Application Note

Integra Telecom SIP Trunking: Connecting Cisco Unified Communications Manager 8.5(1) via the Cisco Unified Border Element using SIP

June 3, 2011

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Note: Testing was conducted in tekVizion lab.

Introduction

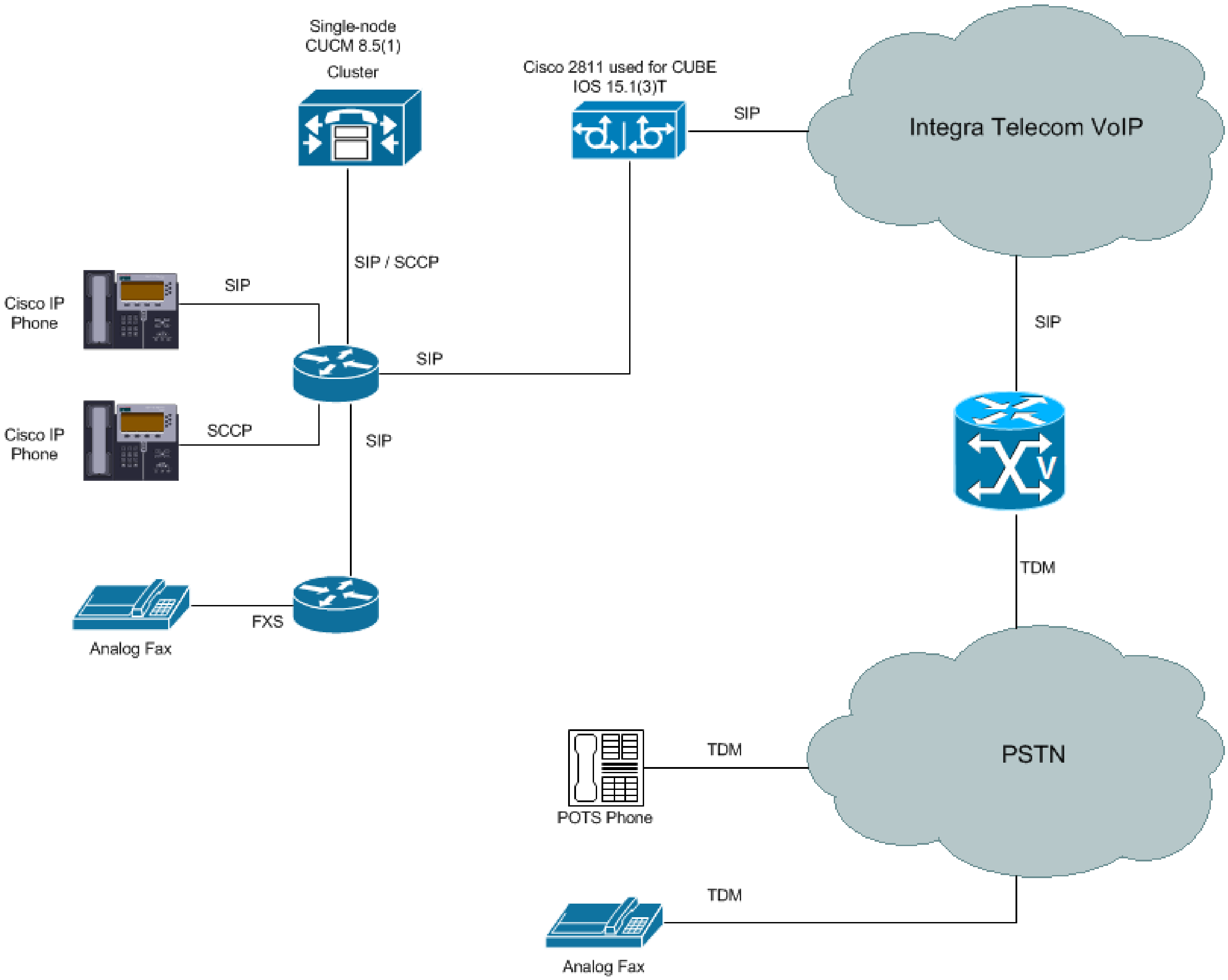
Service Providers today, such as Integra Telecom, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. Integra Telecom SIP Trunking is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either Analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element provides demarcation, security, interworking and session management services.

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 8.5(1) with a Cisco Unified Border Element (Cisco UBE) for connectivity to the Integra Telecom SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 8.5(1)/Cisco UBE) to PSTN via Integra Telecom SIP Trunking. This document does not address 911 emergency outbound calls. For 911 feature service details contact Integra Telecom, directly.
- Testing was performed in accordance to Cisco's Service Provider SIP Trunk Validation Test Plan and all features were verified. Key features verified are:
  - Basic Calls
  - Basic Calls with Calling Name and Number as allowed or restricted
  - DTMF Relay
  - Call Conference (Intra-site, PSTN)
  - Call Transfer (Blind, Attended, Early Attended)
  - Hold and Resume
  - Voice Mail
  - T.38 Fax G3/SG3
  - Simultaneous Calls
  - Auto Attendant
  - International Calls
  - G.711 Fax G3/SG3
  - Call Forwarding – Find Me (Unconditional, Busy, No Reply)
  - Codec negotiation
  - Dial Plans
  - PRACK with SDP early-media cut-through
- The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Integra Telecom's SIP network and Cisco Unified Communications. The configuration described in this document details the important commands to have enabled for interoperability to be successful and care must be taken, by the network administrator deploying Cisco UBE, to ensure these commands are set per each dial-peer requiring to interoperate to Integra Telecom's SIP network.
- This application note does not cover the use of calling search spaces (CSS) or partitions on Cisco Unified Communications Manager. To understand and learn how to apply CSS and partitions refer to the cisco.com link below: http://www.cisco.com/en/US/partner/docs/voice_ip_comm/cucm/admin/8_0_2/ccmsys/a03ptcss.html

Note: Testing was conducted at tekVizion Labs®.

Network Topology

Figure 1. Lab Network Topology



System Components

Hardware Components

- Cisco 2811 (used as Cisco Unified Border Element)
- Cisco 3845 with VIC-4FXS/DID (used for analog fax and hardware MTP)
- Cisco Unified Communications Manager (single-node cluster consisting of one Cisco MCS 7800 Series server)
- Cisco IP Phones

Note: Testing was conducted at tekVizion Labs®.

Software Requirements

- Cisco Unified Communications Manager 8.5.1.10000-26
- Cisco Unified Border Element, IOS version 15.1(3)T (c2800nm-spservicesk9-mz.151-3.T.bin)
- Cisco 3845 router, IOS version 15.1(3)T (c3845-adventerprisek9-mz.151-3.T.bin)

Note: Testing was conducted at tekVizion Labs®.

Features

Features Supported

- Voice calls using G.729 and G.711 codecs
- RFC 3261 support
- Early media cut-through with DTMF relay before 200 OK
- Calling number presentation / restriction
- Call conferencing
- Call transfer (attended and unattended)
- Call hold and resume
- Call forwarding
- DTMF relay (RFC 2833)
- T.38 fax
- G.711 pass-through fax

Features Not Supported

- Caller ID update via SIP UDATE method

Caveats

- When a PSTN to CPE call is transferred by the CPE to a second PSTN number, the outgoing call leg fails due to CUBE bug in which CUBE fails to offer the correct codec list to the service provider.

Note: Testing was conducted at tekVizion Labs®.

Call Flows

In the sample configuration presented here, CUCM is provisioned with four-digit directory numbers corresponding to the last four DID digits. No digit manipulation is performed on the CUBE.

For incoming PSTN calls, the CUBE presents the full ten-digit DID number to CUCM. A CUCM Translation Pattern strips all but the last four digits and routes the call based on those digits. Voice calls are routed to IP phones; fax calls are routed via a 4-digit route pattern to a SIP trunk that terminates on the router hosting the analog fax endpoint.

CPE callers make outbound PSTN calls by dialing a “90” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, the router hosting that endpoint delivers all digits including the “90” prefix. A “90.@” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the CUBE.

Figure 2. Outbound Voice Call

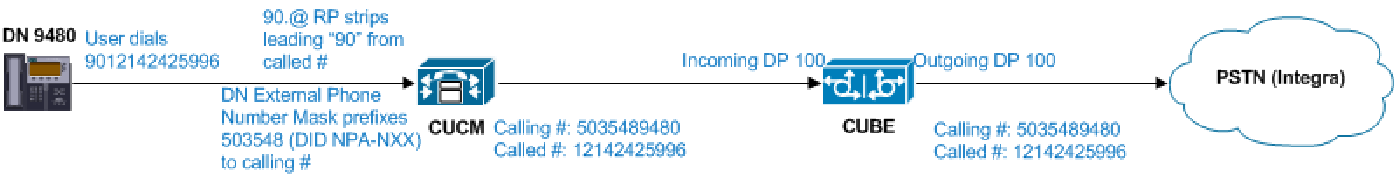


Figure 3. Inbound Voice Call

Figure 4. Outbound fax call

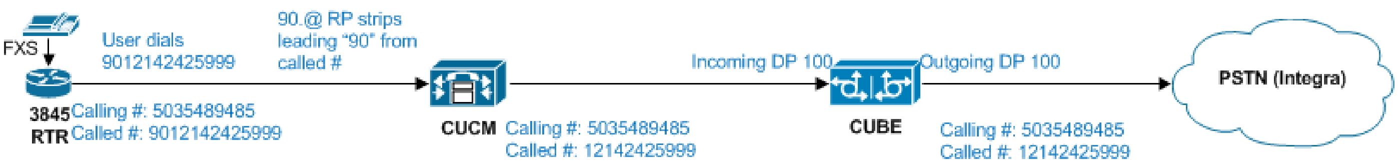
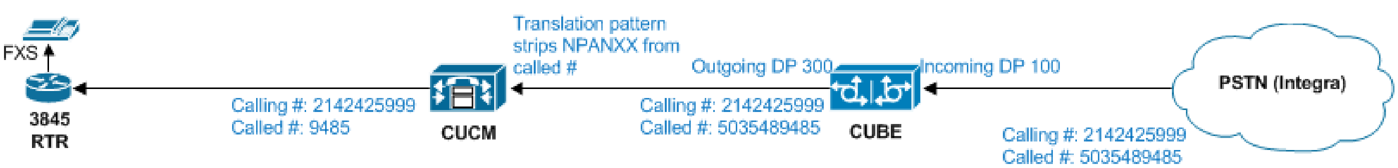


Figure 5. Inbound fax call



Note: Testing was conducted at tekVizion Labs®.

Configuration

Configuring Cisco Unified Border Element

Critical commands are marked in **Bold** with footnotes at the bottom of the page

Version Information:

```
Cisco IOS Software, 2800 Software (C2800NM-SPSERVICESK9-M), Version 15.1(3)T, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2010 by Cisco Systems, Inc.
Compiled Mon 15-Nov-10 21:41 by prod_rel_team

ROM: System Bootstrap, Version 12.4(13r)T11, RELEASE SOFTWARE (fc1)

MTPnetB uptime is 7 weeks, 4 days, 11 hours, 38 minutes
System returned to ROM by power-on
System image file is "flash:c2800nm-spservicesk9-mz.151-3.T.bin"
Last reload type: Normal Reload


This product contains cryptographic features and is subject to United
States and local country laws governing import, export, transfer and
use. Delivery of Cisco cryptographic products does not imply
third-party authority to import, export, distribute or use encryption.
Importers, exporters, distributors and users are responsible for
compliance with U.S. and local country laws. By using this product you
agree to comply with applicable laws and regulations. If you are unable
to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

If you require further assistance please contact us by sending email to
export@cisco.com.

Cisco 2811 (revision 53.51) with 512000K/12288K bytes of memory.
Processor board ID FTX1411ALT7
2 FastEthernet interfaces
DRAM configuration is 64 bits wide with parity enabled.
239K bytes of non-volatile configuration memory.
126000K bytes of ATA CompactFlash (Read/Write)


License Info:

License UDI:

-------------------------------------------------
Device#   PID                   SN
-------------------------------------------------
*0        CISCO2811             FTX1411ALT7


Configuration register is 0x2102
```


Note: Testing was conducted at tekVizion Labs®.

Running Configuration:

```
version 15.1
service config
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname MTPnetB
!
boot-start-marker
boot system flash c2800nm-spservicesk9-mz.151-3.T.bin
boot-end-marker
!
!
logging buffered 51200 warnings
no logging console
!
aaa new-model
!
!
!
!
!
!
!
aaa session-id common
clock timezone CST -6 0
clock summer-time CDT recurring
!
dot11 syslog
ip source-route
!
!
ip cef
!
!
!
ip domain name lab.tekvizion.com[1]
ip name-server 10.64.1.3[2]
no ipv6 cef
multilink bundle-name authenticated
!
!
!
!
!
!
voice rtp send-recv[3]
!
voice service voip
 ip address trusted list[4]
  ipv4 207.173.7.100
  ipv4 10.70.18.2
 allow-connections sip to sip[5]
```

[1] Optional. The domain name must match the CUCM enterprise parameter "Cluster Fully Qualified Domain Name" and must resolve to a DNS A-record. See "DNS Configuration" and "CUCM Configuration" below

[2] IP address of DNS server

[3] Establish a 2-way voice path when a Real Time Protocol channel is opened

[4] Required for toll fraud prevention. IP addresses listed here are those of trusted VoIP peers.

```
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw[6]
 sip
  early-offer forced[7]
  midcall-signaling passthru[8]
!
voice class codec 1[9]
 codec preference 1 g729br8
 codec preference 2 g729r8
 codec preference 3 g711ulaw
!
!
!
!
!
voice-card 0
!
crypto pki token default removal timeout 0
!
!
!
!
license udi pid CISCO2811 sn FTX1411ALT7
username cisco privilege 15 password 0 cisco
!
!
!
!
!
!
!
interface FastEthernet0/0
 ip address 10.70.18.15 255.255.255.0
 duplex full
 speed 100
!
interface FastEthernet0/1
 description This IP address is associated with IntegraTelecom.
 ip address 174.46.0.150 255.255.255.128
 duplex auto
 speed auto
!
ip forward-protocol nd
no ip http server
no ip http secure-server
!
!
ip route 0.0.0.0 0.0.0.0 174.46.0.129
!
logging esm config
!
!
!
!
control-plane
!
!
```

---

[5] Allow SIP to SIP call processing.
[6] Enable T.38 fax relay. Remaining parameters are optional; this example disables redundancy for both low-speed T.30 messaging and high-speed page data, and enables G.711 fallback if T.38 negotiation fails.
[7] Configures CUBE to send a SIP INVITE with SDP on an outbound call leg (Delayed Offer to Early Offer)
[8] Enable support for SIP re-INVITE supplementary services
[9] Codec preference list, referenced from the dial-peers with the "voice-class codec *n*" configuration command

Note: Testing was conducted at tekVizion Labs®.

```
!
!
mgcp profile default
!
sccp local FastEthernet0/0
!
dial-peer voice 100 voip
 description 1+10 digits to IntegraTelecom
 destination-pattern ^1[2-9]..[2-9]......$[10]
 session protocol sipv2
 session target sip-server[11]
 voice-class codec 1[12]
 voice-class sip early-offer forced[13]
 dtmf-relay rtp-nte[14]
!
dial-peer voice 102 voip
 description International Calls to IntegraTelecom
 destination-pattern ^011T[15]
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
!
dial-peer voice 300 voip
 description 503548XXXX calls to CUCM
 huntstop
 destination-pattern ^503548....$[16]
 session protocol sipv2
 session target dns:clus3pubsub.lab.tekvizion.com[17]
 voice-class codec 1
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
!
dial-peer voice 20 voip
 description all calls from CUCM
 huntstop
 session protocol sipv2
 session target dns:clus3pubsub.lab.tekvizion.com
 incoming called-number .T[18]
 voice-class codec 1
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
!
dial-peer voice 10 voip
 description calls from Service Provider
 session protocol sipv2
 session target sip-server
 incoming called-number .T
 voice-class codec 1
```

[10] Outbound dial peer for 1+10-digit called numbers
[11] Sip-server resolves to the Integra Telecom SBC IP address, and is defined in the sip-ua section below.
[12] Reference to codec preference list “voice class codec 1”, above
[13] Enable delayed-offer to early-offer
[14] Forward DTMF tones by using RTP with the Named Telephone Event (NTE) payload type (RFC 2833).
[15] Outbound dial peer for international calls (US market)
[16] Outbound dial peer (to CUCM). The pattern here should match service provider-assigned DID numbers.
[17] Reference to DNS A record. In a single-subscriber deployment, this could either be a DNS reference, or it could be an IP address: “ipv4:xxx.xxx.xxx.xxx”. In a multiple-node cluster, this would be a reference to a DNS SRV record.
[18] This is the default inbound dial peer

Note: Testing was conducted at tekVizion Labs®.

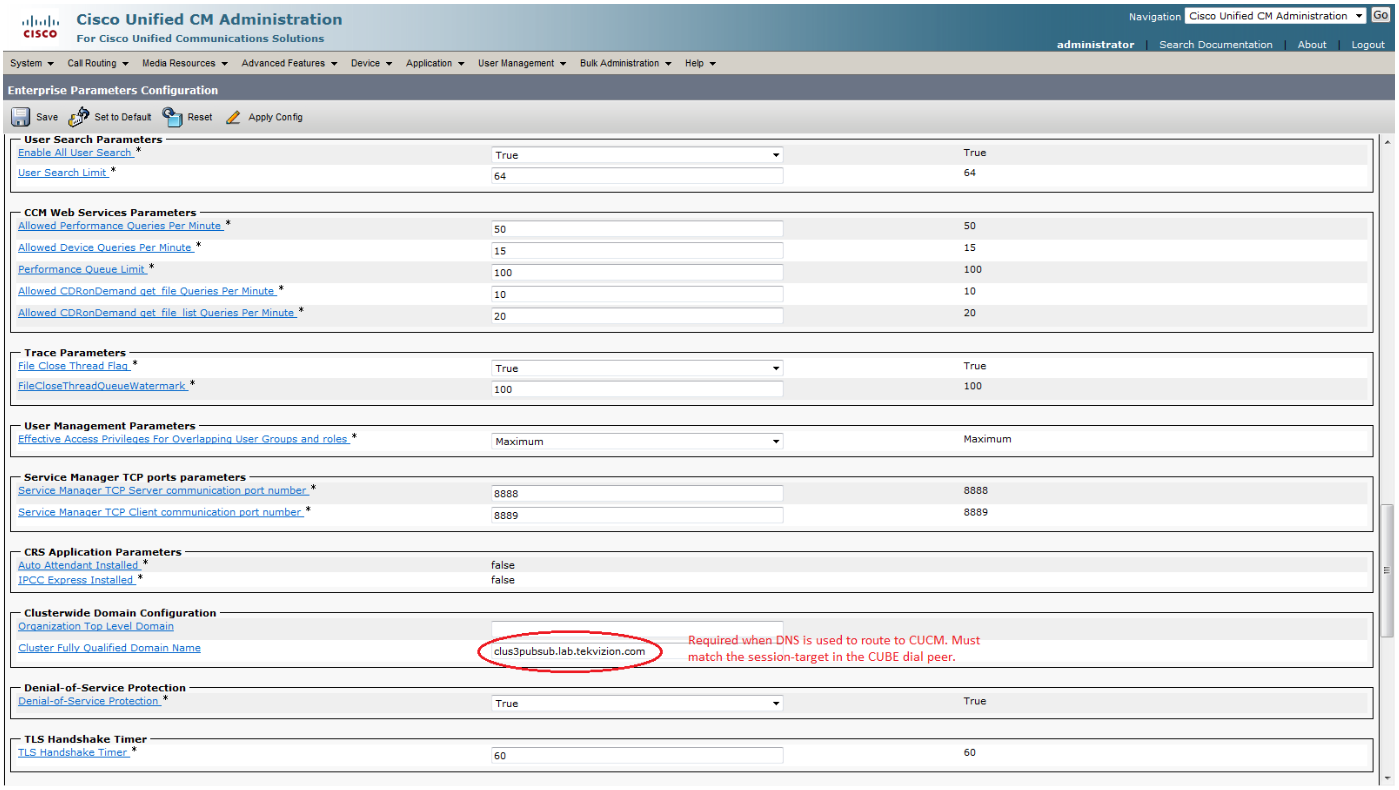
```
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
!
!
gateway
 timer receive-rtp 1200
!
sip-ua
 authentication username 5035489480 password 7 131504411E08543A733C3C60603642[19]
 retry invite 7
 retry bye 7
 retry cancel 7
 retry register 10
 sip-server ipv4:207.173.7.100:5060[20]
!
!
!
line con 0
 exec-timeout 0 0
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 0 0
 password *********
 transport input all
!
scheduler allocate 20000 1000
end
```

[19] Authentication credentials provided by service provider
[20] Service provider signaling address.

Note: Testing was conducted at tekVizion Labs®.

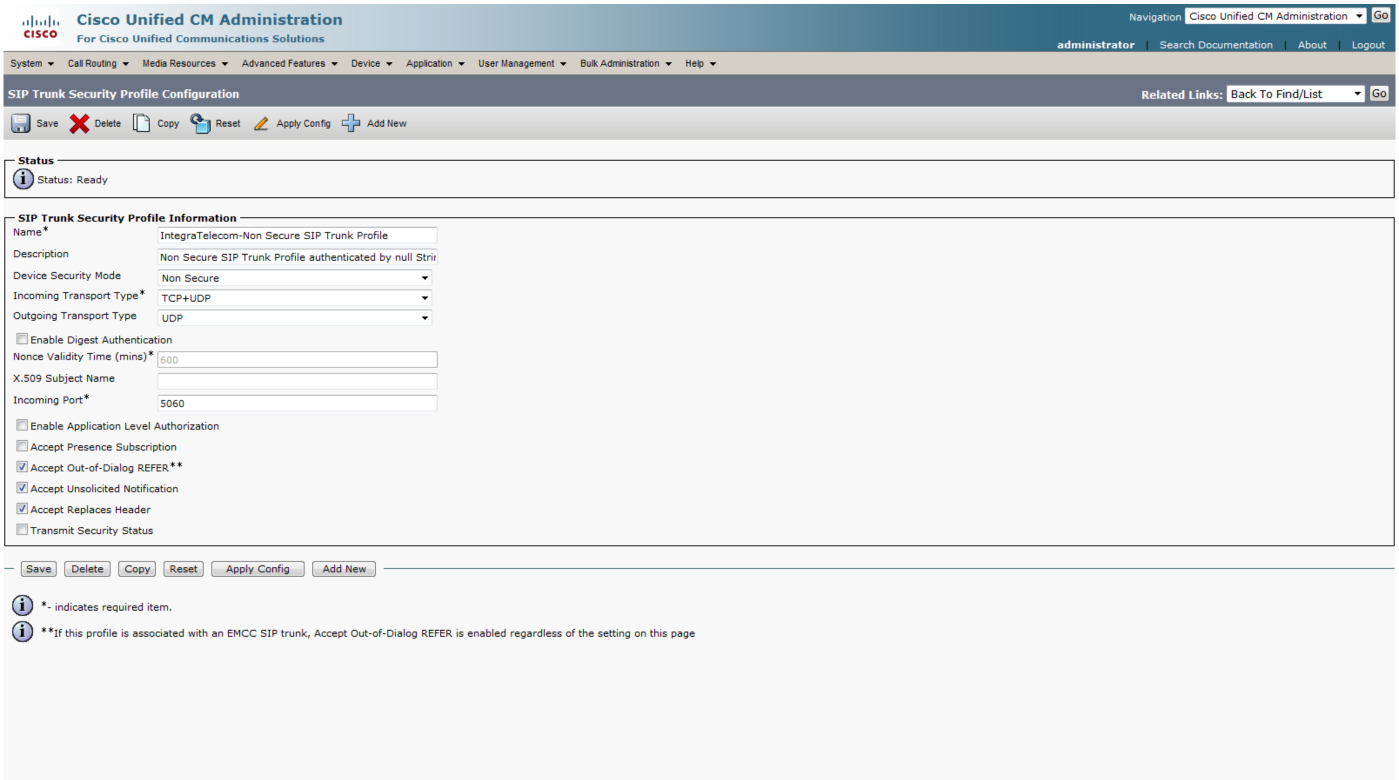
Configuring the Cisco Unified Communications Manager

Figure 6. Enterprise Parameters



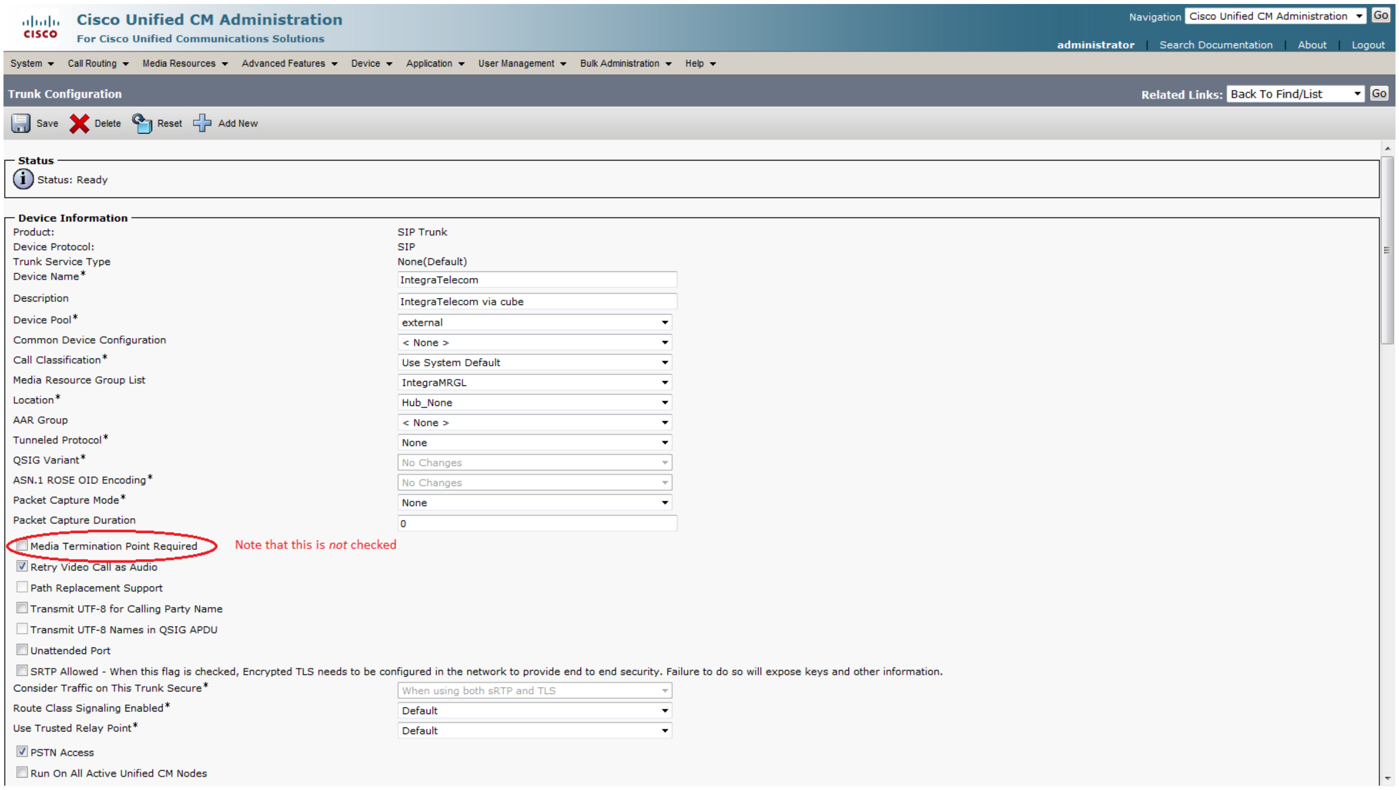
Note: Testing was conducted at tekVizion Labs®.

Figure 7. SIP Trunk Security Profile



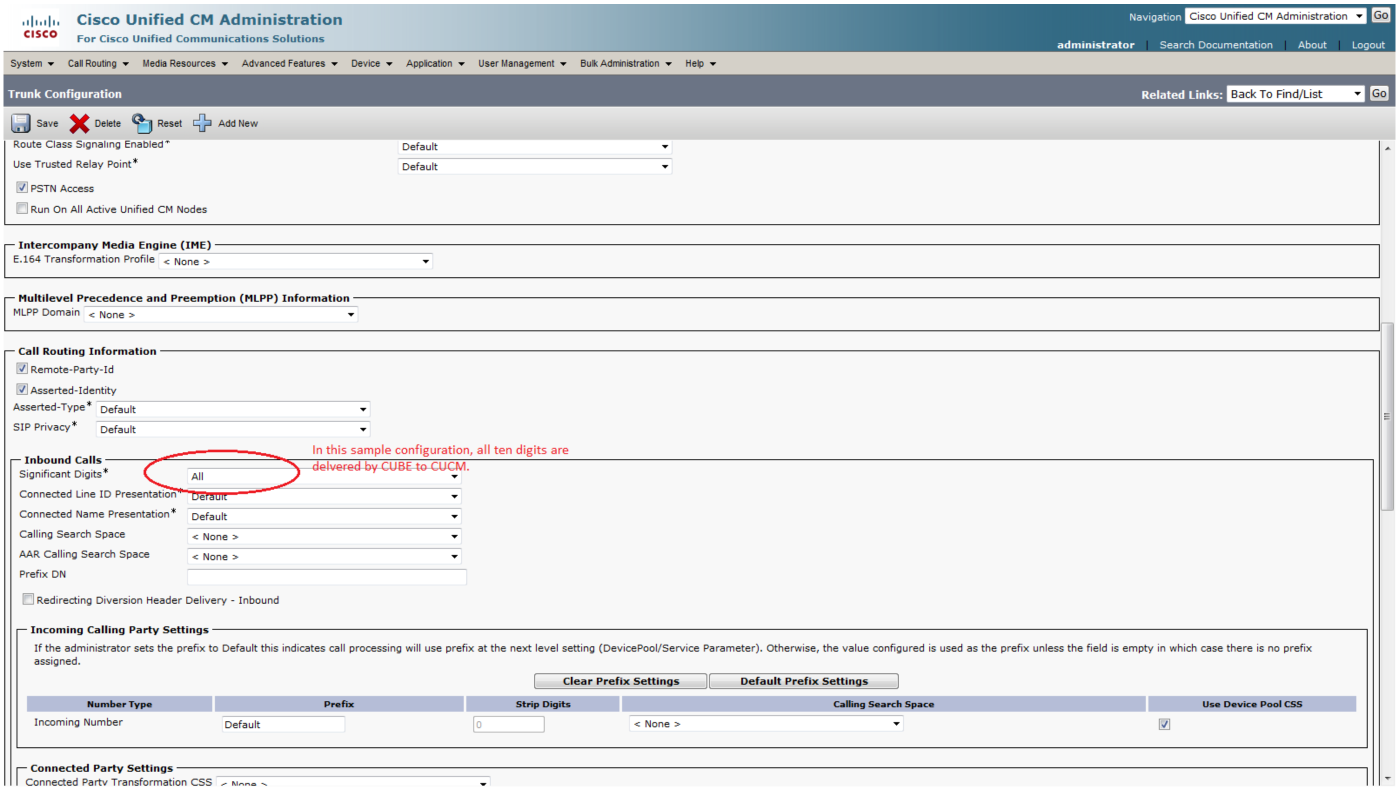
Note: Testing was conducted at tekVizion Labs®.

Figure 8. SIP Trunk to Integra Telecom via CUBE



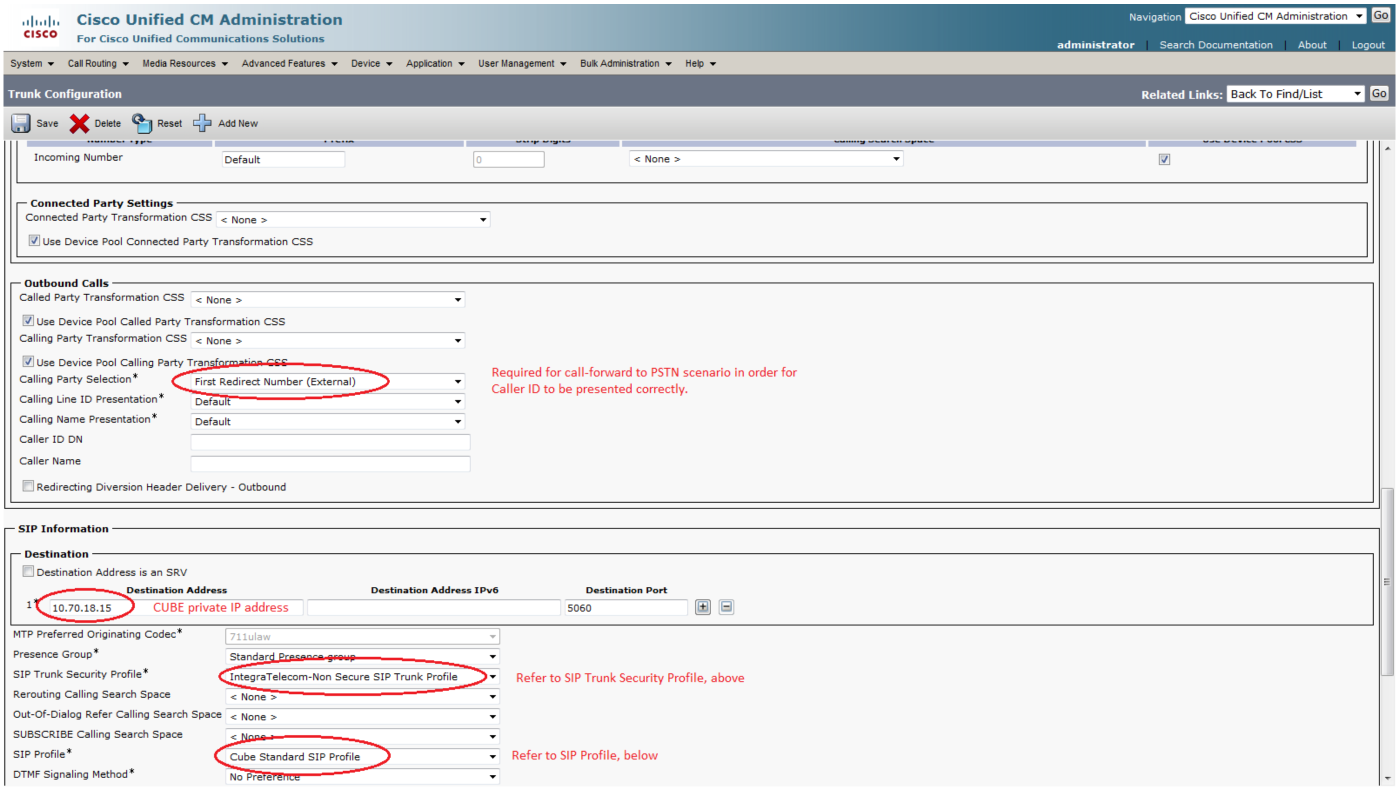
Note: Testing was conducted at tekVizion Labs®.

Figure 9. SIP Trunk to Integra Telecom via CUBE (cont.)



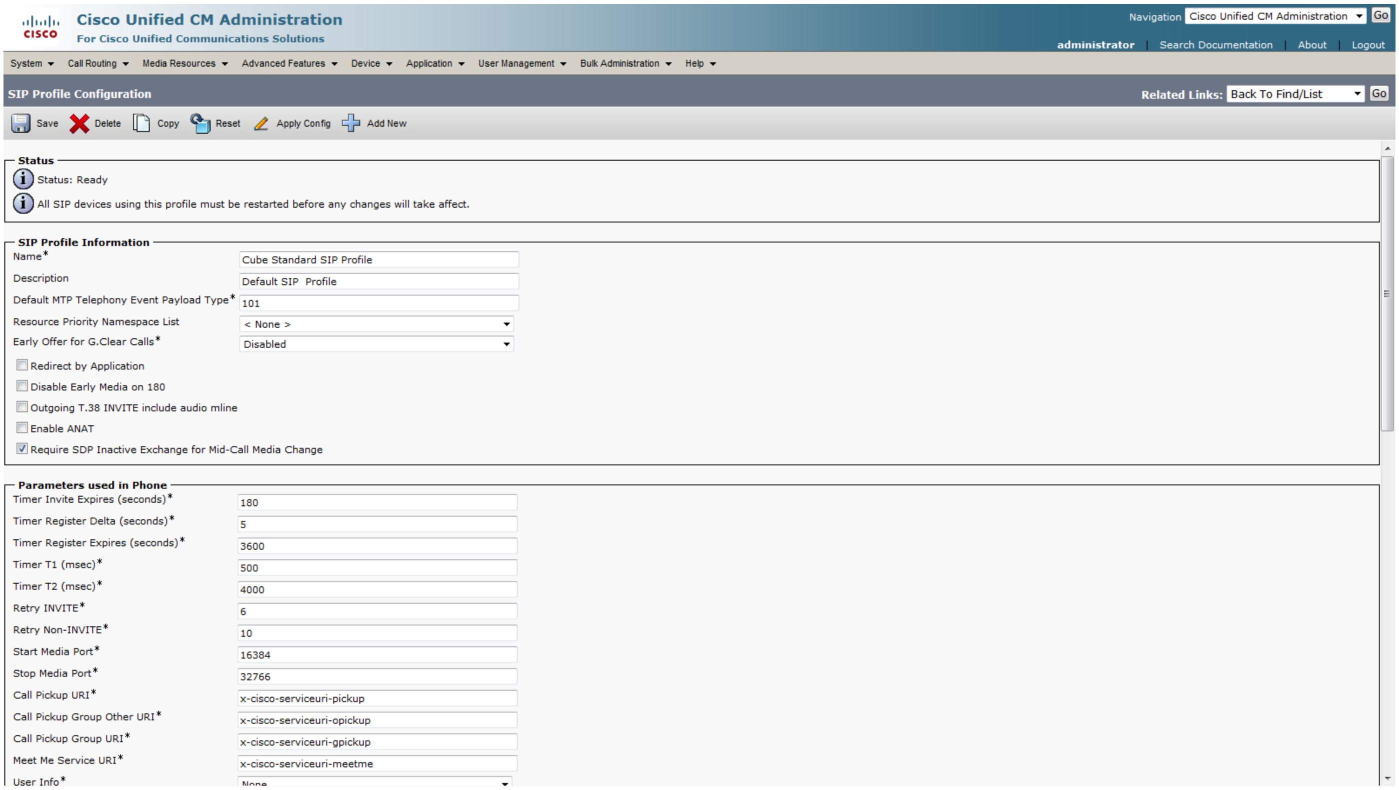
Note: Testing was conducted at tekVizion Labs®.

Figure 10. SIP Trunk to Integra Telecom via CUBE (cont.)



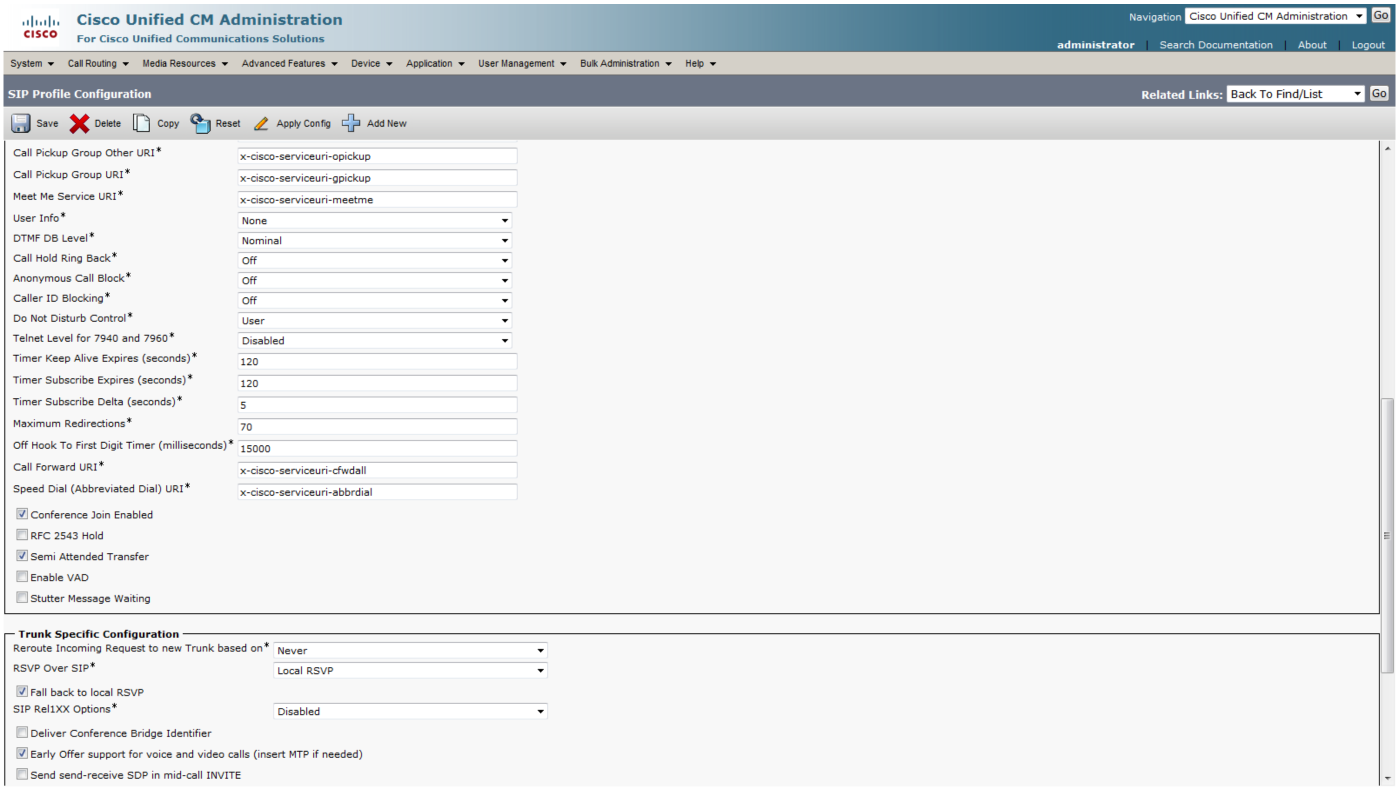
Note: Testing was conducted at tekVizion Labs®.

Figure 11. SIP Profile



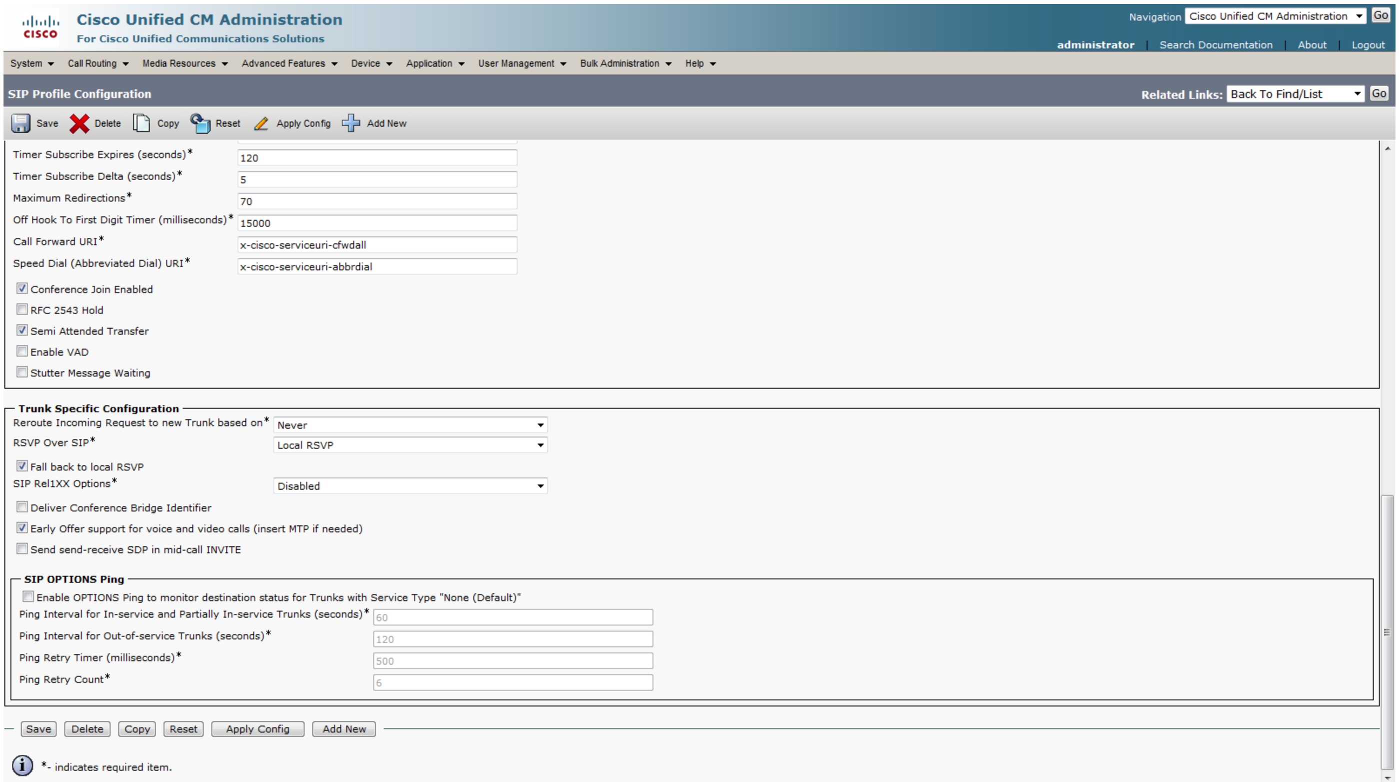
Note: Testing was conducted at tekVizion Labs®.

Figure 12. SIP Profile (cont.)



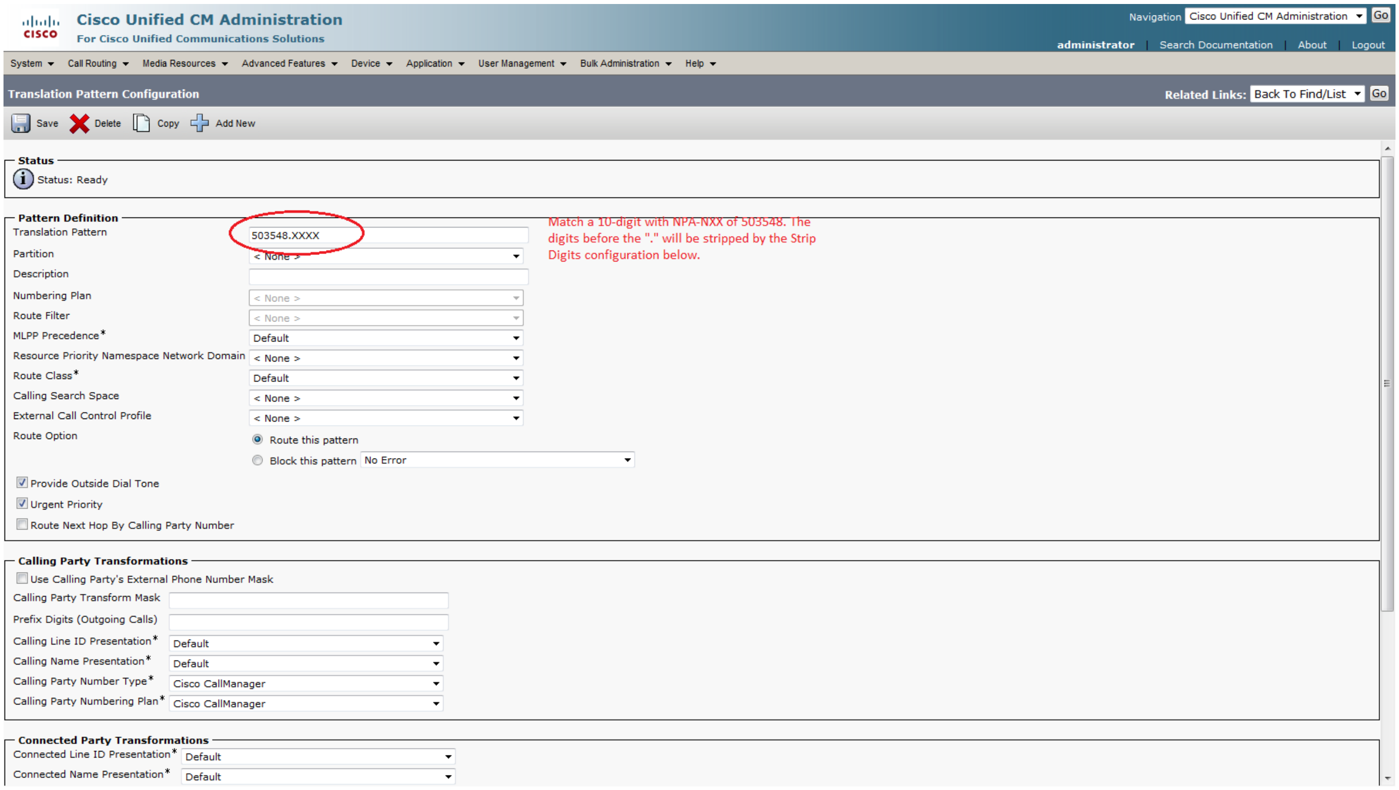
Note: Testing was conducted at tekVizion Labs®.

Figure 13. SIP Profile (cont.)



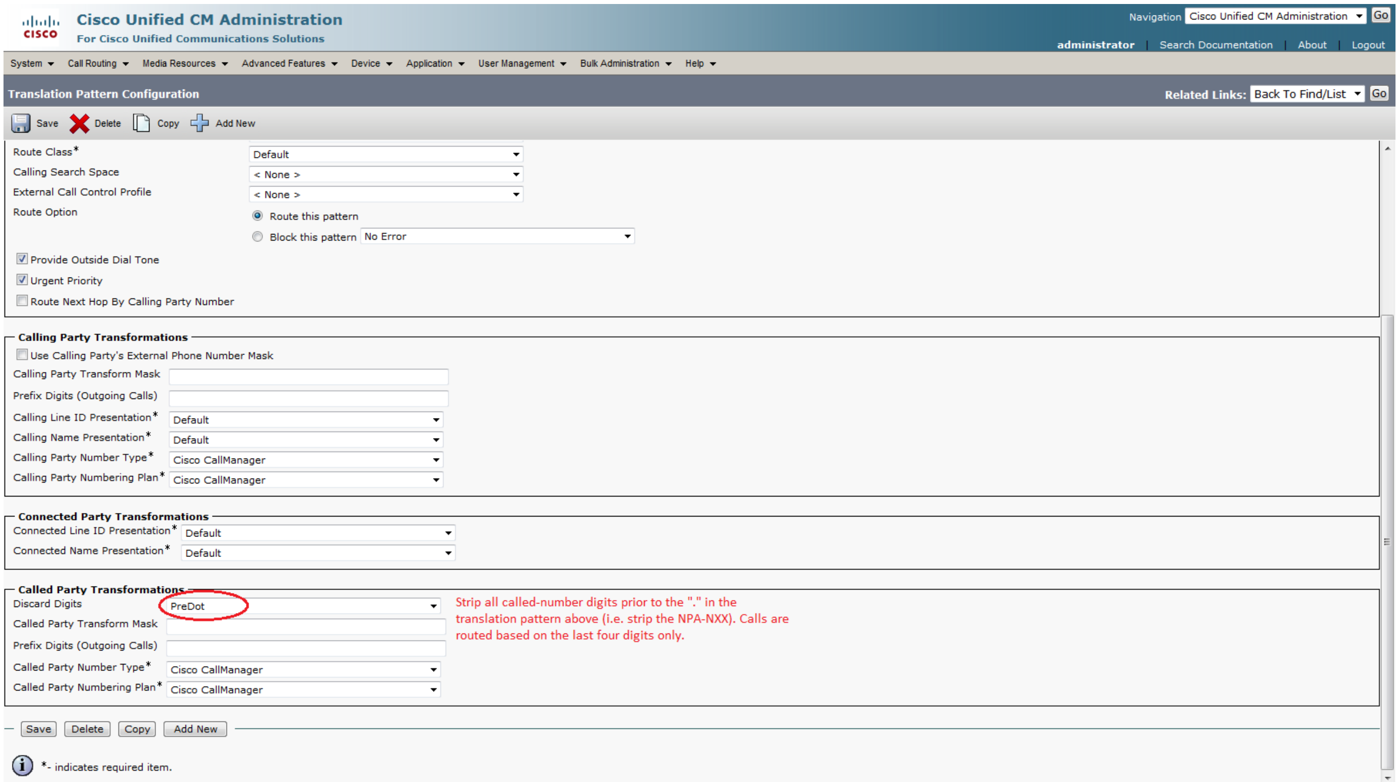
Note: Testing was conducted at tekVizion Labs®.

Figure 14. Translation Pattern for incoming calls from Integra Telecom



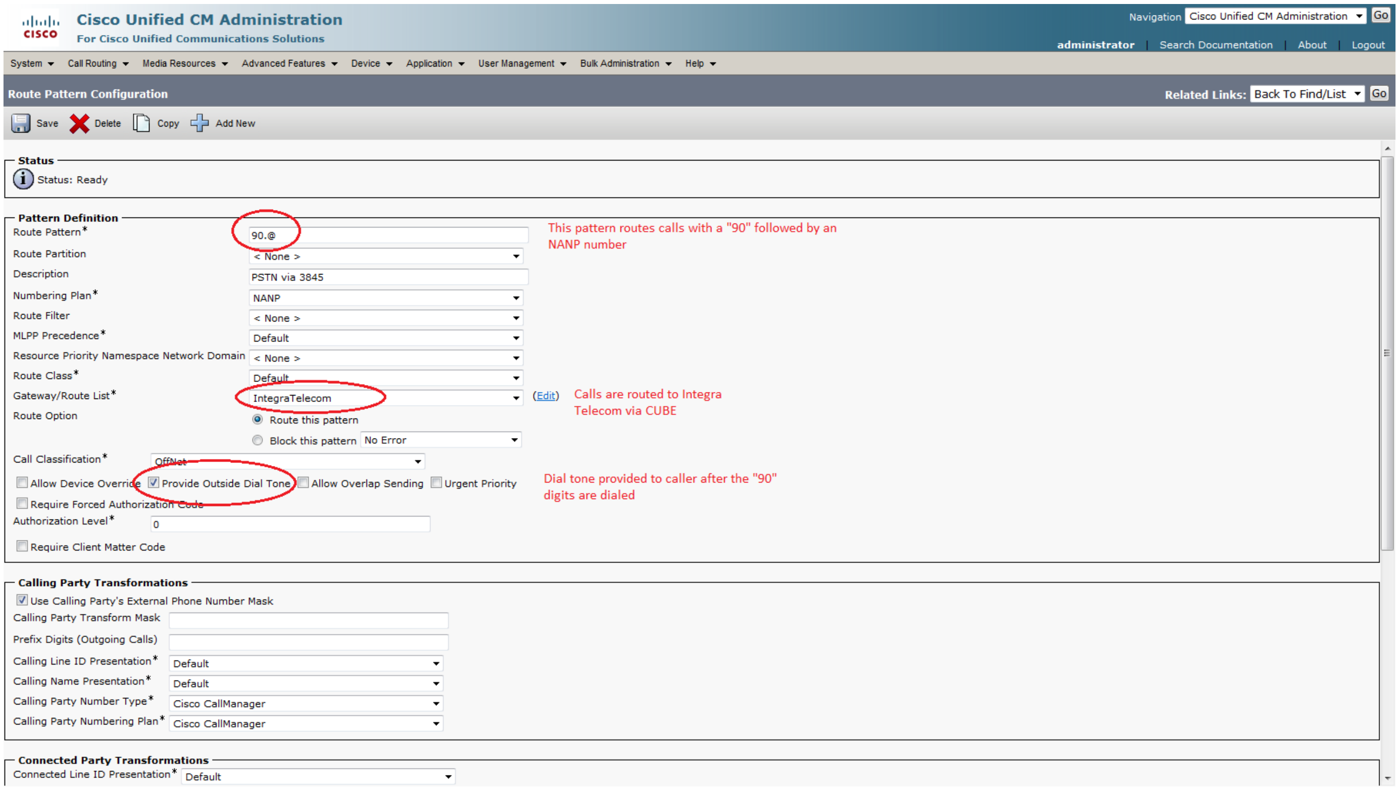
Note: Testing was conducted at tekVizion Labs®.

Figure 15. Translation Pattern for incoming calls from Integra Telecom (cont.)



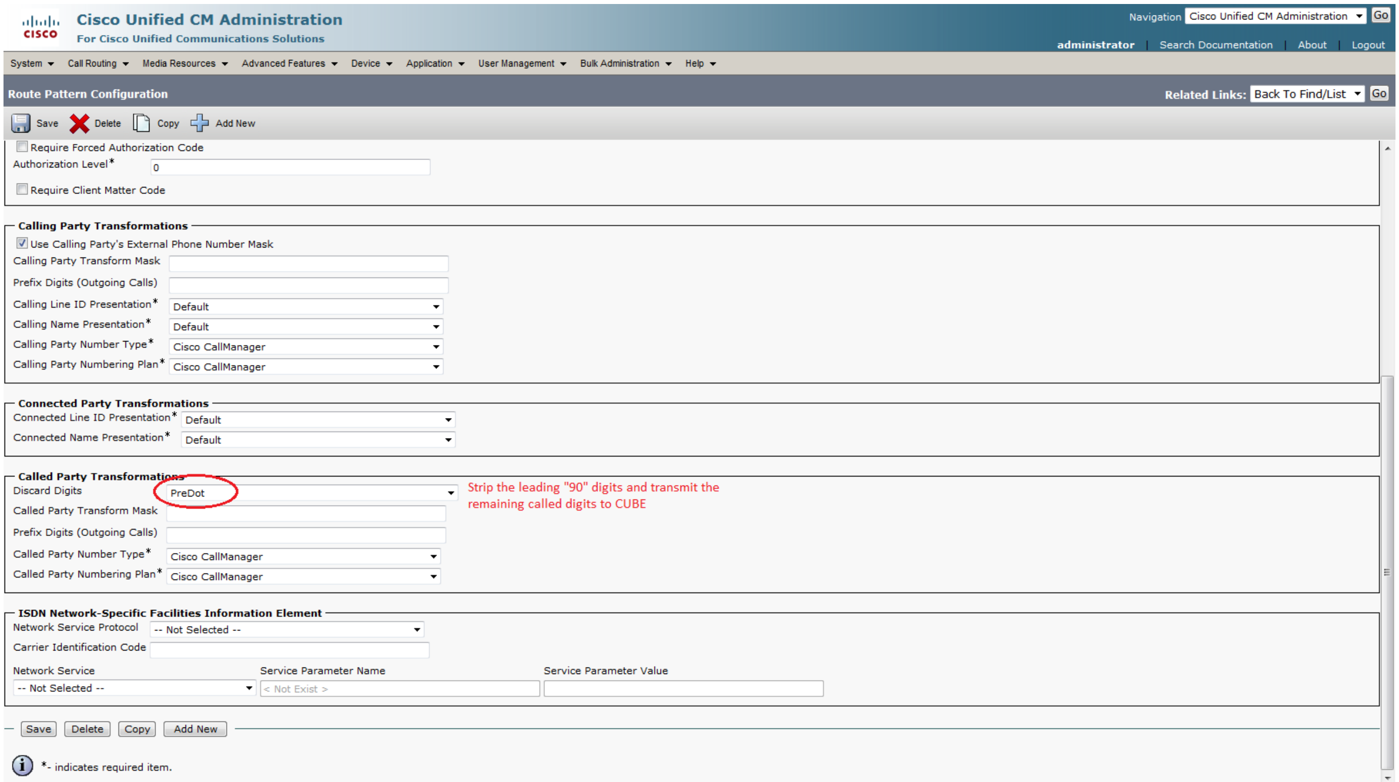
Note: Testing was conducted at tekVizion Labs®.

Figure 16. Route Pattern Configuration for SIP trunk to Integra Telecom via CUBE



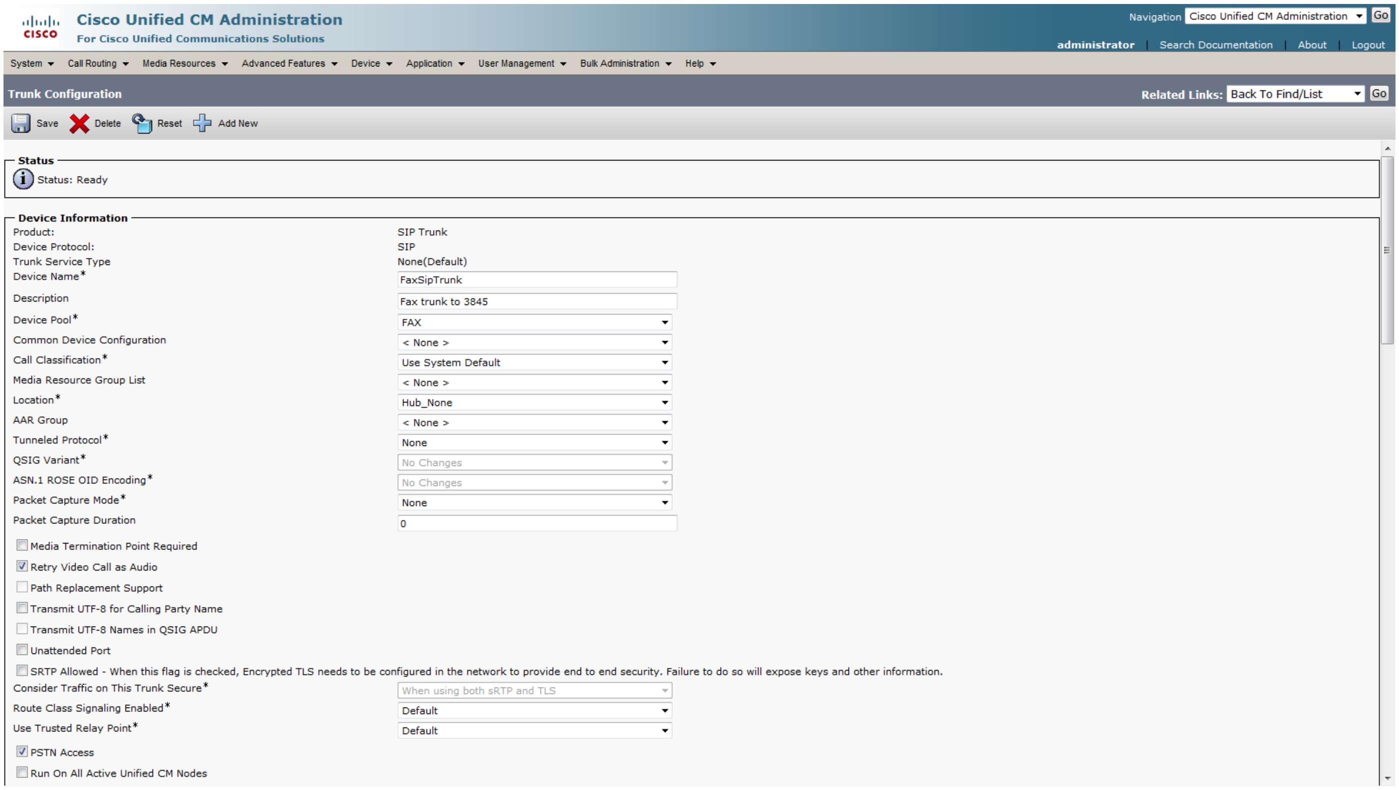
Note: Testing was conducted at tekVizion Labs®.

Figure 17. Route Pattern Configuration for SIP trunk to Integra Telecom via CUBE (cont)



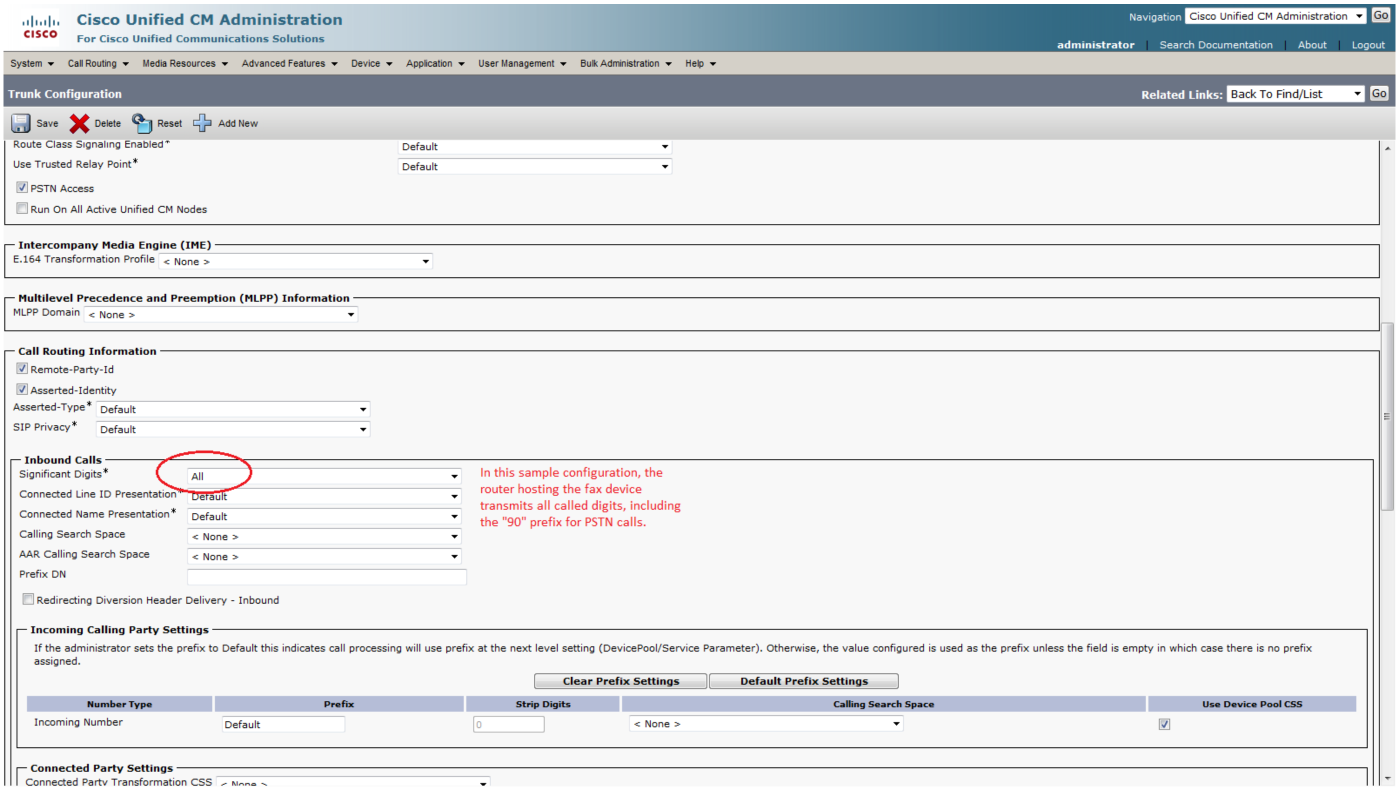
Note: Testing was conducted at tekVizion Labs®.

Figure 18. SIP Trunk to Cisco 3845 (for analog fax)



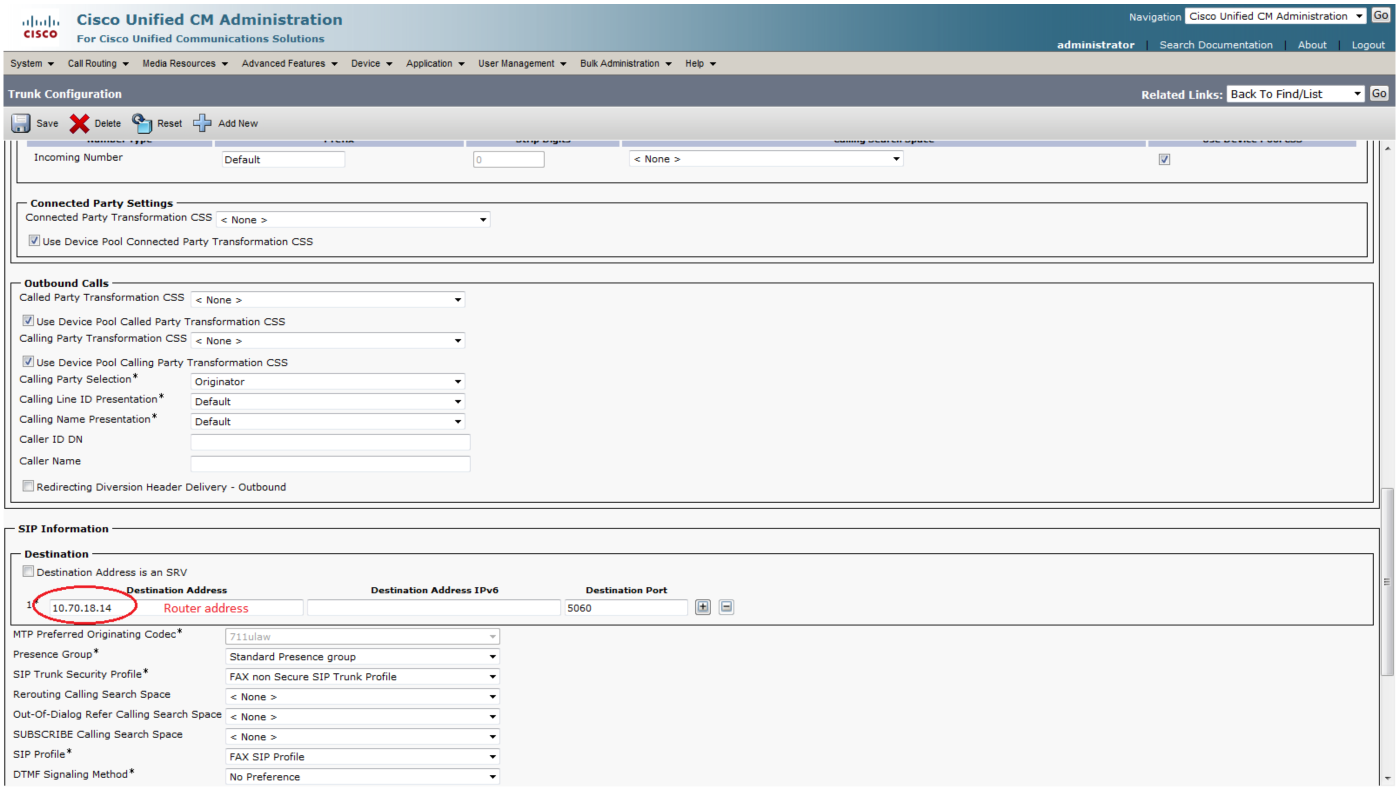
Note: Testing was conducted at tekVizion Labs®.

Figure 19. SIP Trunk to Cisco 3845 (for analog fax) (cont)



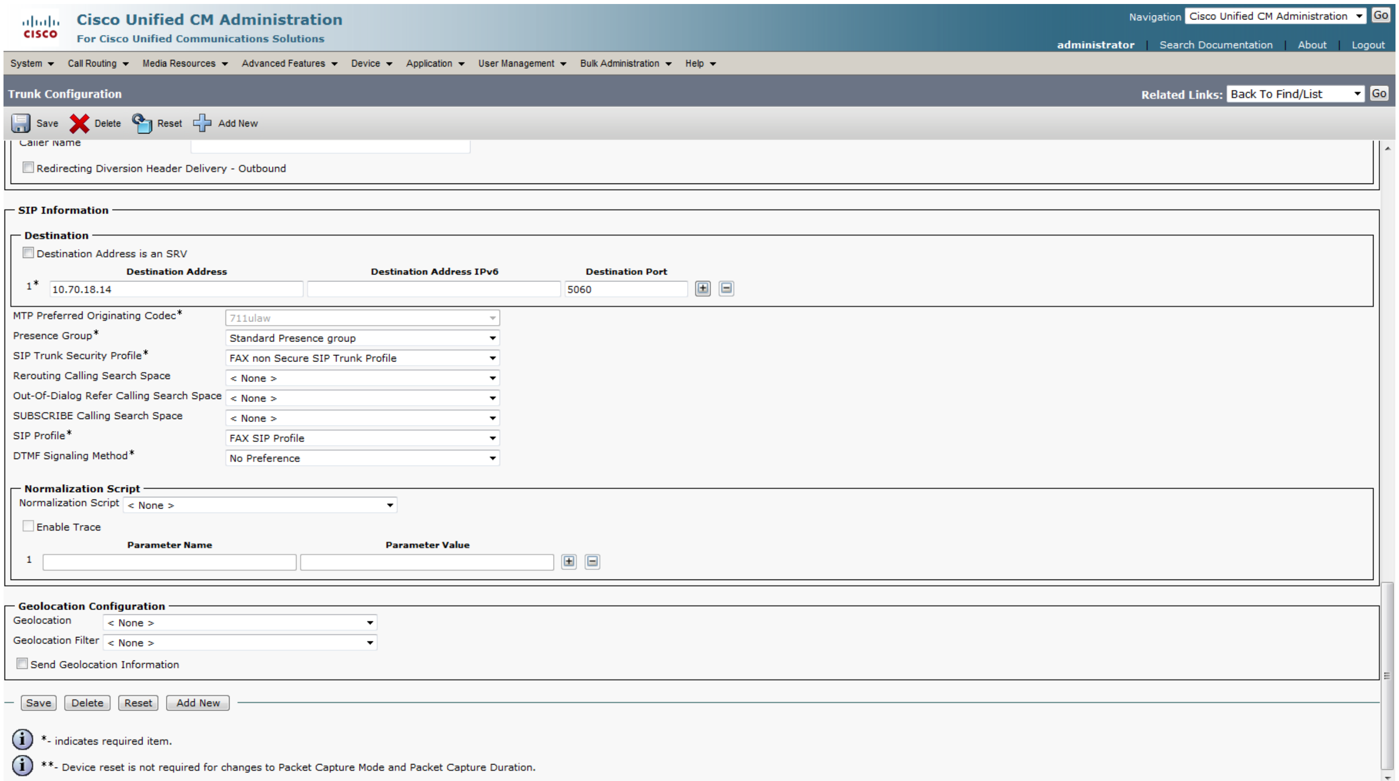
Note: Testing was conducted at tekVizion Labs®.

Figure 20. SIP Trunk to Cisco 3845 (for analog fax) (cont)



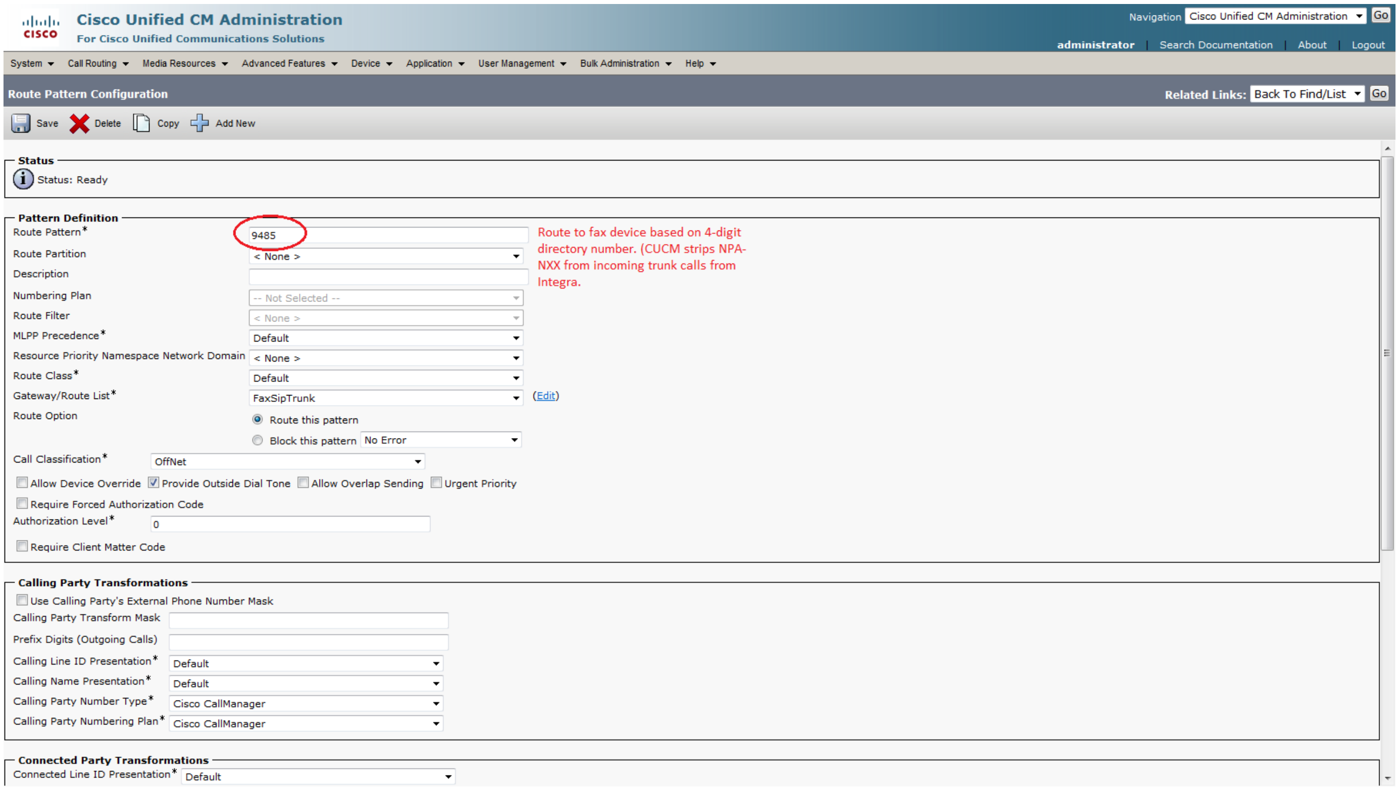
Note: Testing was conducted at tekVizion Labs®.

Figure 21. SIP Trunk to Cisco 3845 (for analog fax) (cont)



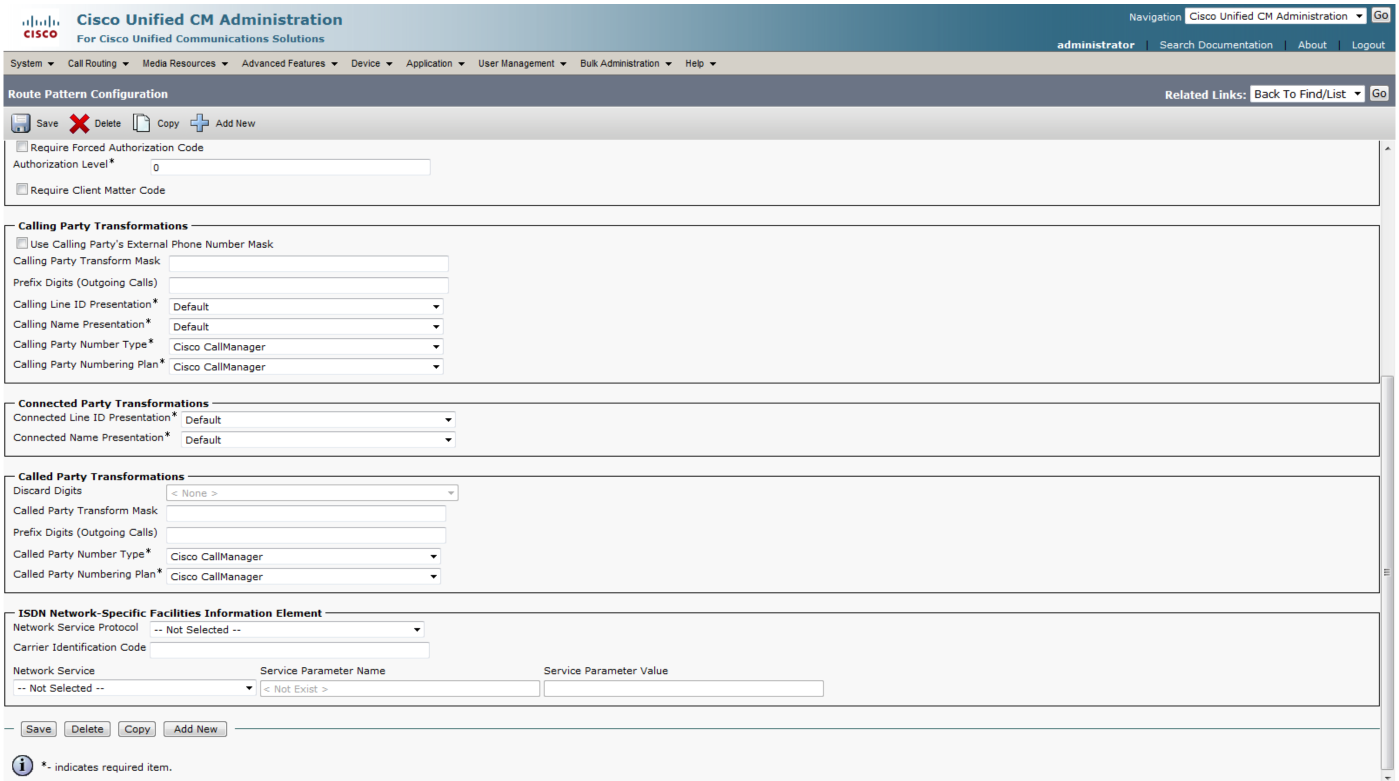
Note: Testing was conducted at tekVizion Labs®.

Figure 22. Route Pattern Configuration (fax)



Note: Testing was conducted at tekVizion Labs®.

Figure 23. Route Pattern Configuration (fax) (cont)



Note: Testing was conducted at tekVizion Labs®.

(Optional) Configuring a Cisco 3845 for analog fax

Cisco 3845 configuration for analog fax endpoint

Note: only the configuration elements relevant to the fax endpoint and the SIP trunk to CUCM are shown

```
dial-peer voice 400 pots
 description POTS dial peer for analog fax endpoint
 destination-pattern 9485[21]
 fax rate voice
 port 0/0/0
!
dial-peer voice 201 voip
 description Outbound dial peer for PSTN fax clls via CUCM
 destination-pattern 90T[22]
 session protocol sipv2
 session target dns:clus3pubsub.lab.tekvizion.com
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay sg3-to-g3[23]
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw[24]
 no vad
!
dial-peer voice 300 voip
 description incoming dial peer for fax calls
 incoming called-number 9485
 codec g711ulaw
 fax-relay sg3-to-g3
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
```

[21] CUCM sends 4-digit extension for inbound fax calls to this endpoint.
[22] For fax calls to PSTN caller dials 9+PSTN number. CUCM strips the “9” before forwarding to CUBE.
[23] Enable sg3 spoofing.
[24] Enable T.38 for fax. Fall back to G.711 if T.38 is unsuccessful.

Note: Testing was conducted at tekVizion Labs®.

Acronyms

Acronym	Definitions
SIP	Session Initiation Protocol
MGCP	Media Gateway Control Protocol
SCCP	Skinny Client Control Protocol
Cisco UCM	Cisco Unified Communications Manager
Cisco UBE	Cisco Unified Border Element

Note: Testing was conducted at tekVizion Labs®.

Important Information

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