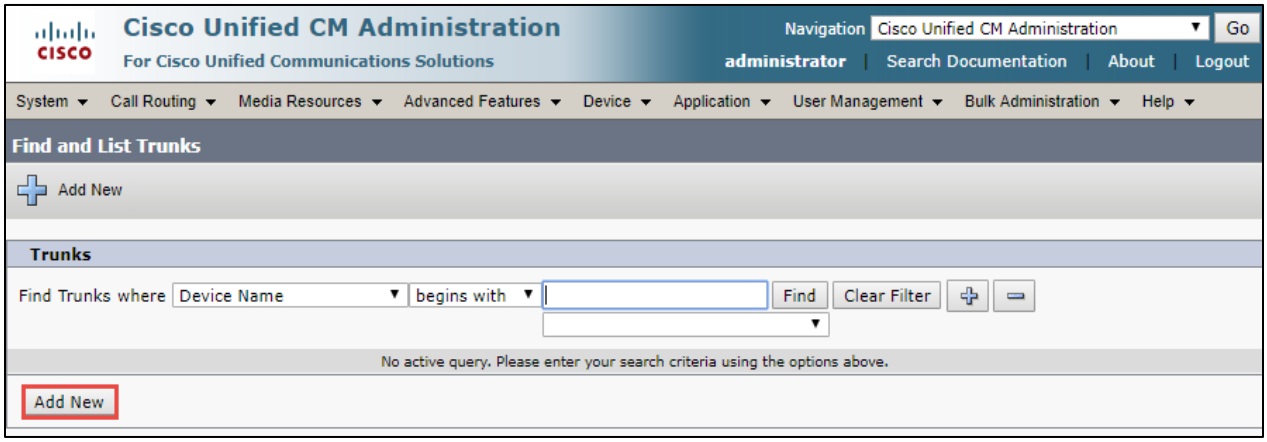


Figure 10: Add New Trunk to Cisco UBE

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

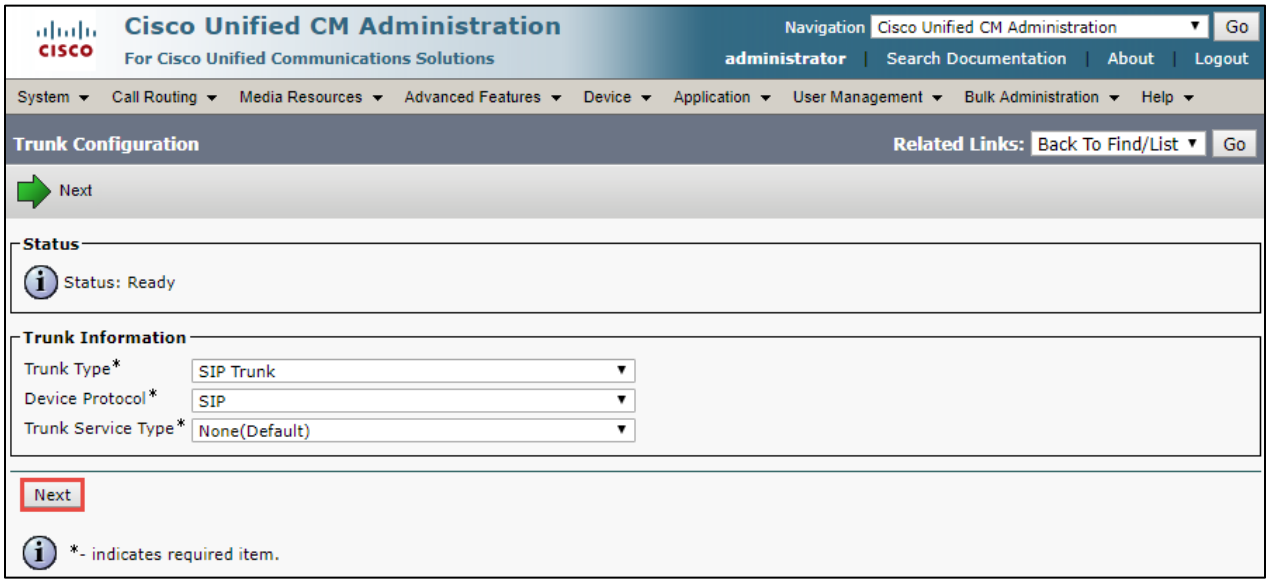


Figure 11: Add SIP Trunk Type

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

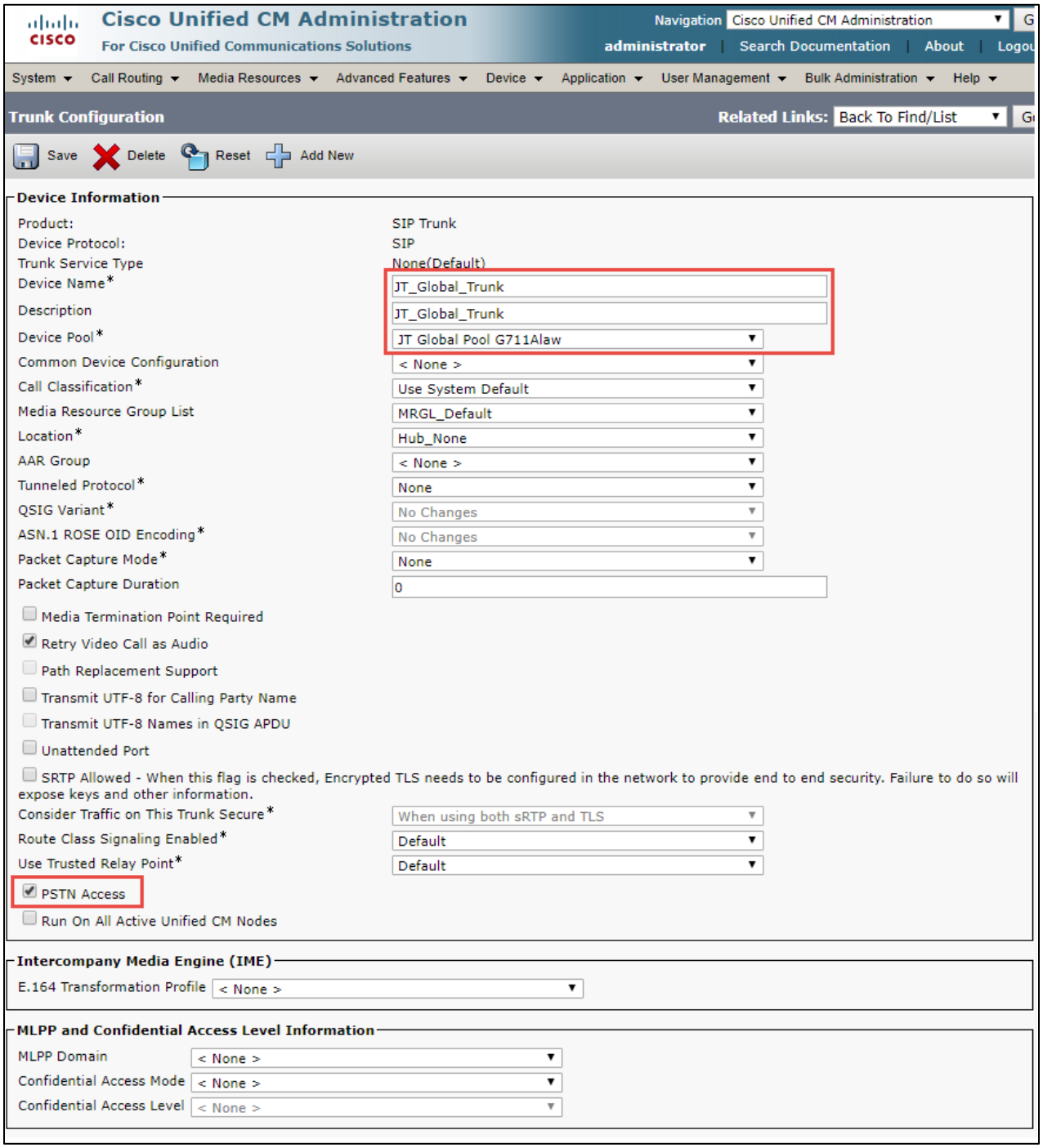


Figure 12: SIP Trunk to Cisco UBE

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

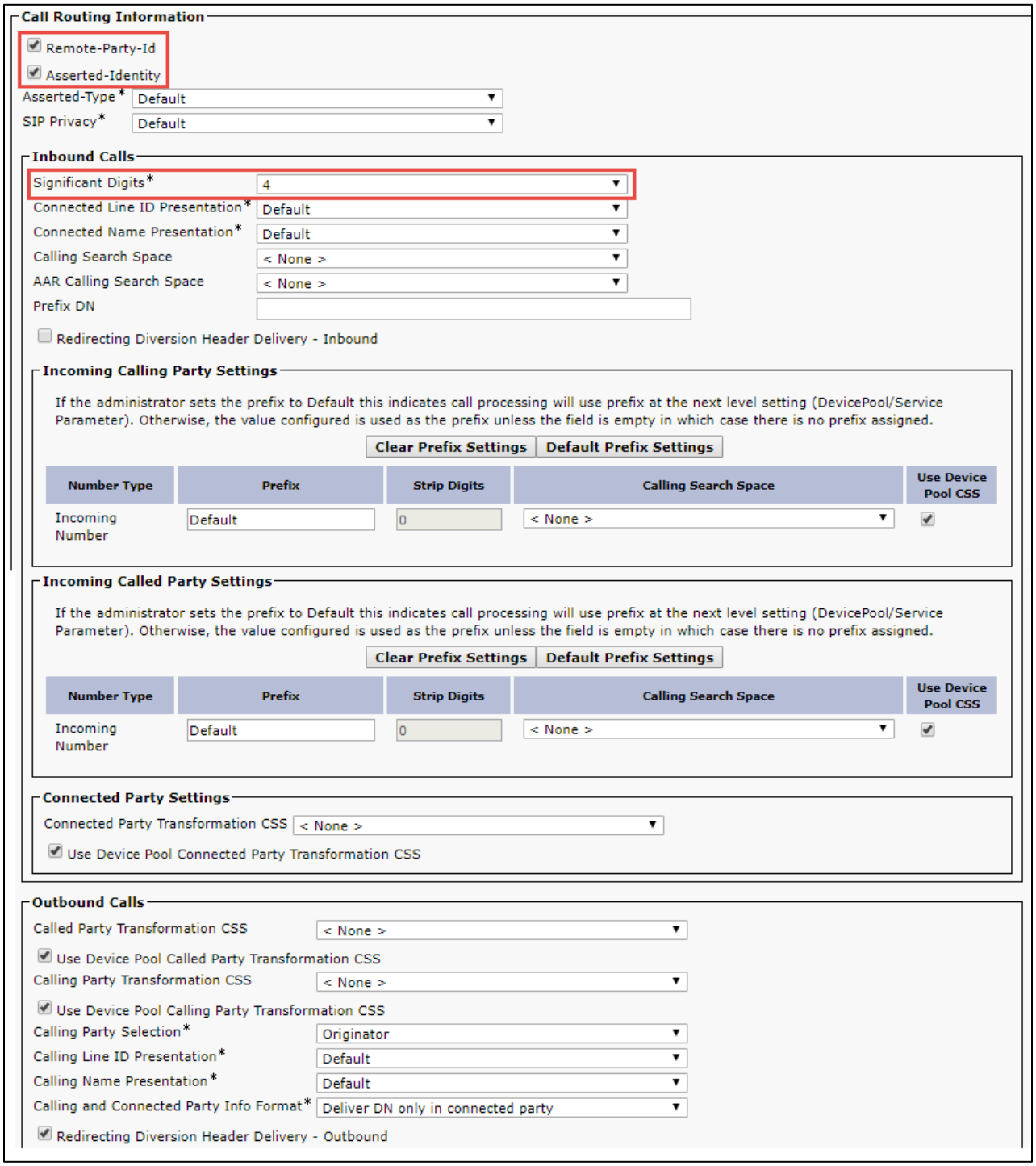


Figure 13: SIP Trunk to Cisco UBE (Cont.)

- Configure the Virtual LAN IP address of the Cisco UBE and the Destination Port

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

- 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

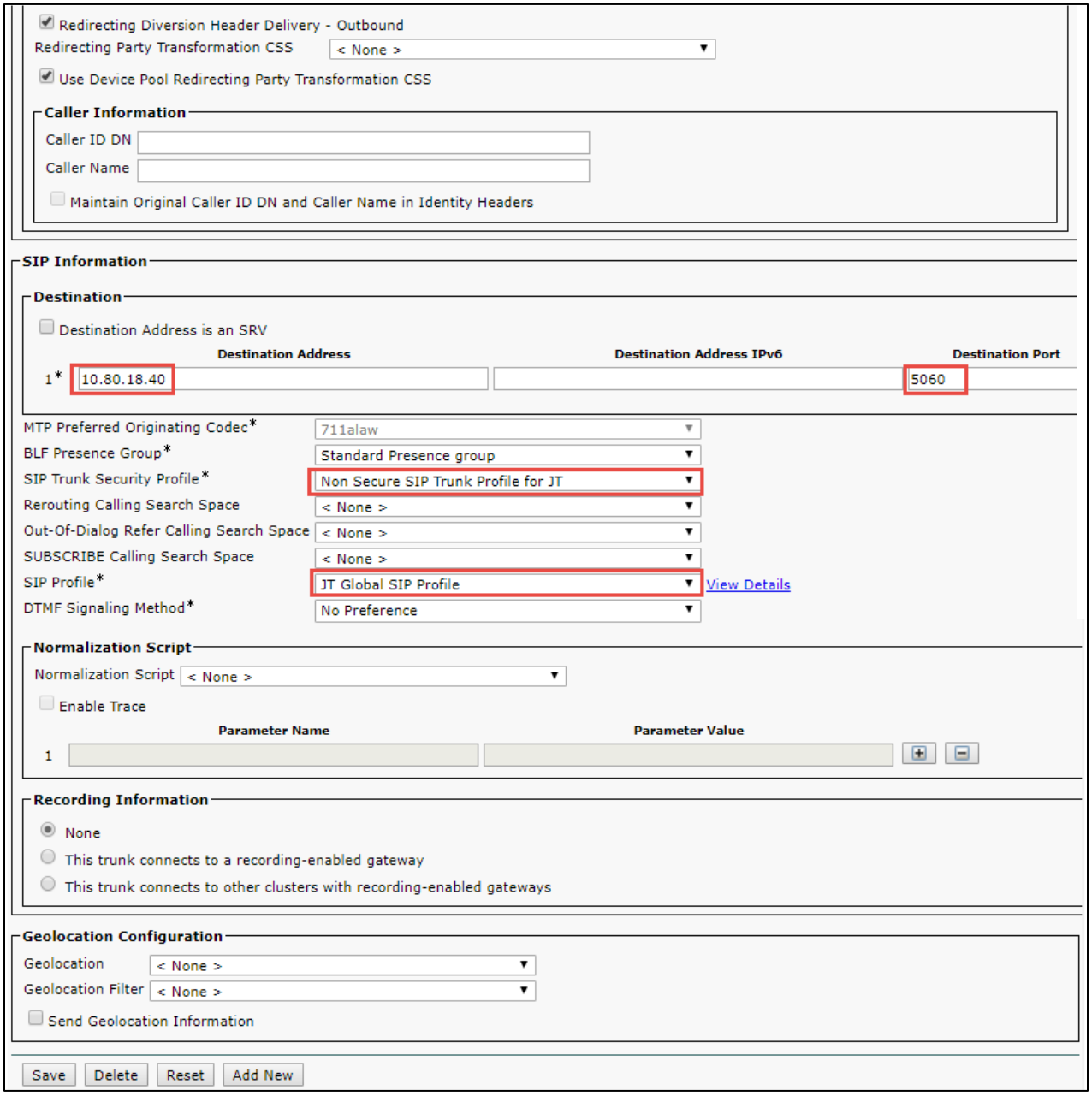


Figure 14: SIP Trunk to Cisco UBE (Cont.)

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Trunk configuration from Cisco UCM to Fax Gateway:

Navigation: Devices → Trunk → Add New

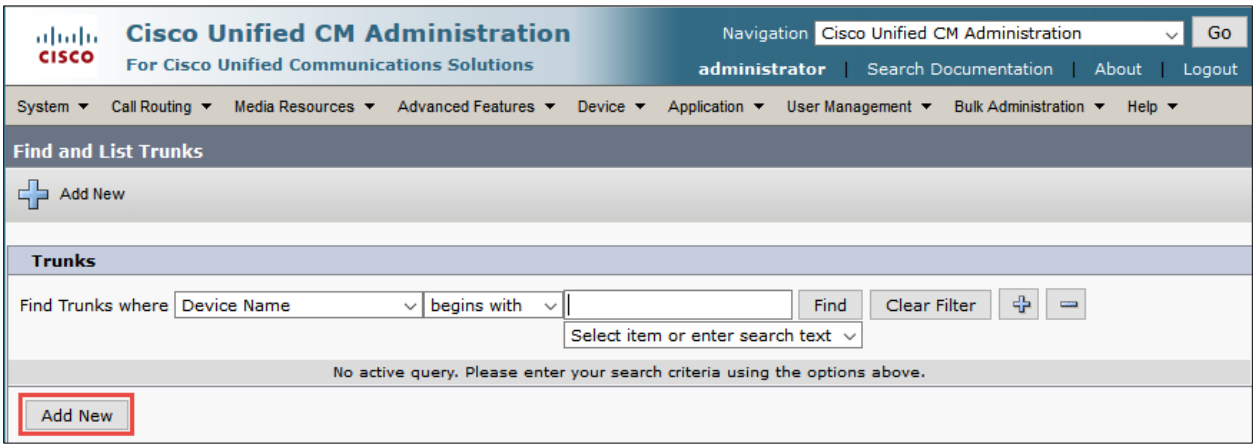


Figure 15: Add New Trunk to Fax Gateway

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

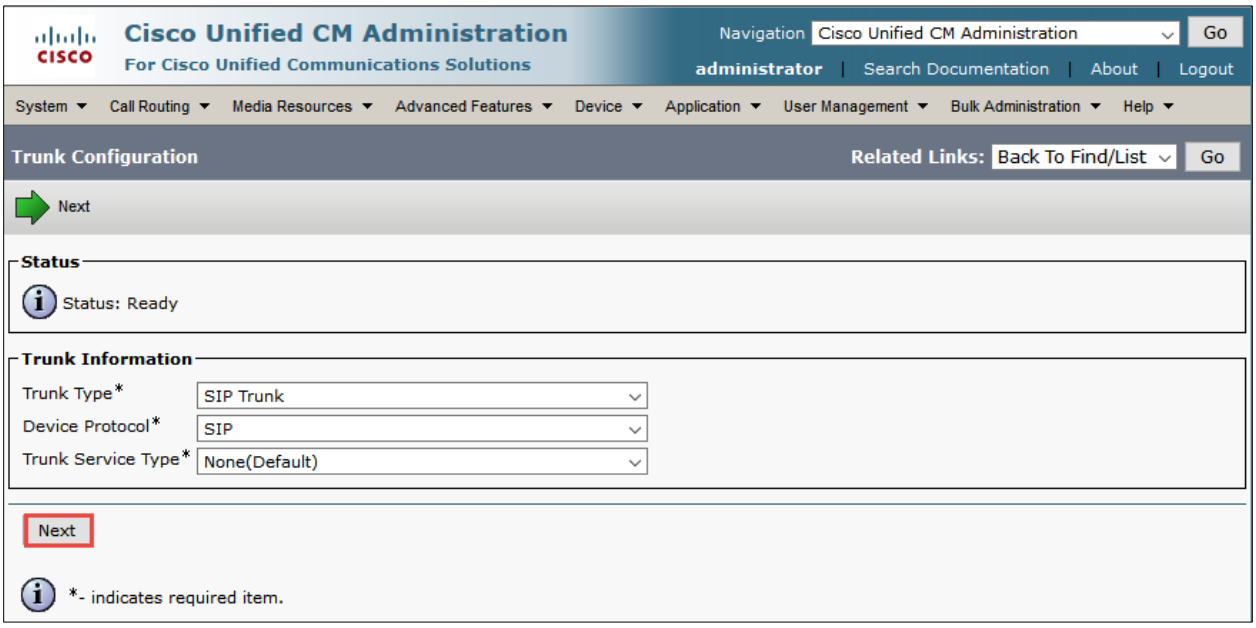


Figure 16: Add SIP Trunk Type

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration
administrator | Search Documentation | About | Log

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Trunk Configuration
Related Links: Back To Find/List

Save | Delete | Reset | Add New

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	JT_Fax_Gateway
Description	JT_Fax_Gateway
Device Pool*	JT Global Pool G711Alaw
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [ ] Path Replacement Support
- [ ] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU
- [ ] Unattended Port
- [ ] SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default

- [ ] PSTN Access
- [ ] Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

MLPP and Confidential Access Level Information

MLPP Domain	< None >
Confidential Access Mode	< None >
Confidential Access Level	< None >

Figure 17: SIP Trunk to FAX Gateway

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

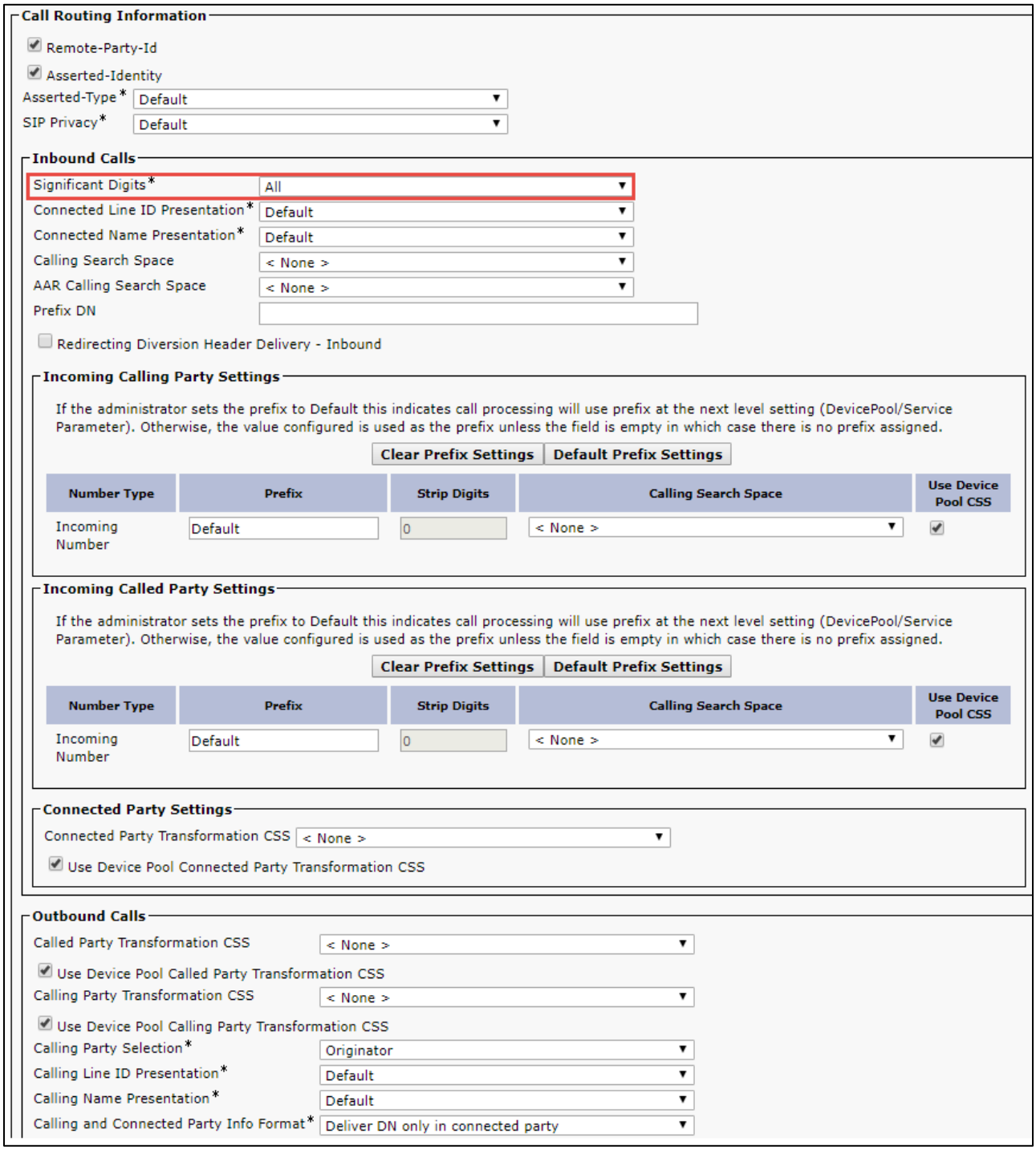


Figure 18: SIP Trunk to FAX Gateway (Cont.)

- Configure the IP address of Fax Gateway and the Destination Port
- Configure the SIP Trunk Security Profile and SIP Profile as shown below

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

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

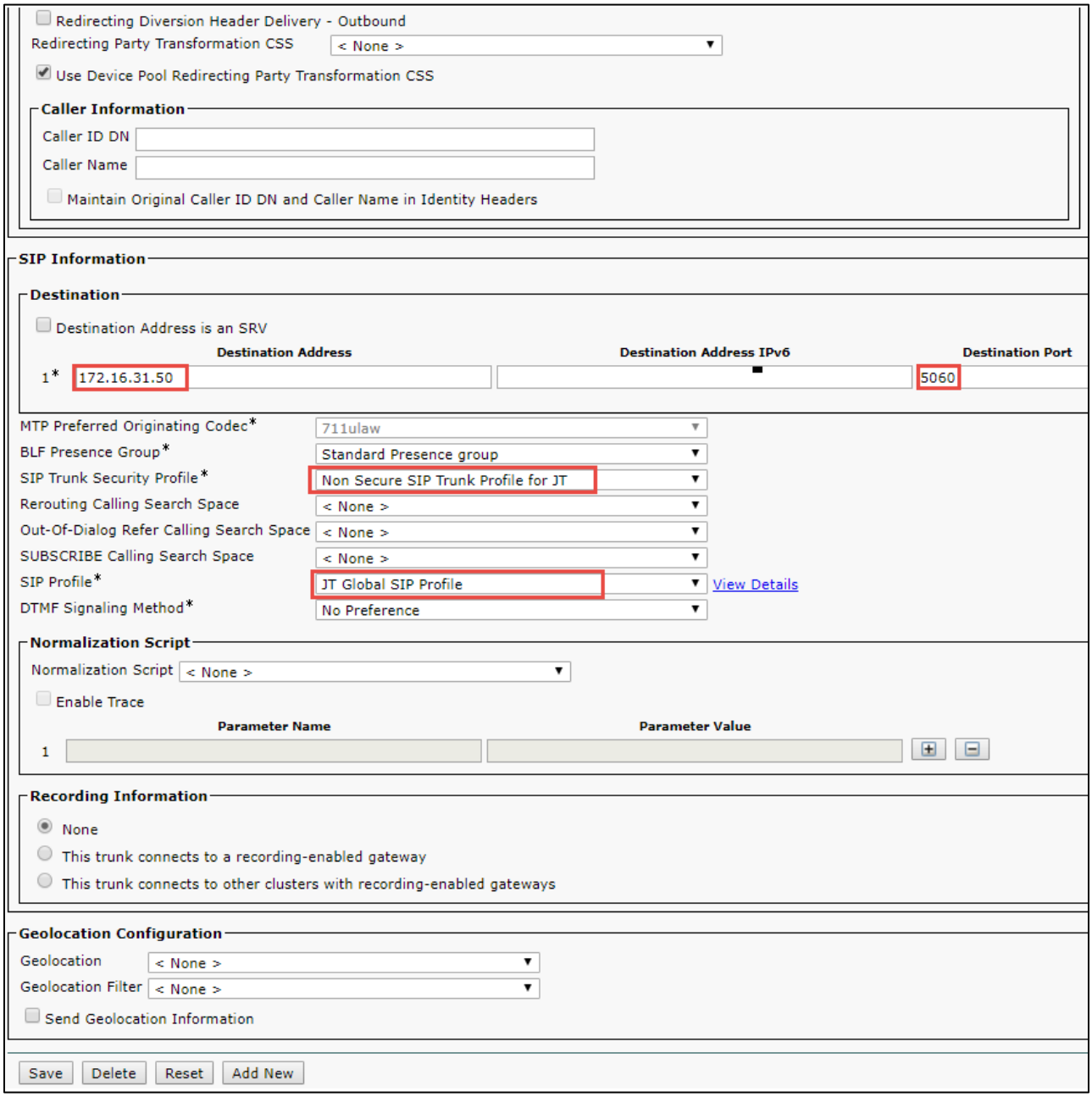


Figure 19: SIP Trunk to FAX Gateway (Cont.)

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Routing configuration

Route Pattern for Cisco UBE:

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

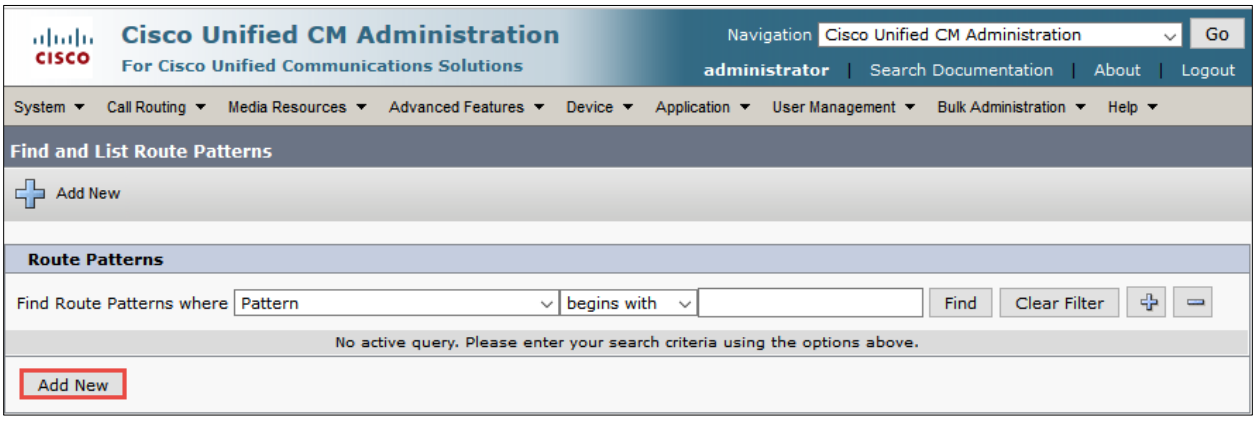


Figure 20: Add New Route Pattern for Cisco UBE

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

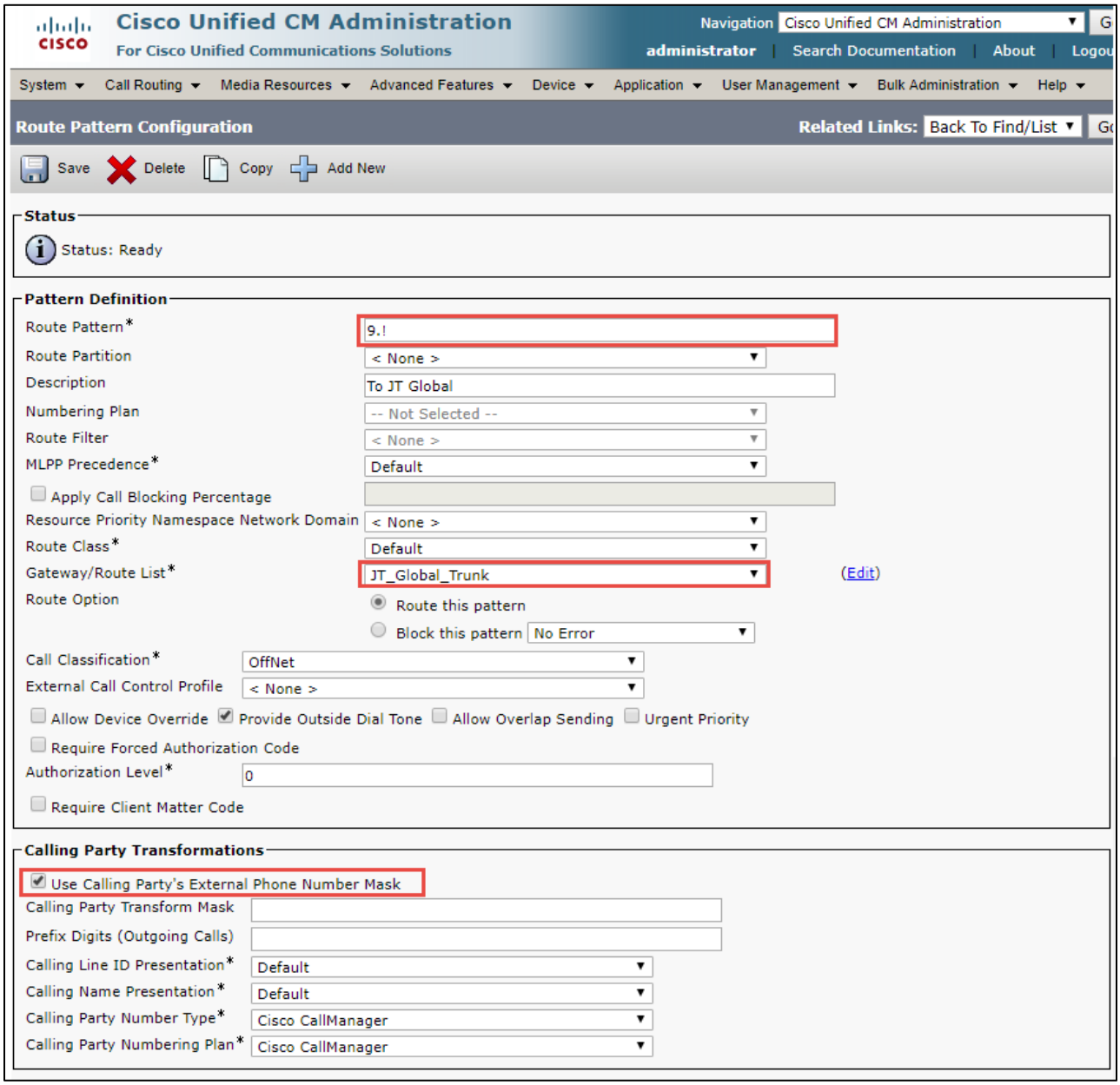


Figure 21: Route Pattern Configuration for Cisco UBE-PSTN Access

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

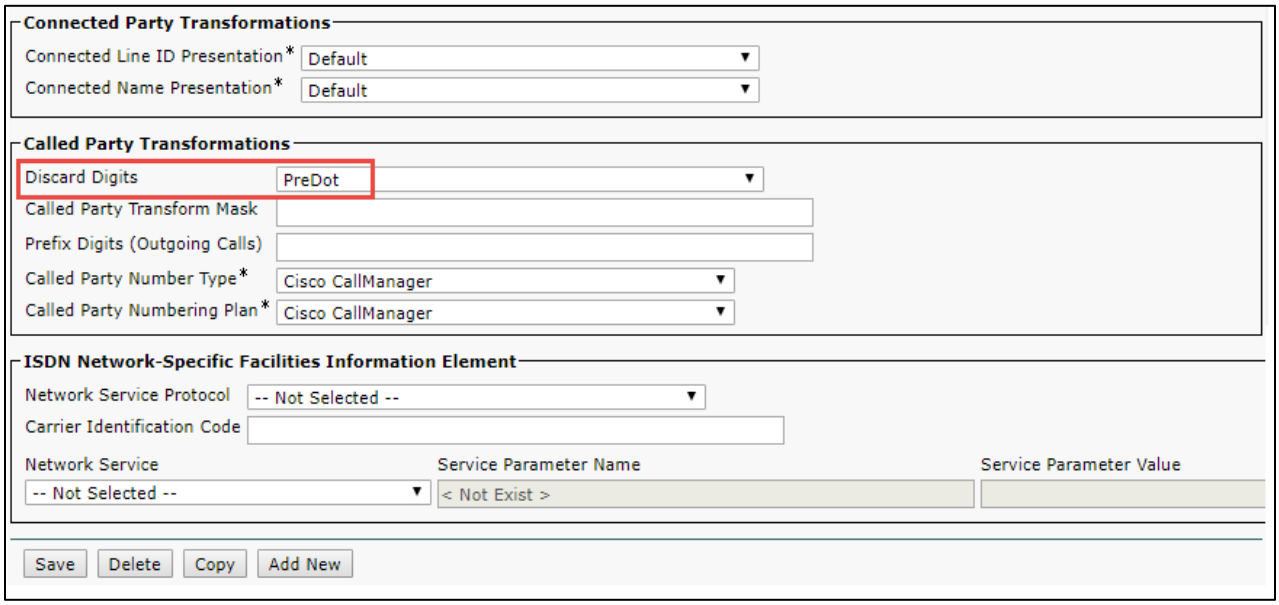


Figure 22: Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.)

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

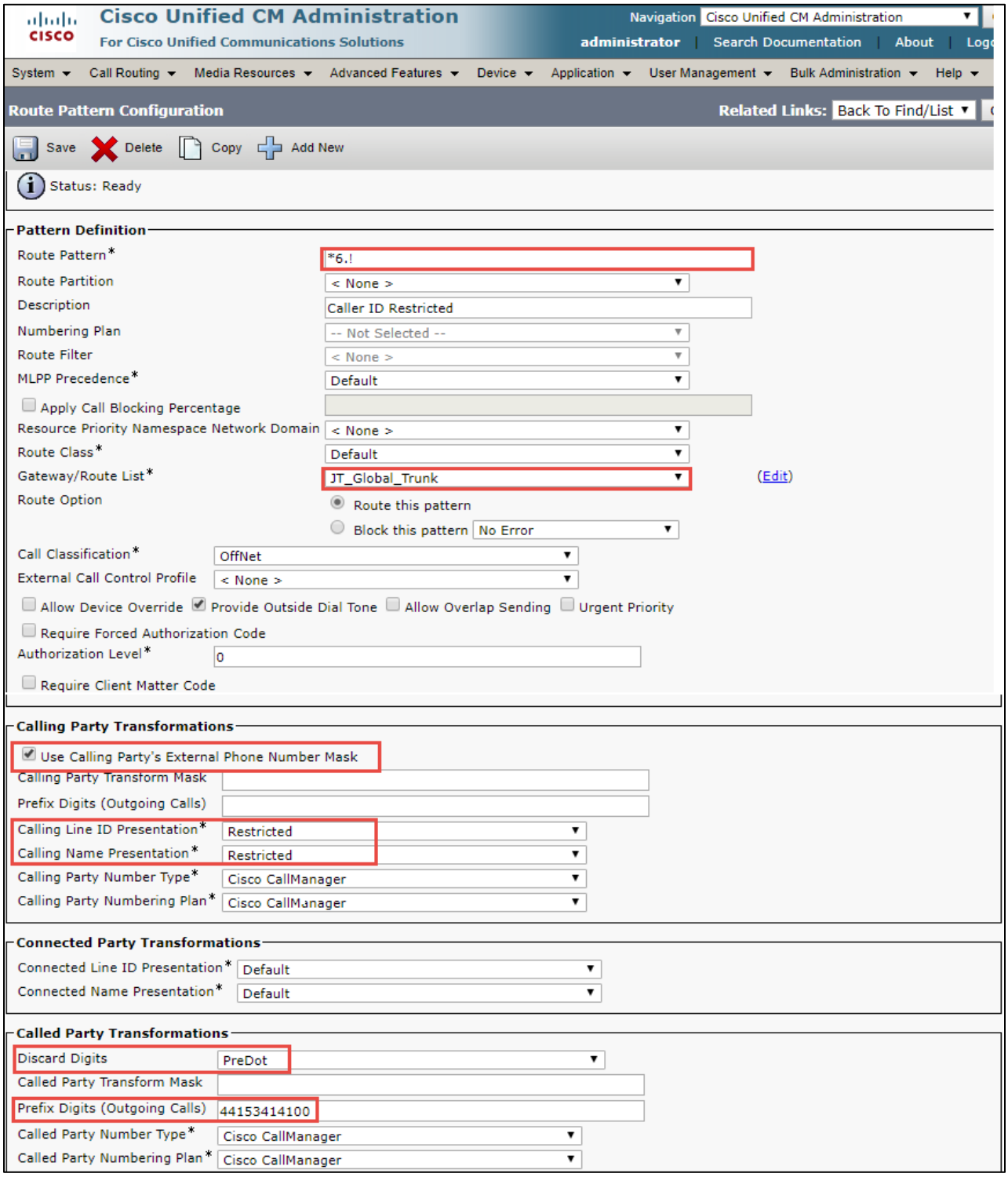


Figure 23: Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

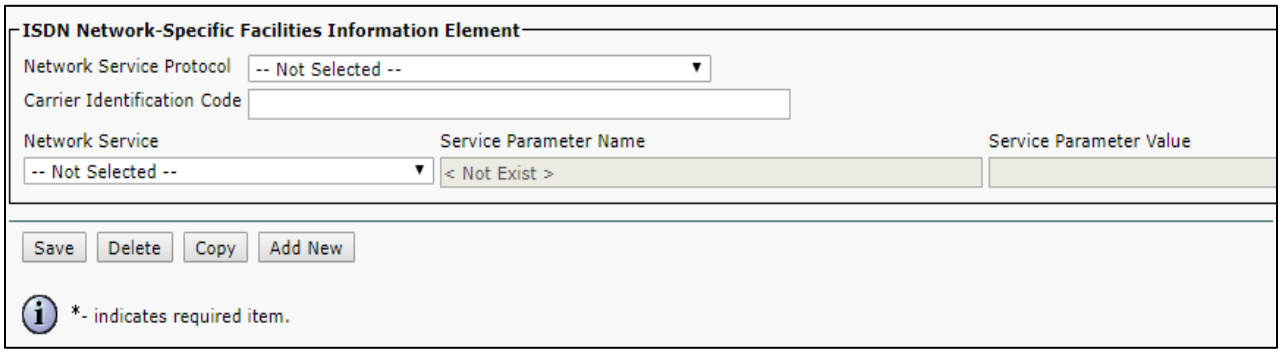


Figure 24: Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call (Cont.)

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Route Pattern for Fax Gateway:

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

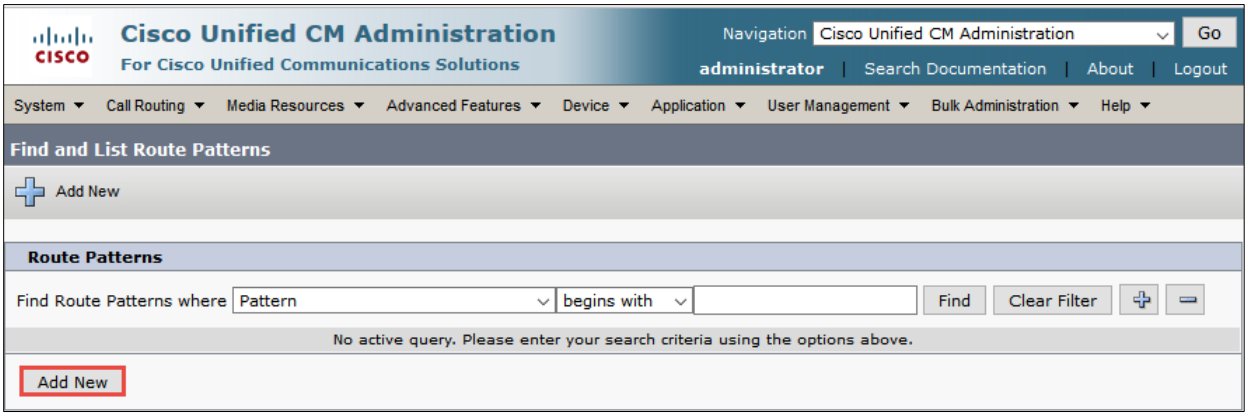


Figure 25: Add New Route pattern to Fax Gateway

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration
administrator | Search Documentation | About | Logou

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Route Pattern Configuration

Related Links: Back To Find/List

Save | Delete | Copy | Add New

Status

Status: Ready

Pattern Definition

Route Pattern*	6215
Route Partition	< None >
Description	JT Fax Gateway
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	JT_Fax_Gateway (Edit)
Route Option	Route this pattern / Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >

Allow Device Override | Provide Outside Dial Tone | Allow Overlap Sending | Urgent Priority

Require Forced Authorization Code

Authorization Level* 0

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Figure 26: Route Pattern Configuration for Fax Gateway

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

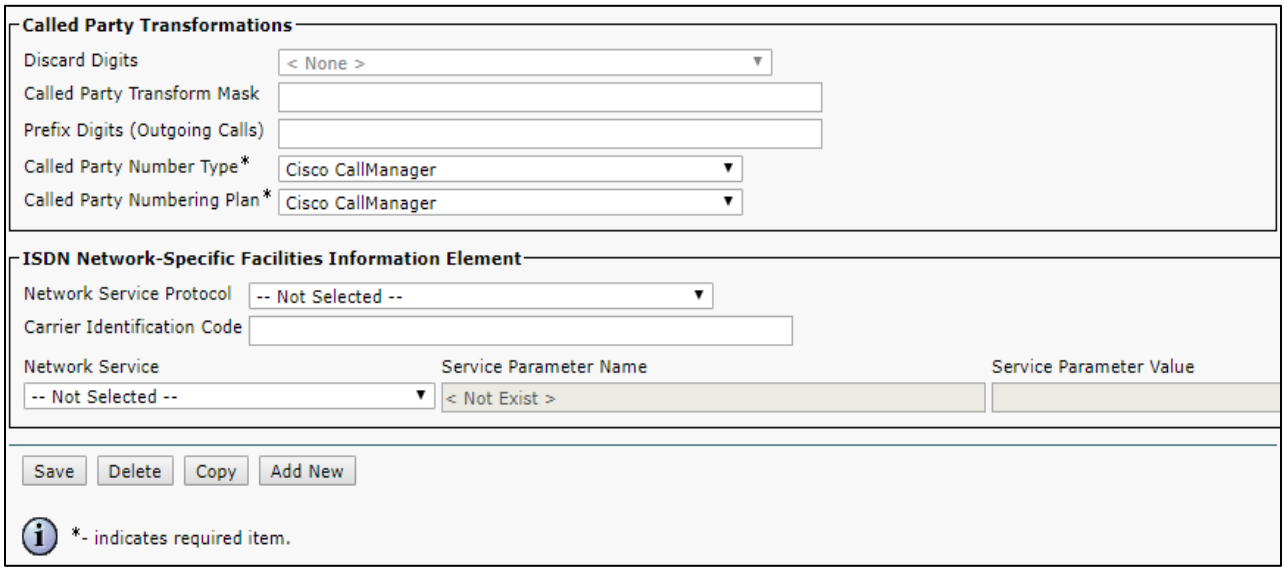


Figure 27: Route Pattern Configuration for Fax Gateway (Cont.)

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

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 pass-through g711alaw
 no fax-relay sg3-to-g3
 sip
  rel1xx disable
  midcall-signaling passthru
  g729 annexb-all
!
```

Codecs

G711alaw 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 g711alaw
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711alaw
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Dial peer

Outbound Dial-peer to Cisco UCM:

```
dial-peer voice 800 voip
 description Outbound Gateway to CUCM
 service session
 destination-pattern 9T
 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 disable
 fax protocol pass-through g711alaw
 no vad
```

Inbound Dial-peer from Cisco UCM:

```
dial-peer voice 900 voip
 description Inbound from JT Fax
 service session
 session protocol sipv2
 session transport udp
 incoming called-number 621.
 voice-class codec 1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

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 6215
 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 JT Global Fax
 station-id number +441534716215
 caller-id enable
!
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Configuration example

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

```
cme.in.tekvizion.com>en
Password:
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
!
!
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
!
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 pass-through g711alaw
 no fax-relay sg3-to-g3
 sip
  rel1xx disable
  midcall-signaling passthru
  g729 annexb-all
!
!
voice class codec 1
 codec preference 1 g711alaw
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711alaw
!
!
!
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
voice translation-rule 1
 rule 1 /\(^..........$\)/ /+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
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 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
ip route 0.0.0.0 0.0.0.0 172.16.29.1
!
ipv6 route ::/0 2620:96:C000:8::1
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
!
snmp-server community public RO
snmp-server location Chennai
!
!
!
!
ipv6 access-list ipv6
 permit ipv6 any any
!
control-plane
!
!
voice-port 0/0/0
!
voice-port 0/0/1
 no echo-cancel enable
 no vad
 cptone IN
 station-id name JT Global Fax
 station-id number +441534716215
 caller-id enable
!
voice-port 0/3/0:15
!
voice-port 0/2/0:15
!
voice-port 0/1/0
!
voice-port 0/1/1
 no echo-cancel enable
 no vad
 cptone IN
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 station-id name JT Global Fax 2
 station-id number +441534716212
 caller-id enable
!
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 45 pots
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 huntstop
 service session
 destination-pattern 6215
 no digit-strip
 port 0/0/1
 forward-digits all

dial-peer voice 46 pots
 service session
 destination-pattern 6212
 no digit-strip
 port 0/1/1
 forward-digits all
!
!
dial-peer voice 800 voip
 description Outbound Gateway to CUCM
 service session
 destination-pattern 9T
 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 disable
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 900 voip
 description Inbound from JT Fax
 service session
 session protocol sipv2
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 session transport udp
 incoming called-number 621.
 voice-class codec 1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
!
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 Administrator password tekV1z10n
 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
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

```
 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#
```

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.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

Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS. IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES

© 2018 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters
CiscoSystems International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

AsiaPacific Headquarters
Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

Cisco Systems has more than 200 offices in the following countries and regions. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at http://www.cisco.com/go/offices.

Argentina • Australia • Austria • Belgium • Brazil • Bulgaria • Canada • Chile • China PRC • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE • Finland • France • Germany • Greece • Hong Kong SAR • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia • Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Scotland • Singapore • Slovakia • Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey Ukraine • United Kingdom • United States • Venezuela • Vietnam • Zimbabwe

© 2016 Cisco Systems, Inc. All rights reserved.
CCENT, Cisco Lumin, Cisco Nexus, the Cisco logo and the Cisco Square Bridge logo are trademarks of Cisco Systems, Inc.; Changing the Way We Work, Live, Play, and Learn is a service mark of Cisco Systems, Inc.; and Access Registrar, Aironet, BPX, Catalyst, CCDA, CCDP, CCVP, CCIE, CCIP, CCNA, CCNP, CCSP, Cisco, the Cisco Certified Internetwork Expert logo, Cisco IOS, Cisco Press, Cisco Systems, Cisco Systems Capital, the Cisco Systems logo, Cisco Unity, EtherFast, EtherSwitch, Fast Step, Follow Me Browsing, FormShare, GigaDrive, HomeLink, Internet Quotient, IOS, iPhone, iQ Expertise, the iQ logo, iQ Net Readiness Scorecard, iQuick Study, LightStream, Linksys, Meeting Place, MGX, Networking Academy, Network Registrar, Packet, PIX, ProConnect, ScriptShare, SMARTnet, StackWise, The Fastest Way to Increase Your Internet Quotient, and TransPath are registered trademarks of Cisco Systems, Inc. and/or its affiliates in the United States and certain other countries.
All other trademarks mentioned in this document or Website are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (0705R)
Printed in the USA