Application Note

Spectrum Enterprise SIP Trunking:

Cisco Unified Communications Manager 11.5.x with Cisco Unified Border Element (CUBE 11.5.2) on ISR 4321/K9 [IOS-XE 16.3.2] using SIP

October 24, 2017

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Introduction

Service Providers today, such as Spectrum Enterprise[1], are offering alternative methods to connect to the PSTN via their IP networks. 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.

A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and Spectrum network, Cisco Unified Border Element (CUBE) ISR 4321/K9 running IOS 16.x can be used. The Cisco Unified Border Element 16.x provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.x connected to Spectrum network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco UCM (Cisco Unified Communications Manager). Only configuration settings specifically required for Spectrum Enterprise 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 a Cisco Unified Communications Manager (CUCM) 11.5.x and Cisco Unified Border Element (CUBE) on ISR 4321/K9 [IOS - 16.x] for connectivity to Spectrum Enterprise SIP Trunking services. The deployment model covered in this application note is CPE (CUCM 11.5.x) to PSTN (Spectrum Enterprise).
- Testing was performed in accordance to Spectrum Enterprise 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 interoperability with Cisco Unity Connection (CUC).
- The Cisco UCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Spectrum Enterprise SIP Trunking Service(s) 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 Spectrum Enterprise SIP Trunking services.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on Cisco UCM. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.html

[1] Spectrum Enterprise is a division of Charter Communications following a merger with Time Warner Cable and acquisition of Bright House Networks.

Network Topology

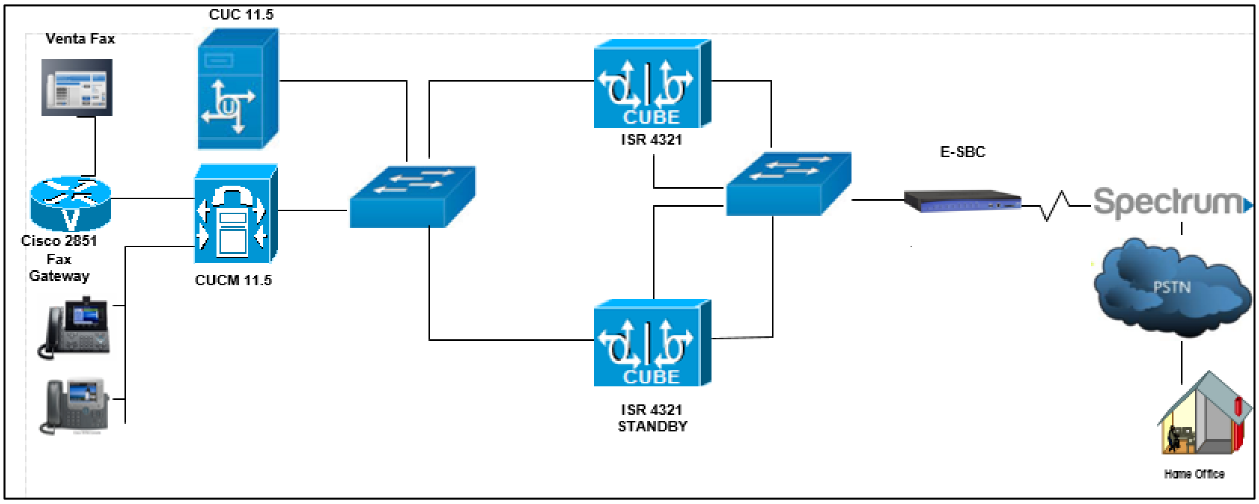


Figure 1: Network Topology

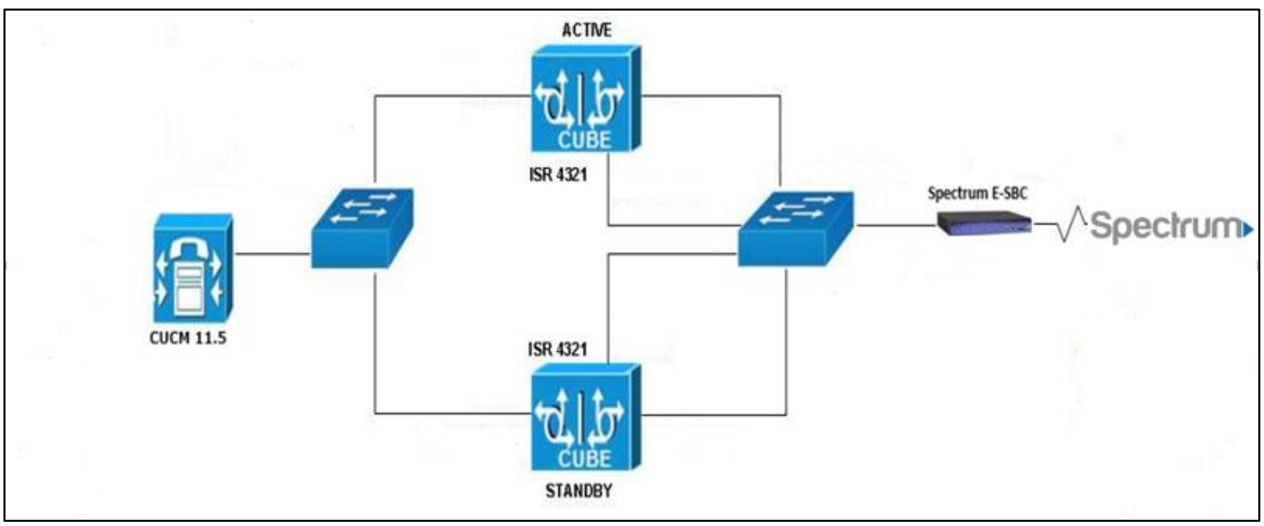


Figure 2: Cisco UBE High Availability

System Components

Hardware Requirements

- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as Cisco UBE
- Cisco ISR4321/K9 (1RU) processor with 1651964K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
- Processor board ID FLM1925W0WZ
- Cisco 2851 Fax Gateway
- IP phones 9971 (SIP) and 7975 (SCCP)
- Spectrum eSBC – Provided and Managed by Spectrum Enterprise

Software Requirements

- Cisco Unified Communications Manager 11.5.x
- Cisco Unity Connection 11.5.x
- IOS 16.3.2 for ISR 4321/K9 Cisco Unified Border Element
- Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.3.2, RELEASE SOFTWARE (fc4)
- Cisco IOS XE Software, Version 16.03.02
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway
- Spectrum eSBC – Provided and Managed by Spectrum Enterprise

Features

Features Supported

- Incoming and outgoing off-net calls using G711ULAW voice codecs
- Call Hold
- Call Transfer (unattended and attended)
- Call Conference
- Call Forward (all, busy, no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF Relay (both directions) (RFC2833)
- Media flow-through on Cisco UBE
- Fax (G711 Pass-through)

Features Not Supported

- Cisco IP phones used in this test do not support Blind Transfer
- G729 is not supported by Service Provider
- T38 Fax was not available for testing during the test cycle
- Cisco does not support Blind Call Transfer

Caveats

- Caller ID is not updated after attended or unattended transfers to off-net phones. This is due to a limitation on Cisco UBE and will be resolved in the next release. The issue does not impact the calls.

Configuration

Configuring Cisco Unified Border Element

Network Interface

Configure Ethernet IP address and sub interface. The IP address and VLAN encapsulation used are for illustration only, the actual IP address can vary. For SIP trunks two IP addresses must be configured - for LAN and WAN.

```
interface GigabitEthernet0/0/0
 ip address 10.249.248.110 255.255.0.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.249.248.98 exclusive
!
interface GigabitEthernet0/0/1
 description LAN
 ip address 10.80.18.12 255.255.255.0
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 10.80.18.10 exclusive
```

Global CUBE Settings

In order to enable CUBE IP2IP gateway functionality, enter the following:

```
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711ulaw
 sip
  rel1xx supported "rel100"
  session refresh
  header-passing
  asserted-id pai
  privacy pstn
  conn-reuse
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
  g729 annexb-all
```

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
fax protocol	Specifies the fax protocol
asserted-id	Specifies the type of privacy header in the outgoing SIP requests and response messages
early-offer forced	Enables SIP Delayed-Offer to Early-Offer globally
midcall-signaling passthru	Passes SIP messages from one IP leg to another IP leg

Codecs

G711Ulaw is used as the preferred codec for this testing and changed the preferences according to the test plan description

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

Dial Peer

The CUBE uses dial-peer to route the call based on the digit to route the call accordingly.

```
dial-peer voice 10 voip
 description Incoming from CUCM
 huntstop
 session protocol sipv2
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip early-offer forced
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 g711ulaw
 no vad
!
dial-peer voice 20 voip
 description Outgoing to Spectrum
 huntstop
 destination-pattern [0-9]T
 session protocol sipv2
```

```
 session target ipv4:10.249.248.100
 voice-class codec 1
 voice-class sip options-ping 60
 voice-class sip early-offer forced
 voice-class sip profiles 1
 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 g711ulaw
 no vad
!
dial-peer voice 30 voip
 description Incoming from Spectrum
 huntstop
 session protocol sipv2
 incoming called-number 469T
 voice-class codec 1
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 g711ulaw
 no vad
!
dial-peer voice 40 voip
```

```
description Outgoing to CUCM
huntstop
destination-pattern 469T
session protocol sipv2
session target ipv4:10.80.18.3
voice-class codec 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 g711ulaw
no vad
```

Call Flow

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

For incoming PSTN calls, the Cisco UBE presents the full ten-digit DID number to Cisco UCM. The Cisco UCM picks up the last 4 significant Digits configured under SIP Trunk and routes the call based on those 4 digits. Voice calls are routed to IP phones; Fax calls are routed via a 4-digit route pattern over a SIP trunk that terminates on the Fax Gateway and in turn to the VentaFax client connected to the Fax Gateway.

CPE callers make outbound PSTN calls by dialing a "9" prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, Cisco fax Gateway sends to Cisco UCM the DID with leading access code "9". A "9.@" route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the Cisco UBE for Voice call or Fax. For PBX to PBX via Spectrum Enterprise, Caller dial 9 prefix followed by the target 1+10Digit DID no for that extension number, 9 was stripped and the 1 +10 digits number was send to Cisco UBE, Cisco UBE sends the full 1+ 10 digits DID under Dial Peer 20 and send to Spectrum network which will direct back to Cisco UBE and handled same as normal incoming PSTN call. For International calls same pattern 9 followed by 011, country code and calling no is used.

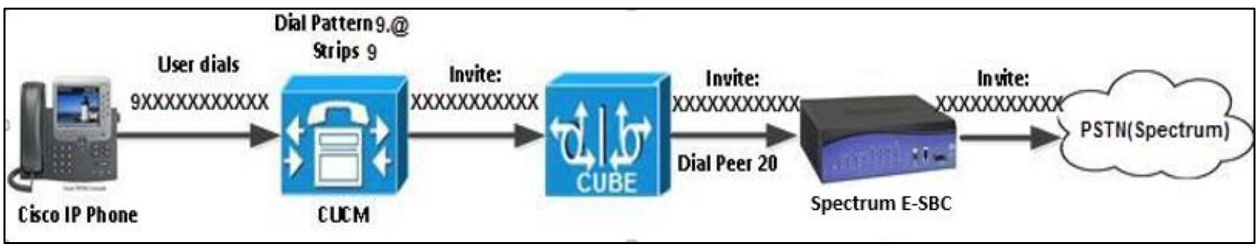


Figure 3: Outbound Voice Call

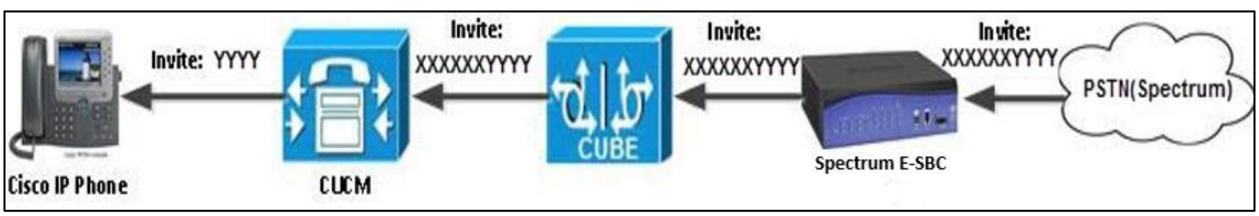


Figure 4: Inbound Voice Call

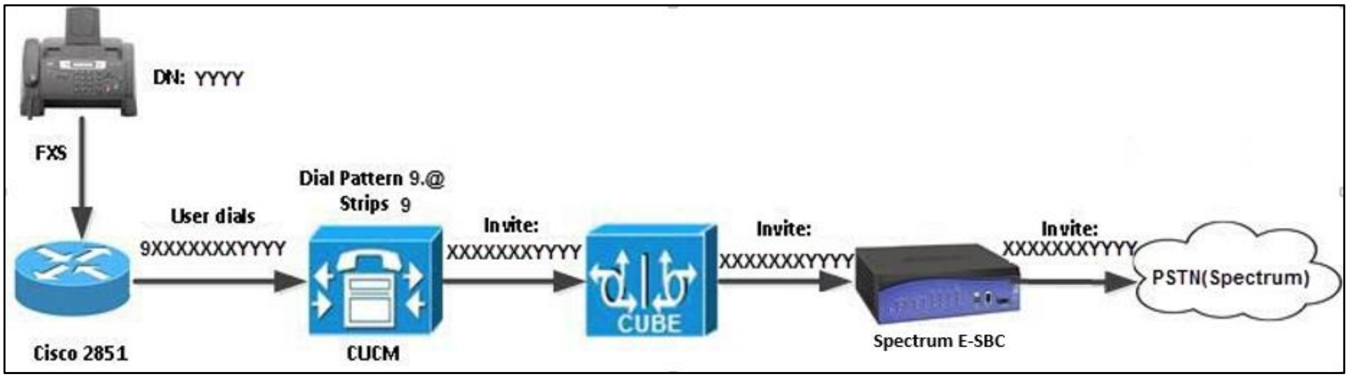


Figure 5: Outbound Fax Call

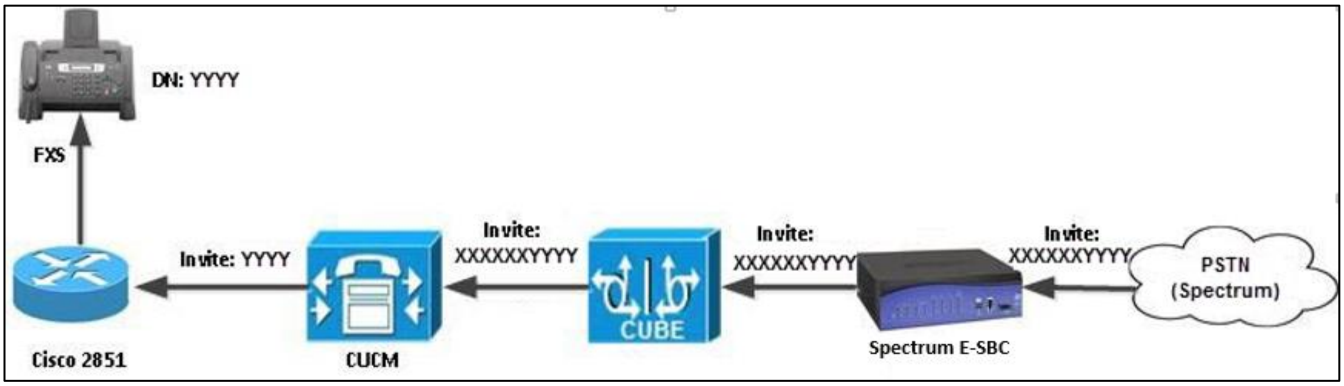


Figure 6: Inbound Fax Call

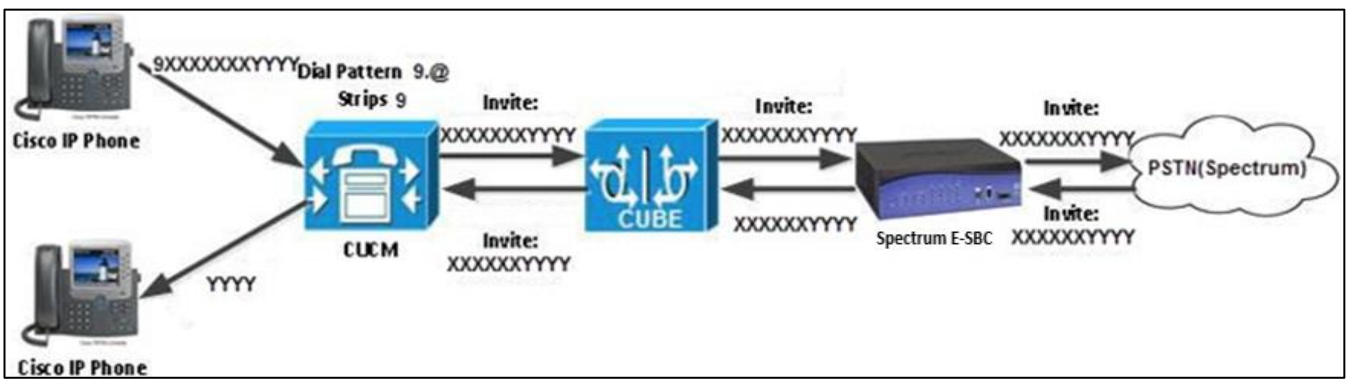


Figure 7: PBX to PBX via Spectrum Call

Configuration Example

The following configuration snippet contains a sample configuration of the CUBE with all parameters mentioned previously

Active Cisco UBE

```
User Access Verification

Username: cisco

Password:

CUBE2#sh run

Building configuration...


version 16.3

service timestamps debug datetime msec

service timestamps log datetime msec

no platform punt-keepalive disable-kernel-core

!

hostname CUBE2

!

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-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
 certificate self-signed 01
  30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 32353438 34343332 3436301E 170D3137 30363237 32303334
  34345A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 35343834
  34333234 36308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
  0A028201 0100C68F 550FDD45 ED38A673 62CE87C5 63533BB8 961BA455 815FF15F
  B176F1FC CFE7B1CB B7339913 CE6C099B 81BFC015 E07BBF68 28447C7F 1110A35D
  9CBC5B45 ABF264EC 2E3B759F D011A2C4 F15AE9D0 4B0639B7 F10F9BC5 6FC8DCE3
  30A95FC9 4D2428E5 D87B4A36 02F264C9 9EEC37A2 8234BE96 64AC5269 AB37F8A4
  CDFCF43B 9DAA4608 45A68382 210032EA ECE3DD81 CDAC3CAC 33F04215 7F79AF16
  DA46BAAD 4AE6D7A2 50647AB6 243B37F1 3ED7660D D9B83510 9E68B935 AA9FC6B0
  9C4FD9FA F77AB037 9802FE33 16F7CC59 A0FA9835 3755D690 693A1156 156641B3
  F6271738 3AC8A0E9 389A49E3 FA0F04CE 3C690B25 5908ED3B 4AD33C83 218704D6
```

```
  55C8BC50 CEAB0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
  301F0603 551D2304 18301680 14C48FB9 0BE2A1C0 428392FE 5266116D 7AA657B4
  ED301D06 03551D0E 04160414 C48FB90B E2A1C042 8392FE52 66116D7A A657B4ED
  300D0609 2A864886 F70D0101 05050003 82010100 10591983 4AA2DB41 F7C3A9F5
  33AC972A 601AEC40 9A630D0D D946C6D0 76EF44BE A9D352E1 95F11C50 323063B9
  1ECD71D6 55188003 1C313FD4 A3A93A1A 23932383 5D2B95E9 0204D5C1 585CE98B
  B5EB786C FDC8F1B2 04D26F86 F0AAE8AB C5672FB1 E81C030F B6D7F72A 6D7774D7
  B667D36A FF5DA105 C5CB6091 D4BD1441 38F73AEB 95FAE794 12A82393 601B7313
  28F61935 00FFD91C 4B4A1F09 8D7A7C20 62B106EF 60966B43 4DB025F0 72A032B0
  248CCB29 DB7BB8DD 7D05F674 1DFBEEF9 47F9617B EDDA17B3 9539CAE3 B4205C3C
  D2819C39 17EAA7C0 23943D56 88219C29 E8E1BB66 21510B8C B7DBE6AC 5F5F4945
  6492B908 FCAF5747 8E22F406 F56C243D CC96981E
        quit
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711ulaw
 sip
  rel1xx supported "rel100"
  session refresh
  header-passing
  asserted-id pai
  privacy pstn
  conn-reuse
  early-offer forced
```

```
  midcall-signaling passthru
  privacy-policy passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:469573\1@\2>
"
!
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 0 cisco
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   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
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
 ip address 10.249.248.110 255.255.0.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.249.248.98 exclusive
!
interface GigabitEthernet0/0/1
 description LAN
 ip address 10.80.18.12 255.255.255.0
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
 ip address 10.89.20.10 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
```

```
 no ip address
 shutdown
 negotiation auto
!
interface Vlan1
 no ip address
 shutdown
!
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 10.249.248.97
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 172.16.31.0 255.255.255.0 10.80.18.1
!
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 10 voip
 description Incoming from CUCM
```

```
 huntstop
 session protocol sipv2
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip early-offer forced
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 g711ulaw
 no vad
!
dial-peer voice 20 voip
 description Outgoing to Spectrum
 huntstop
 destination-pattern [0-9]T
 session protocol sipv2
 session target ipv4:10.249.248.100
 voice-class codec 1
 voice-class sip options-ping 60
 voice-class sip early-offer forced
 voice-class sip profiles 1
 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 g711ulaw
 no vad
!
dial-peer voice 30 voip
 description Incoming from Spectrum
 huntstop
 session protocol sipv2
 incoming called-number 469T
 voice-class codec 1
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 g711ulaw
 no vad
!
dial-peer voice 40 voip
 description Outgoing to CUCM
 huntstop
 destination-pattern 469T
 session protocol sipv2
 session target ipv4:10.80.18.3
 voice-class codec 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 g711ulaw
 no vad
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 privilege level 15
 password cisco
 logging synchronous
 login local
 transport input telnet ssh
!
end
```

CUBE2#

Standby Cisco UBE

```
User Access Verification
Username: cisco
Password:
CUBE1#sh run
Building configuration...

version 16.3
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CUBE1
!
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-1270583006
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-1270583006
 revocation-check none
 rsakeypair TP-self-signed-1270583006
!
crypto pki certificate chain TP-self-signed-1270583006
 certificate self-signed 01
  30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 31323730 35383330 3036301E 170D3137 30363237 32303539
  31385A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 32373035
  38333030 36308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
  0A028201 01009357 89CDDFF2 740408D8 9D185E93 E6426DED 9FDA526D E01BE0B9
  2589A47F 06FEDD14 6349C1F3 55C8EA44 4D57D07E 20632E1D C43930BD A0F3EF9F
  A65D105B 9CEB70FC 99865700 6E282D45 41D865DB D57057A4 D5A8B2D4 FFBEB680
  9A579600 F421DB9A C644B287 AF57547B C704EE31 3EB85920 C0BDD17C DD438273
  194EE1F4 868B3722 F6879823 96CAE06A 2F3F52E3 E4EC16F2 180ED78B C85F472C
  9C9850DD 0051069A 89FE6636 42EEEA22 BF39D1D3 039973CC 4F1D7B41 4924767C
  EFBD18A0 5B379A2D 19A7F46E A50A5C36 7B7A1796 F4E50235 92BABE2D 29CDC9E7
  96498FCD 84505728 2E2371AD 2AF67552 77347558 1AE693D0 0990E608 A4B86C23
  AA1217B0 D5470203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
```

```
  301F0603 551D2304 18301680 14871077 5EF40EA7 CDF094B7 98F70346 512C8849
  73301D06 03551D0E 04160414 8710775E F40EA7CD F094B798 F7034651 2C884973
  300D0609 2A864886 F70D0101 05050003 82010100 207D6A92 9E325E56 827134D3
  3B537BD5 F4E10F9B 290B3836 77AA1E5A 9B7F945E 025603F8 D8BF4CF0 A85B5219
  4CB96446 76FCC5BD 31744F18 BA94CFC2 86DDD092 EBD72FE3 8470D1A4 05A2BF95
  46DCA505 E46C0D4C FFAA5B35 A2B142AF 658DD7D7 98D9E168 771E3AAD 2CA4BCF9
  C5F835F3 92FC4F87 DB4E4ACA 33038758 1AA0F13F 317C2A98 375D7E0B 5C5F85C7
  D296FA05 7750D490 7B75FED0 136DEE87 5818B934 470058BF 16341380 AD22A121
  71A7B85B 4AE79B16 70215C44 D9137A13 101F5E3D B949745B 2FE99F42 671D5AC7
  E9BF5D6B EB46D767 8ABEB24D 1649FDB0 7C3866E2 24B2ADB6 11B35200 4EEC2D92
  E7845239 534D484B DF4D44C6 7C05BFAA D459BD31
        quit
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711ulaw
 sip
  rel1xx supported "rel100"
  session refresh
  header-passing
  asserted-id pai
  privacy pstn
  conn-reuse
  early-offer forced
  midcall-signaling passthru
```

```
  privacy-policy passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:469573\1@\2>
"
!
license udi pid ISR4321/K9 sn FDO19220MQ8
license boot level appxk9
license boot level uck9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 0 cisco
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   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
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
 ip address 10.249.248.109 255.255.0.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.249.248.98 exclusive
!
interface GigabitEthernet0/0/1
 description LAN
 ip address 10.80.18.11 255.255.255.0
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
 ip address 10.89.20.11 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
```

```
 shutdown
 negotiation auto
!
interface Vlan1
 no ip address
 shutdown
!
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 10.249.248.97
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 172.16.31.0 255.255.255.0 10.80.18.1
!
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 10 voip
 description Incoming from CUCM
 huntstop
```

```
 session protocol sipv2
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip early-offer forced
 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 nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 20 voip
 description Outgoing to Spectrum
 huntstop
 destination-pattern [0-9]T
 session protocol sipv2
 session target ipv4:10.249.248.100
 voice-class codec 1
 voice-class sip options-ping 60
 voice-class sip early-offer forced
 voice-class sip profiles 1
 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 nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 30 voip
 description Incoming from Spectrum
 huntstop
 session protocol sipv2
 incoming called-number 469T
 voice-class codec 1
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 g711ulaw
 no vad
!
dial-peer voice 40 voip
 description Outgoing to CUCM
 huntstop
 destination-pattern 469T
 session protocol sipv2
 session target ipv4:10.80.18.3
 voice-class codec 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 g711ulaw
 no vad
!
dial-peer voice 889 voip
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 privilege level 15
 password cisco
 logging synchronous
 login local
 transport input telnet ssh
!
end
```

Configuring Cisco Unified Communications Manager

CUCM Version

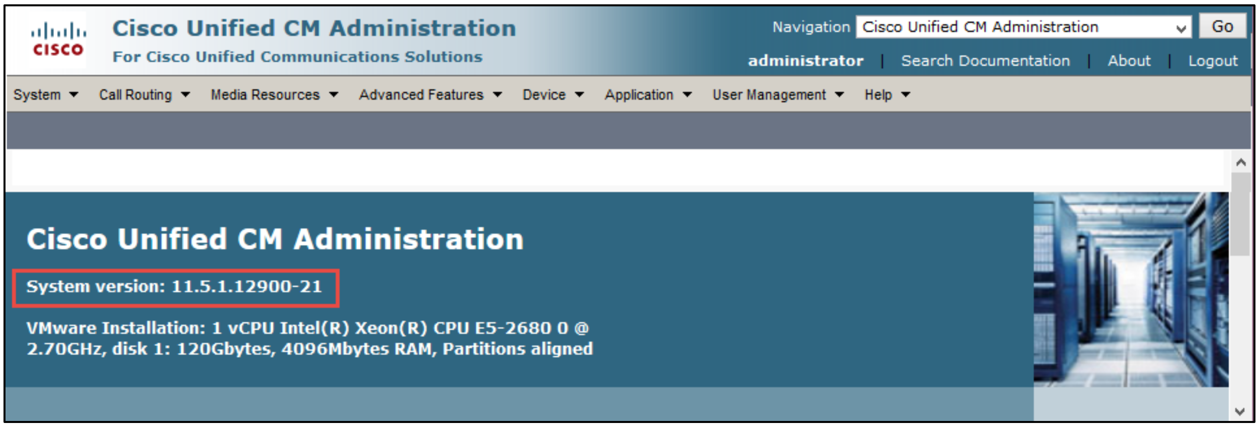


Figure 8: CUCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

1. Select **Server***: Clus21Sub1--CUCM Voice/Video (Active)
2. Select **Service***: Cisco Call Manager (Active)
3. All other fields are set to default values

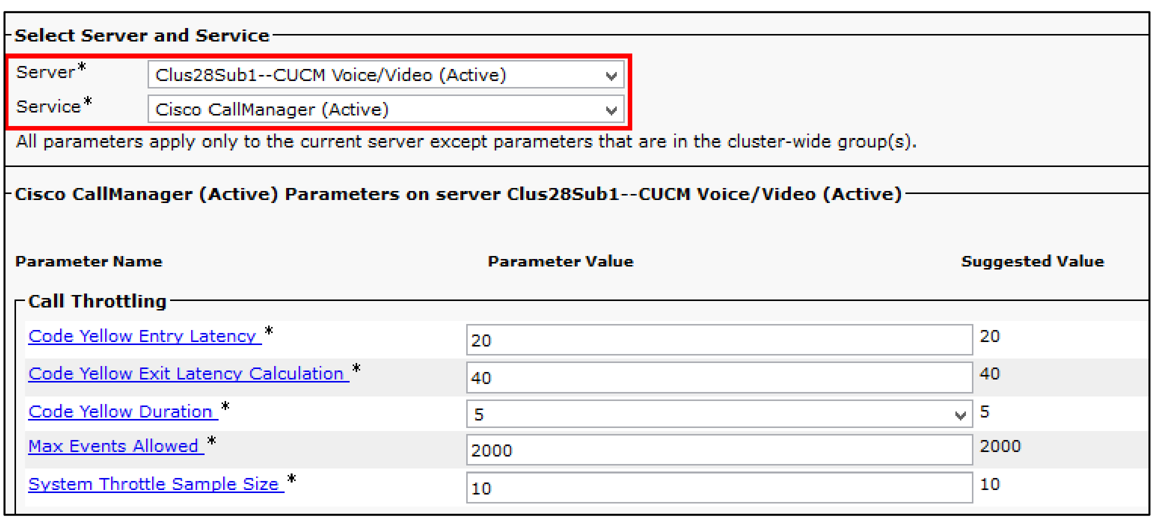


Figure 9: Service Parameters

Offnet Calls via Spectrum Enterprise SIP Trunk

Off-net calls are served by SIP trunks configured between CUCM and the Spectrum Enterprise network and calls are routed via the CUBE

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

1. **Name***: Spectrum Enterprise Non Secure SIP Trunk Profile
2. **Description**: Non Secure SIP Trunk Profile authenticated by null String

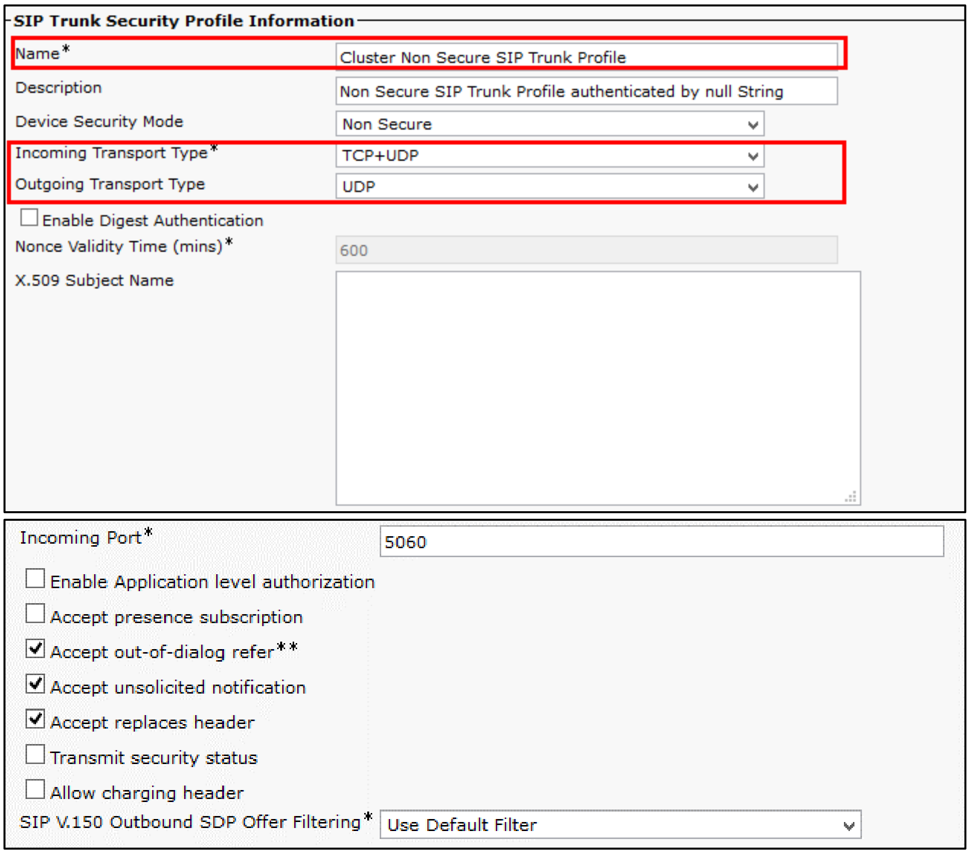


Figure 10: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to Spectrum Enterprise SBC should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.

SIP Profile Configuration

SIP Profile will be later associated with the SIP trunk

Navigation: Device → Device Settings → SIP Profile

1. **Name***: Spectrum SIP Profile
2. **Description:** Spectrum SIP Profile

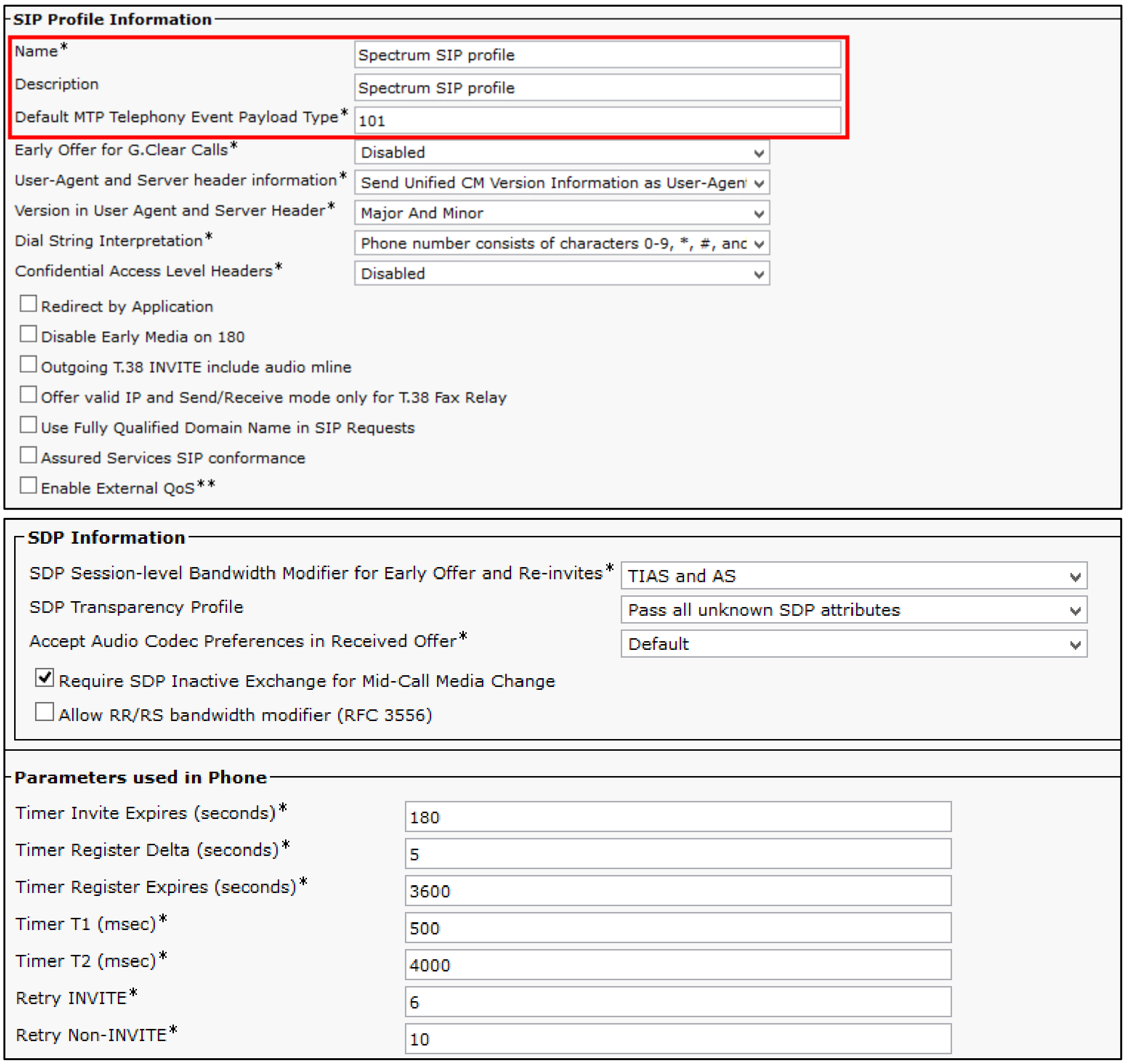


Figure 11: SIP Profile

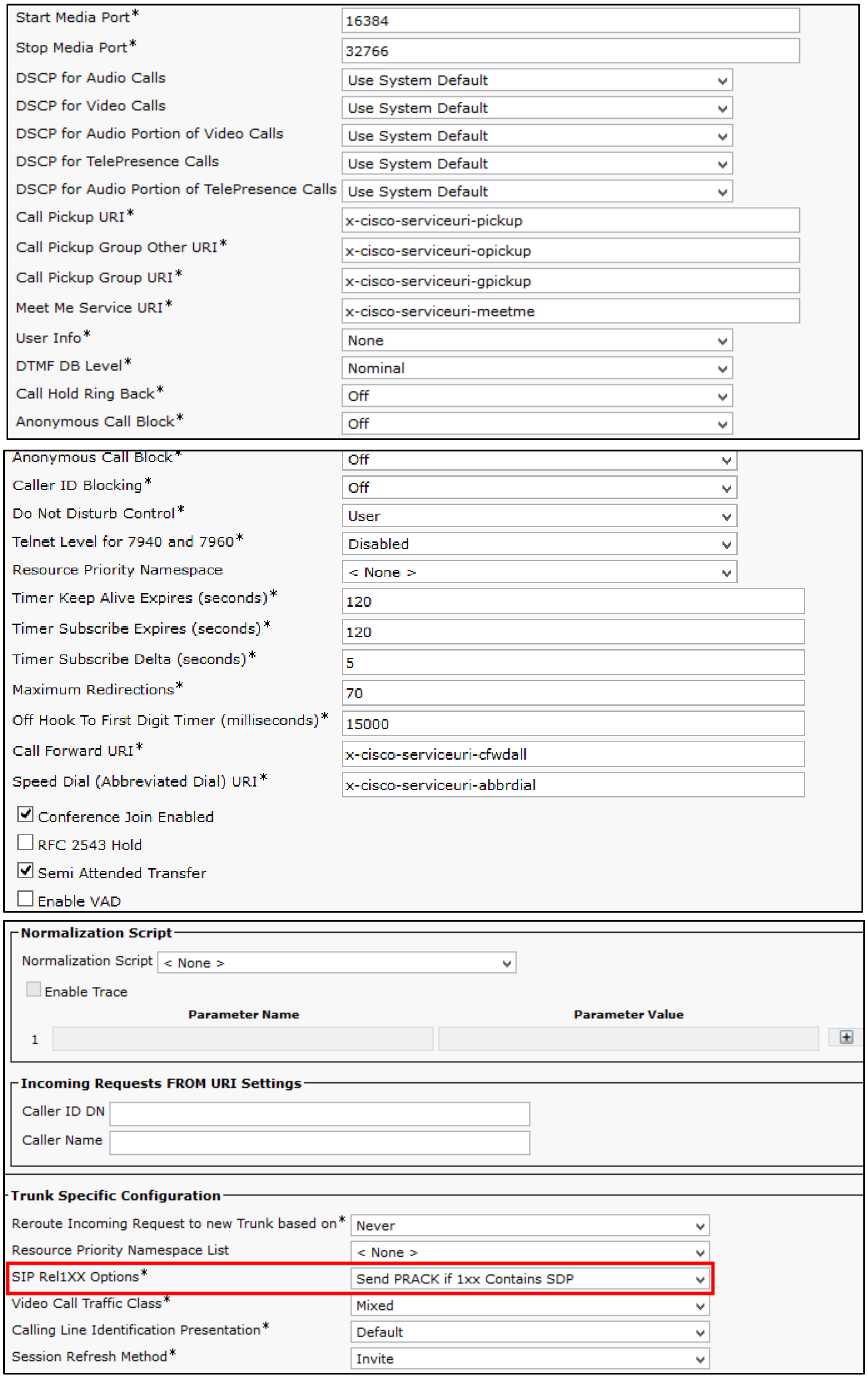


Figure 12: SIP Profile (Cont.)

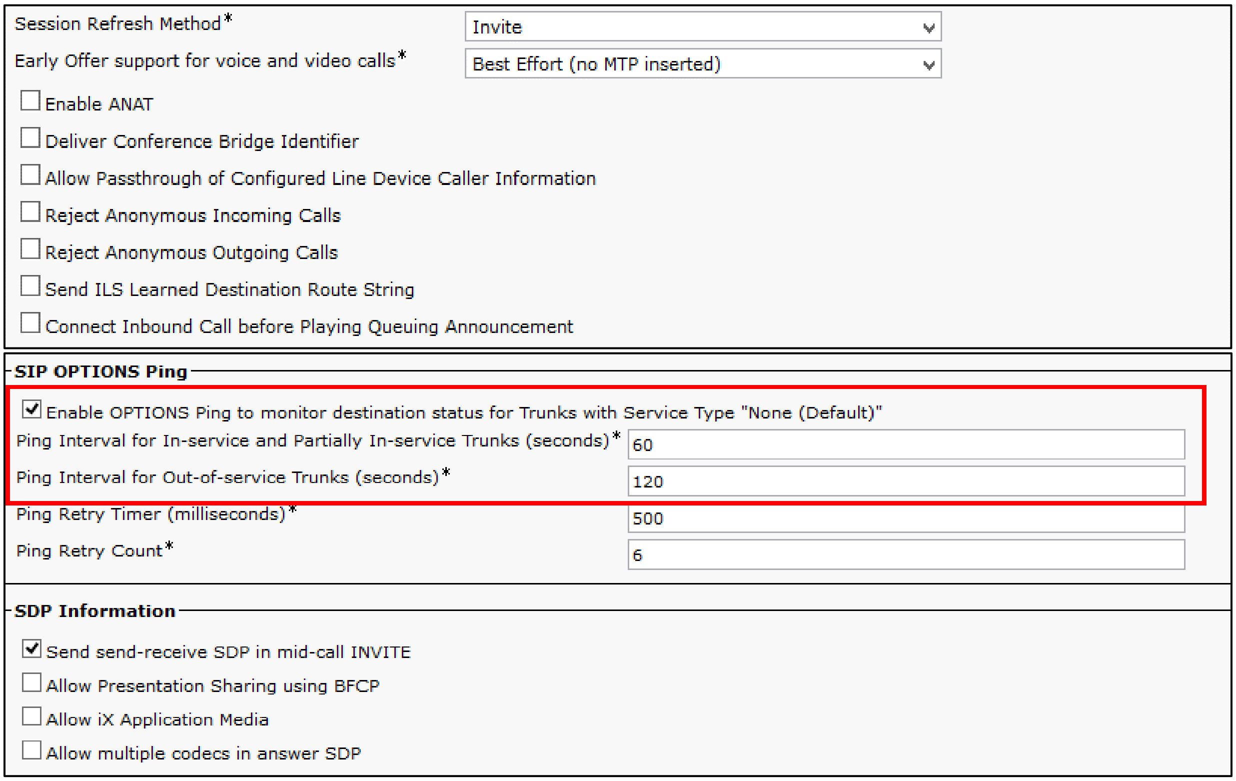


Figure 13: SIP Profile (Cont.)

Explanation

Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
SIP Rel1XX Options	Send PRACK for 1xx Messages	Enable Provisional Acknowledgements (Reliable 100 messages)
Ping Interval for In-service and Partially In-service Trunks (seconds)	60	OPTIONS message parameters- interval time
Ping Interval for Out-of-service Trunks (seconds)	120	OPTIONS message parameters- interval time

SIP Trunk Configuration

Create SIP trunks to CUBE

Navigation: Device → Trunk

Figure 14: SIP Trunks List

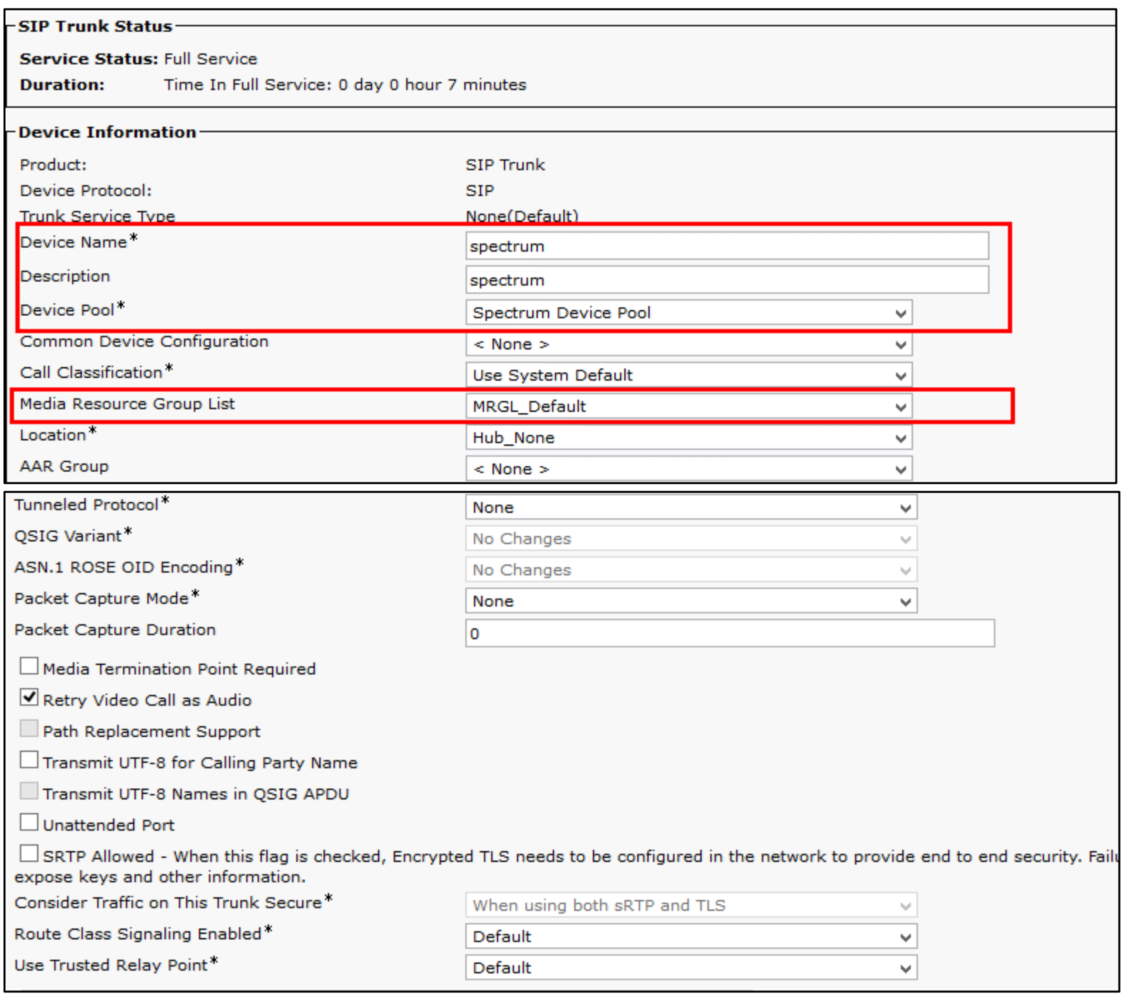


Figure 15: SIP Trunk to CUBE

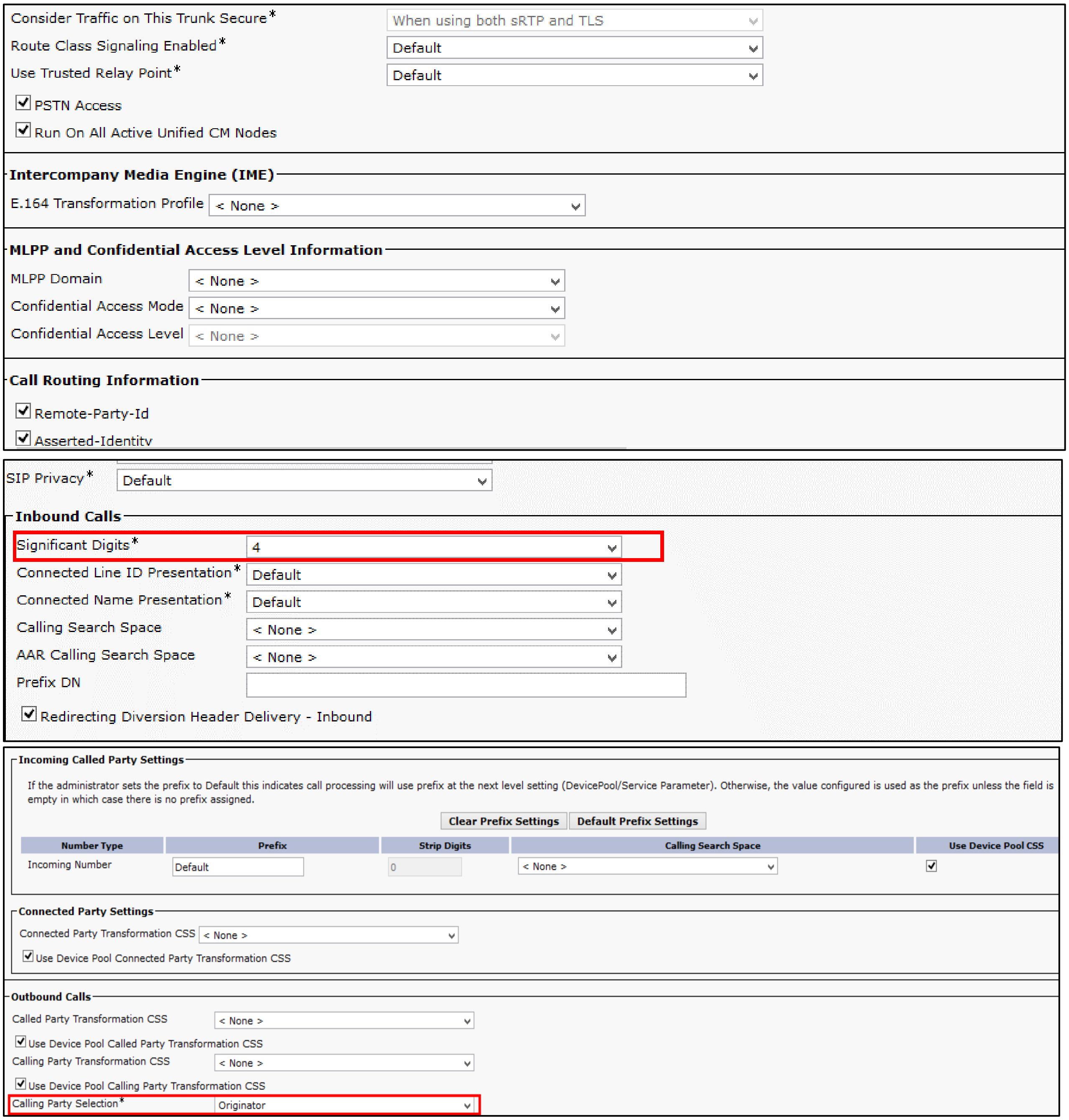


Figure 16: SIP Trunk to CUBE (Cont.)

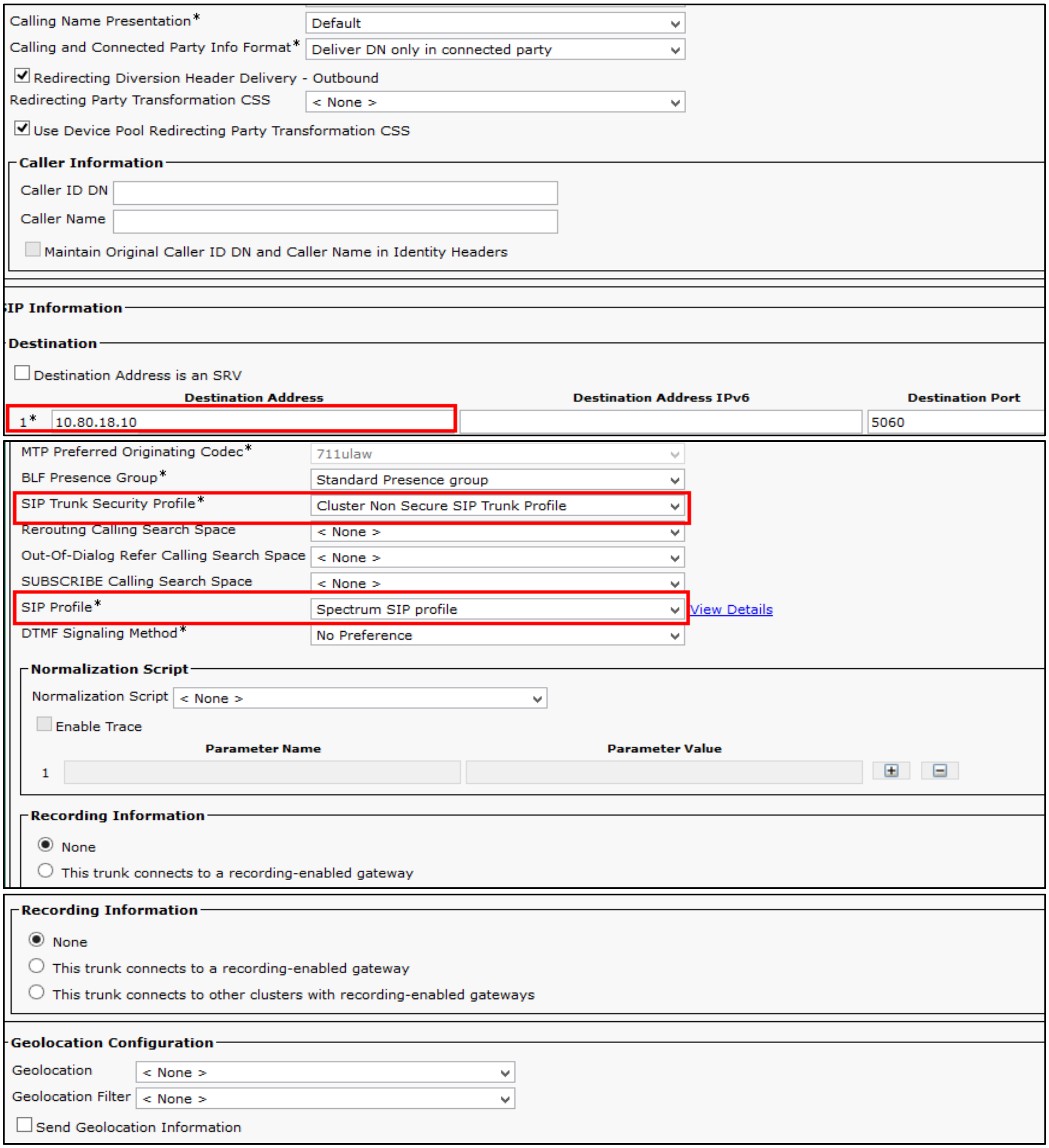


Figure 17: SIP Trunk to CUBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	Spectrum Enterprise	Name for the trunk
Device Pool	G711pool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_MTP	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.18.10	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	Spectrum Enterprise Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Spectrum Enterprise SIP Profile	SIP Profile configured earlier

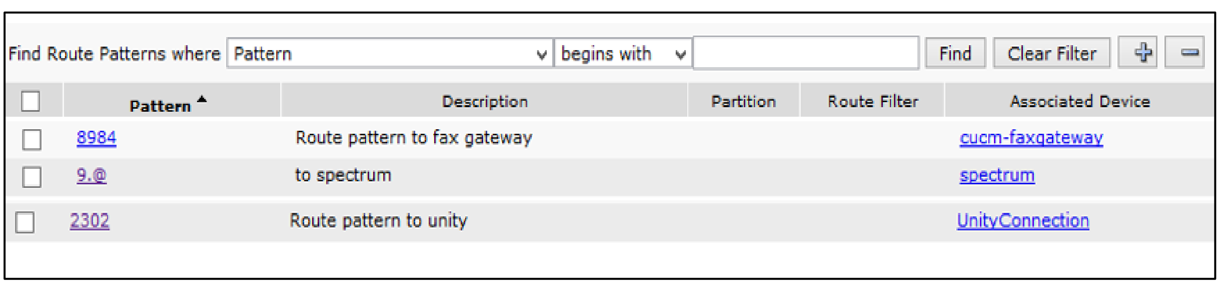
Dial Plan

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial “9”.1+10 digits number to access PSTN via Cisco UBE
  - “9” is removed before sending to Cisco UBE
- For FAX call, Access Code “9”+ 1+10 digits number is used at Cisco Fax gateway
  - “9” is removed at Cisco UCM
  - The rest of the number is sent to Cisco UBE to Spectrum Enterprise network
- Incoming fax call to 2063 will be sent to Cisco Fax gateway



Find Route Patterns where Pattern begins with Find Clear Filter

Pattern	Description	Partition	Route Filter	Associated Device
8984	Route pattern to fax gateway			cucm-faxgateway
9.@	to spectrum			spectrum
2302	Route pattern to unity			UnityConnection

Figure 18: Route Patterns List

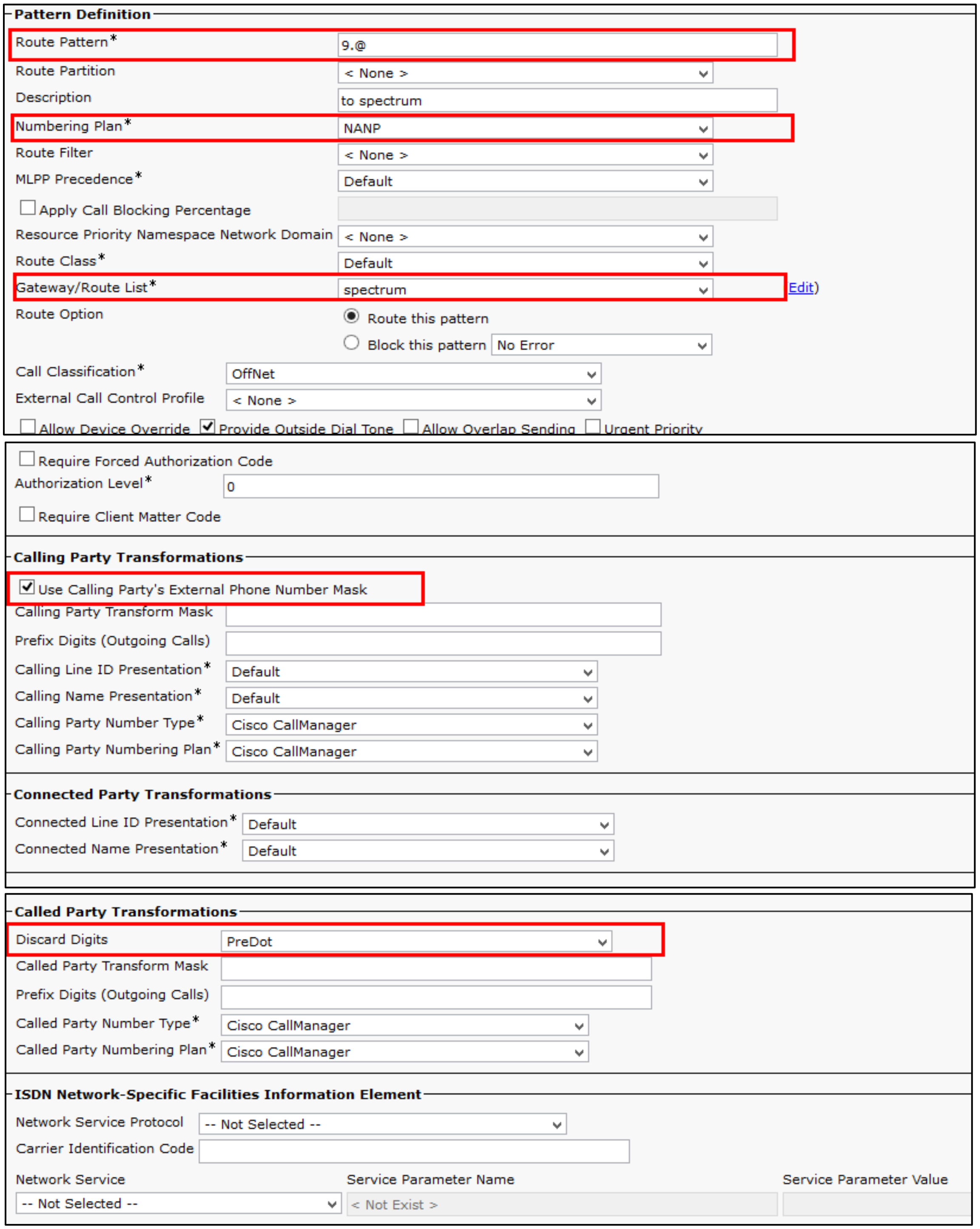


Figure 19: Route Pattern for Voice

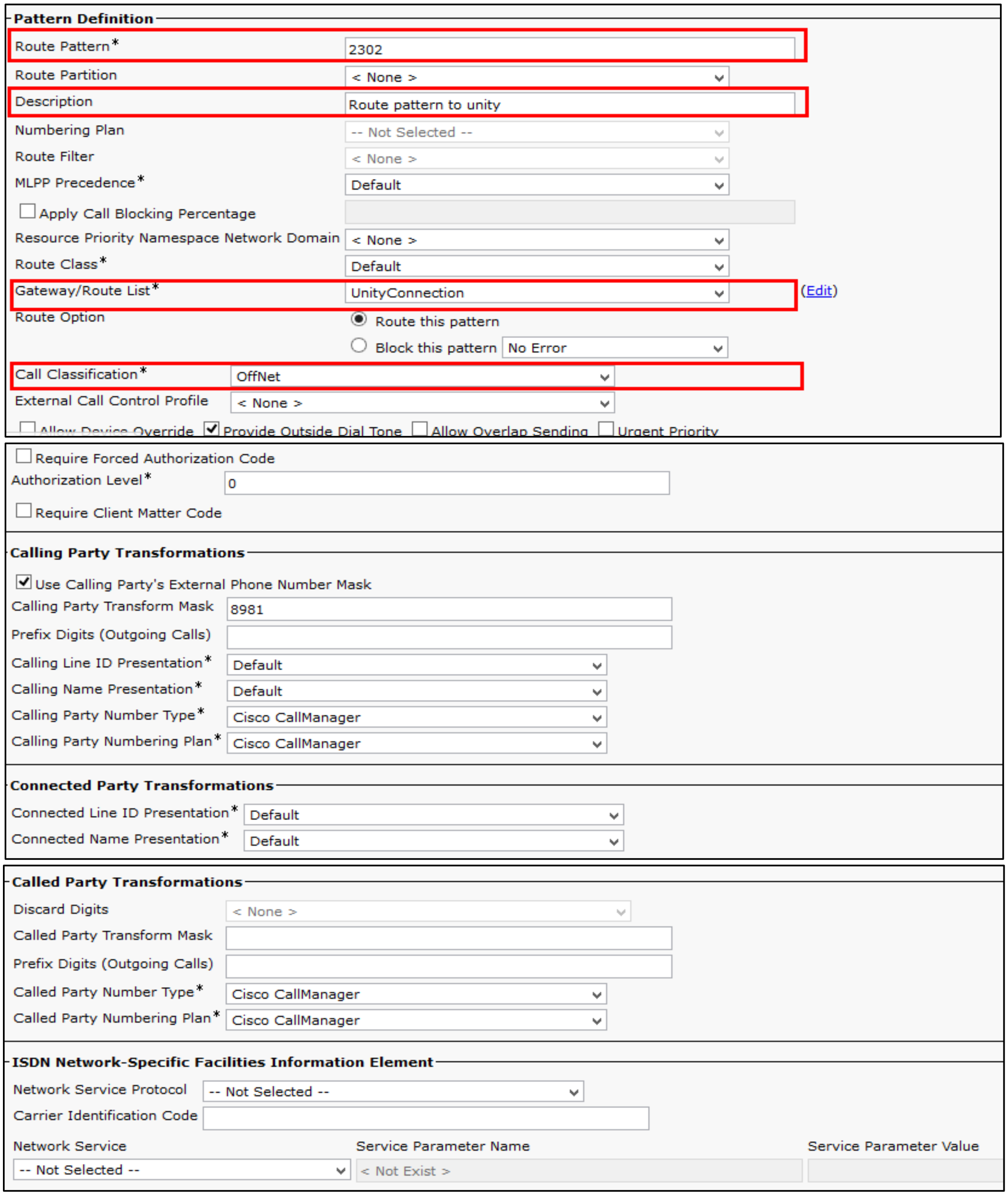


Figure 20: Route Pattern for Voice (Cont.)

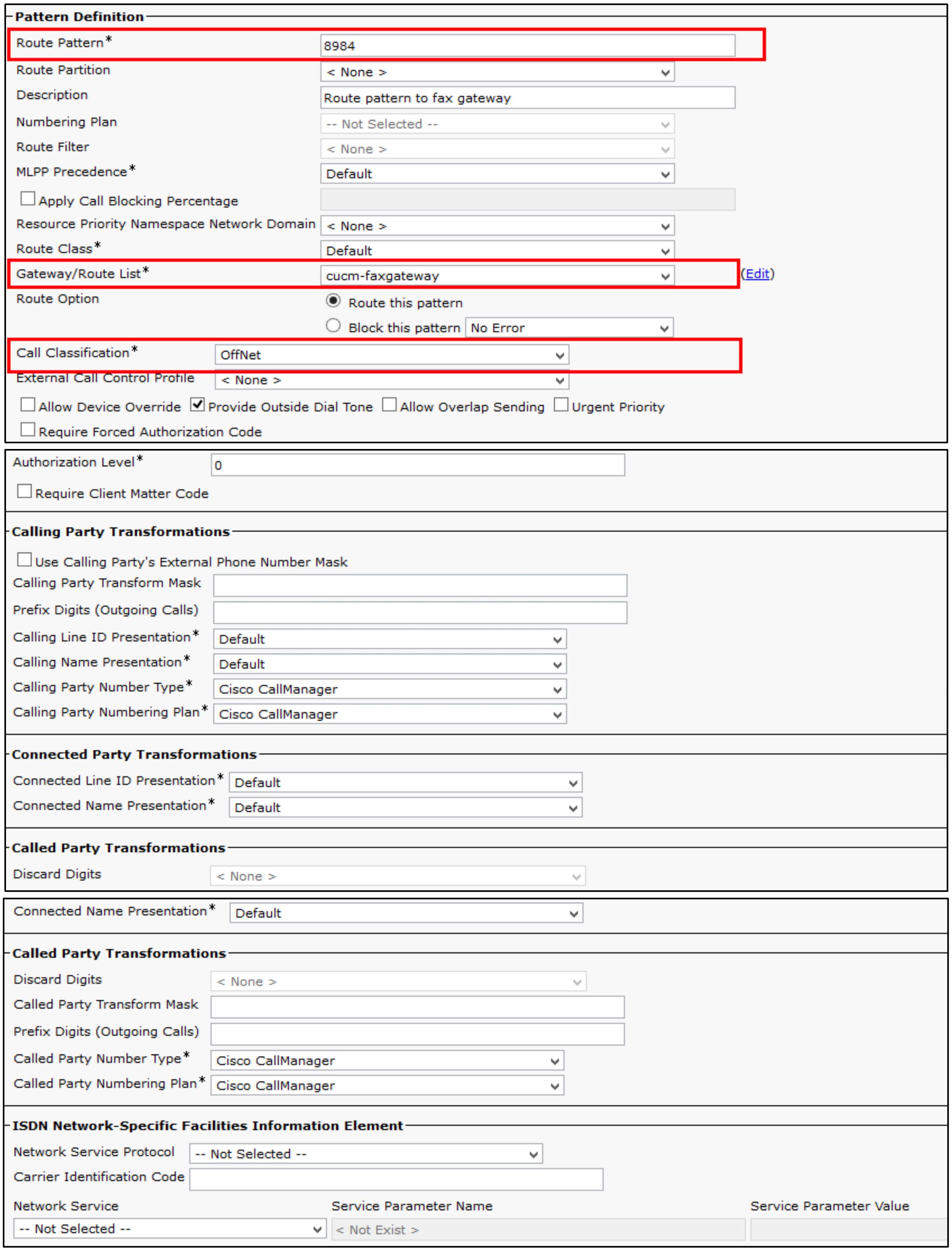


Figure 21: Route Pattern for Fax

Explanation

Setting	Value	Description
Route Pattern	9.@ for Voice & International Calls, 8984 for Fax Call and 2302 for Unity Connection	Specify appropriate Route Pattern
Gateway/Route List	Spectrum Enterprise for Route Pattern 9.@, 8984 for CUCM- Fax Gateway and 2302 for Unity Connection	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 9.@	North American Numbering Plan
Call Classification	OffNet for Route Pattern 9.@, 8984 and 2302	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 9.@	Specifies how to modify digit before they are sent to Spectrum network

Acronyms

Acronym	Definition
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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