Application Note

Spectrum Enterprise SIP Trunking:

Cisco Unified Communications Manager 11.5.1 with Cisco Unified Border Element (CUBE 11.6.0) on ISR 4321/K9 [IOS-XE – 16.5.1b] using SIP

July 15, 2017

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Introduction

Service Providers today, such as Spectrum Enterprise, 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 (Cisco UBE) ISR 4321/K9 running IOS-XE 16.5.1b can be used. The Cisco Unified Border Element provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 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 (Cisco UCM) 11.5.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE 16.5.1b] for connectivity to Spectrum Enterprise SIP Trunking service available in the former Charter Spectrum Business service area[1] (hereafter referred to as Spectrum Enterprise (L-Charter)). The deployment model covered in this application note is CPE (Cisco UCM 11.5.1) 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 (L-Charter) 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 Spectrum Enterprise (L-Charter) SIP Trunking network.

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] Refers to the former Charter Spectrum Business service area for SIP trunking, prior to the acquisition of Time Warner Cable and BrightHouse Networks in 2016.

Network Topology

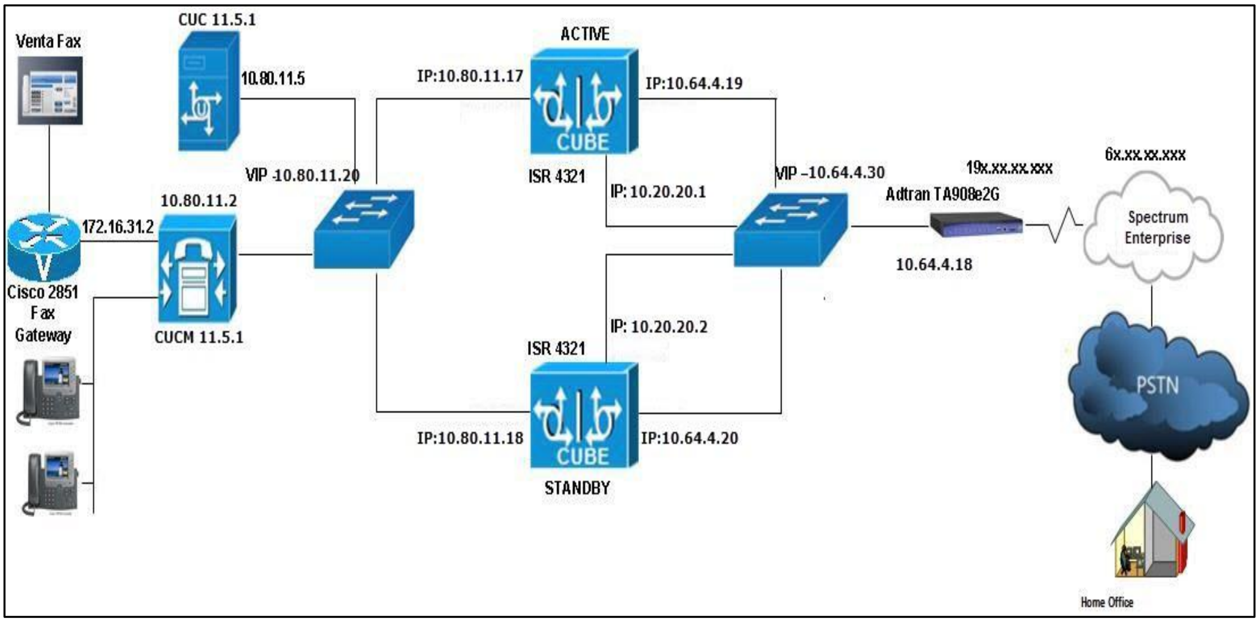

Figure 1: Network Topology

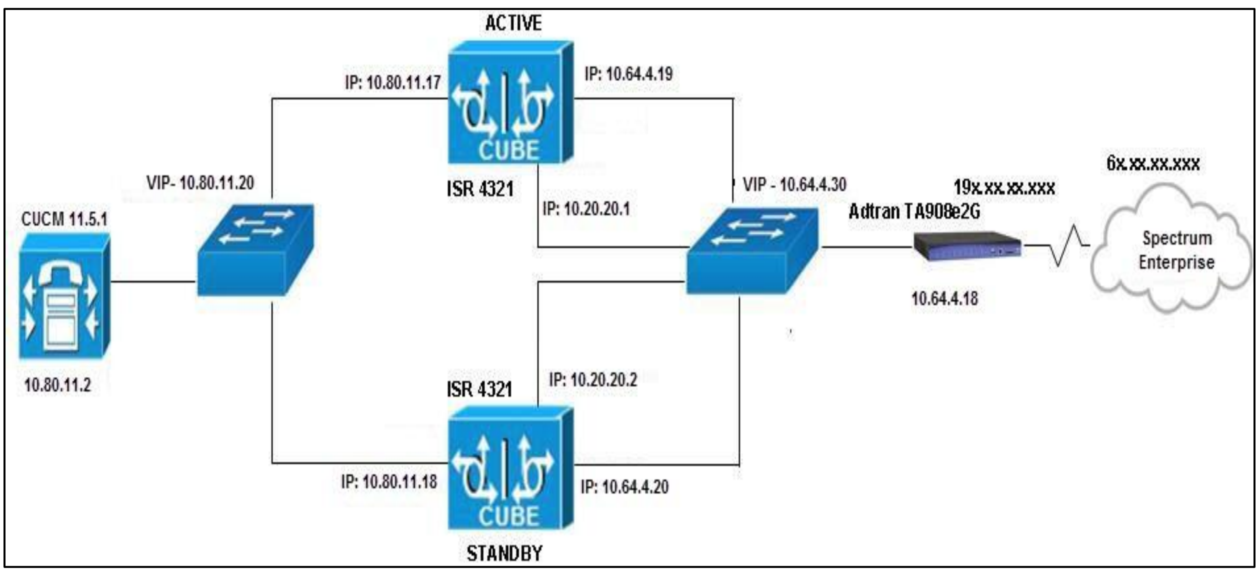

Figure 2: Cisco UBE High Availability

System Components

Hardware Requirements

- Cisco UCSC-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR4431/K9 router as CUBE
- Cisco ISR4431/K9 (1RU) processor with 1684579K/6147K bytes of memory with 4 Gigabit Ethernet interfaces
- Processor board ID FTX1845AJ9S
- Cisco 2851 Fax Gateway
- IP phones 9971 (SIP) and 8945 (SIP)
- Adtran Total_Access_908e_2nd_Gen – Provided and managed by Spectrum Enterprise

Software Requirements

- Cisco Unified Communications Manager 11.5.1
- Cisco Unity Connection 11.5.1
- IOS-XE 16.5.1b for ISR 4321/K9 Cisco Unified Border Element
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway
- Adtran Total_Access_908e_2nd_Gen /R11.4.6.E - Provided and managed by Spectrum Enterprise

Features

Features Supported

- Incoming and outgoing off-net calls using G711ULaw
- Call hold
- Call transfer (unattended and attended)
- Call conference
- Call forward (all, busy and 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 (G.711 pass-through)

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer
- Fax (T.38) and G729 is not supported by Spectrum Enterprise (L-Charter)
- In HA redundancy mode, the primary cube will not take over the primary/active role after a reboot/network outage

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.
- For testing, 911 calls were terminated by Spectrum Enterprise

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 10.64.4.19 255.255.0.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.64.4.30 exclusive
!
interface GigabitEthernet0/0/2
 ip address 10.80.11.17 255.255.255.0
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 10.80.11.20 exclusive
```

Global Cisco UBE Settings

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

```
voice service voip
 no ip address trusted authenticate
 mode border-element
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711ulaw
 sip
  rel1xx supported "rel100"
  header-passing
  asserted-id pai
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
```

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

Cisco UBE uses dial-peers to route the call accordingly based on the digits

```
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 bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 20 voip
 description Outgoing to Adtran
 huntstop
 destination-pattern [0-9]T
 session protocol sipv2
 session target ipv4:10.64.4.18:5060
 voice-class codec 1
 voice-class sip profiles 100
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 30 voip
```

```
 description Incoming from Adtran
 huntstop
 session protocol sipv2
 incoming called-number [37][02][30]T
 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
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 40 voip
 description Outgoing to CUCM
 huntstop
 destination-pattern [37][02][30]T
 session protocol sipv2
 session target ipv4:10.80.11.2
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay ecm disable
 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 200 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.

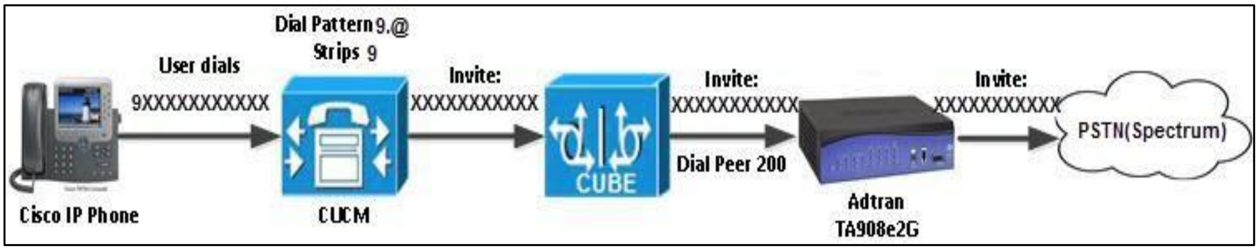

Figure 3: Outbound Voice Call

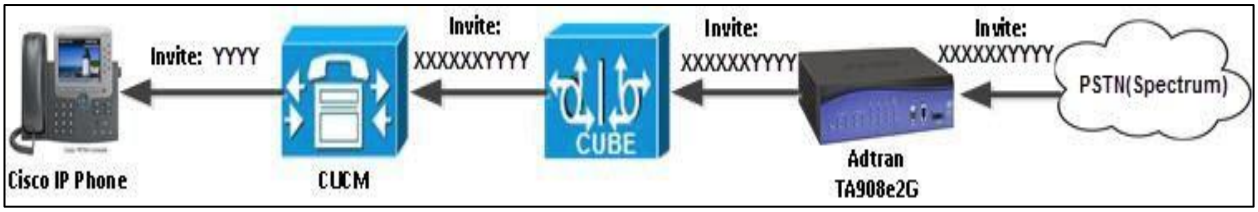

Figure 4: Inbound Voice Call

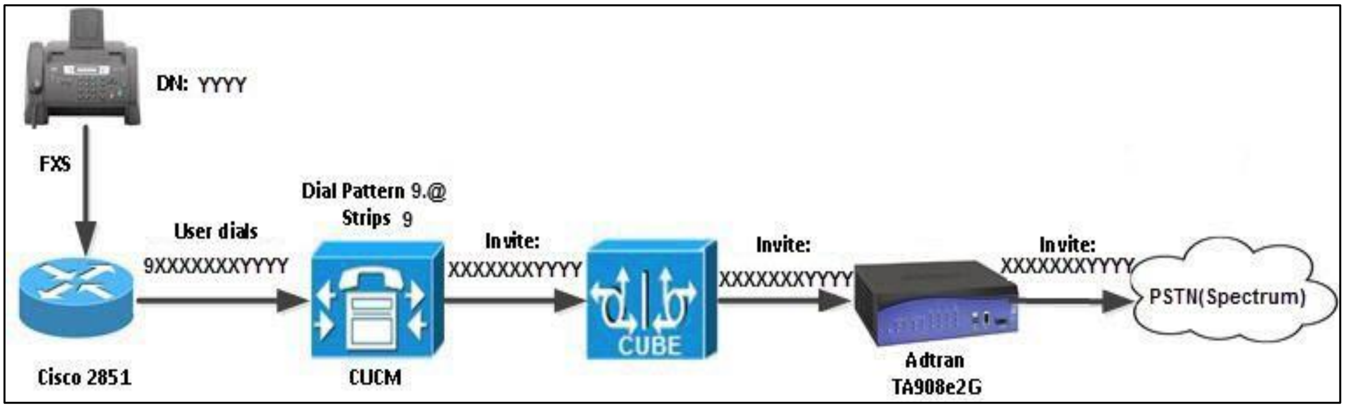

Figure 5: Outbound Fax Call

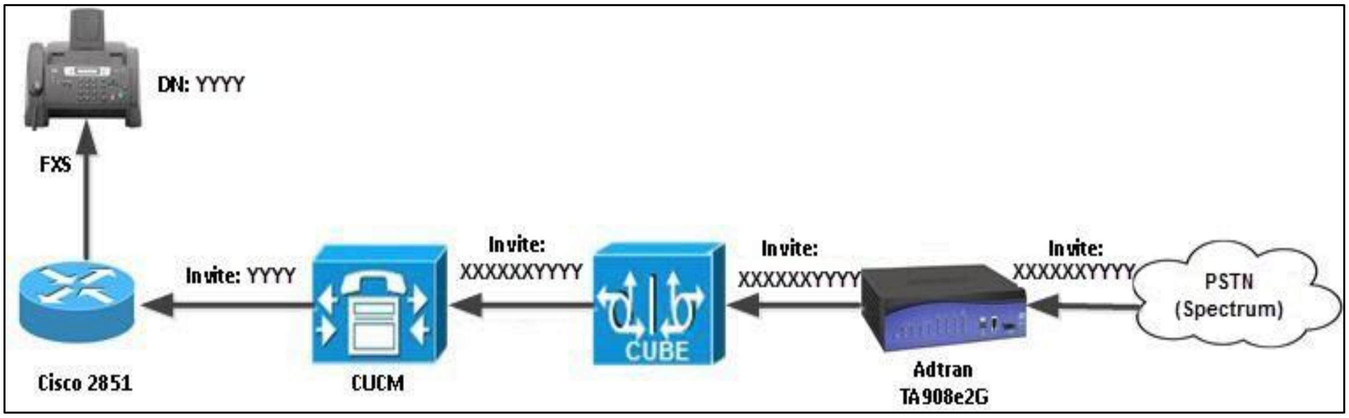

Figure 6: Inbound Fax Call

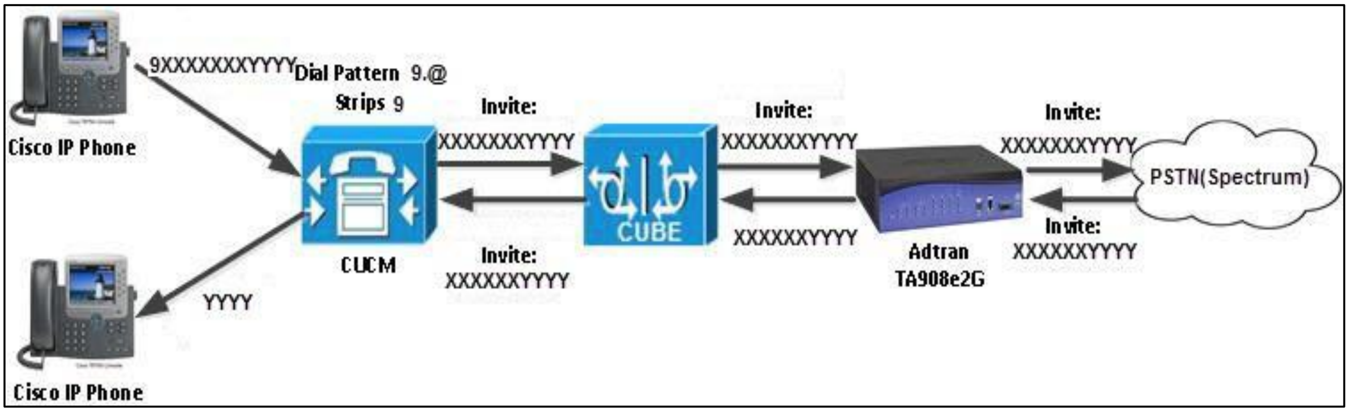

Figure 7: PBX to PBX via Spectrum Call

Configuration Example

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

Active Cisco UBE

User Access Verification

```
   Username:
   Password:
isr4k1spectrum#sh running-config
Building configuration...
Current configuration : 5286 bytes
!
! Last configuration change at 06:17:09 UTC Tue May 30 2017 by cisco
!
version 16.5
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 isr4k1spectrum
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging console emergencies
enable secret 5 $1$JbZh$5I3Lr7oSNCAEcKgkNMyGP0
!
no aaa new-model
```

```
!
ipv6 unicast-routing
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-1179880555
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-1179880555
 revocation-check none
 rsakeypair TP-self-signed-1179880555
!
crypto pki certificate chain TP-self-signed-1179880555
!
voice service voip
 no ip address trusted authenticate
 mode border-element
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711ulaw
 sip
  rel1xx supported "rel100"
  header-passing
  asserted-id pai
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice class sip-profiles 100
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:303835\1@\2"
!
voice-card 0/1
 no watchdog
```

```
!
license udi pid ISR4431/K9 sn FOC18261KJL
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level uck9
license boot level securityk9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 09584B022F540D435B02
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   timers delay 30
   control GigabitEthernet0/0/3 protocol 1
   data GigabitEthernet0/0/3
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/2 line-protocol
!
interface GigabitEthernet0/0/0
 ip address 10.64.4.19 255.255.0.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.64.4.30 exclusive
!
interface GigabitEthernet0/0/1
 no ip address
 negotiation auto
!
```

```
interface GigabitEthernet0/0/2
 ip address 10.80.11.17 255.255.255.0
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 10.80.11.20 exclusive
!
interface GigabitEthernet0/0/3
 ip address 10.20.20.1 255.255.255.0
 negotiation auto
!
interface Service-Engine0/1/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
threat-visibility
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
control-plane
!
voice-port 0/1/0
!
voice-port 0/1/1
 !
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 bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 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 Adtran
 huntstop
 destination-pattern [0-9]T
 session protocol sipv2
 session target ipv4:10.64.4.18:5060
 voice-class codec 1
 voice-class sip profiles 100
 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 Adtran
 huntstop
```

```
 session protocol sipv2
 incoming called-number [37][02][30]T
 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 [37][02][30]T
 session protocol sipv2
 session target ipv4:10.80.11.2
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay ecm disable
 no fax-relay sg3-to-g3
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
sip-ua
!
line con 0
 exec-timeout 0 0
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 login local
```

Standby Cisco UBE

User Access Verification

Username:

Password:

```
isr4k2spectrum#sh running-config
Building configuration...
Current configuration : 5218 bytes
!
! Last configuration change at 06:20:11 UTC Tue May 30 2017 by cisco
!
version 16.5
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 isr4k2spectrum
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging console emergencies
enable secret 5 $1$7YMA$OqK8fiN9WnsjC82D73OUY/
!
no aaa new-model
!
ipv6 unicast-routing
!
subscriber templating
!
```

```
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-988930787
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-988930787
 revocation-check none
 rsakeypair TP-self-signed-988930787
!
crypto pki certificate chain TP-self-signed-988930787
!
voice service voip
 no ip address trusted authenticate
 mode border-element
 allow-connections sip to sip
 redundancy-group 1
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  rel1xx supported "rel100"
  header-passing
  asserted-id pai
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice class sip-profiles 100
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:303835\1@\2"
!
license udi pid ISR4431/K9 sn FOC18232988
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level uck9
license boot level securityk9
!
diagnostic bootup level minimal
spanning-tree extend system-id
```

```
!
username cisco privilege 15 password 7 051F0304171D5458490B
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   timers delay 30
   control GigabitEthernet0/0/3 protocol 1
   data GigabitEthernet0/0/3
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/2 line-protocol
!
interface GigabitEthernet0/0/0
 ip address 10.64.4.20 255.255.0.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.64.4.30 exclusive
!
interface GigabitEthernet0/0/1
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/2
 ip address 10.80.11.18 255.255.255.0
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 10.80.11.20 exclusive
!
interface GigabitEthernet0/0/3
 ip address 10.20.20.2 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
```

```
 no ip address
 negotiation auto
!
threat-visibility
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 10 voip
 description Incoming from CUCM
 huntstop
 session protocol sipv2
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 20 voip
 description Outgoing to Adtran
 huntstop
```

```
 destination-pattern [0-9]T
 session protocol sipv2
 session target ipv4:10.64.4.18:5060
 voice-class codec 1
 voice-class sip profiles 100
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 30 voip
 description Incoming from Adtran
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 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
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 40 voip
 description Outgoing to CUCM
 huntstop
 destination-pattern [37][02][30]T
 session protocol sipv2
 session target ipv4:10.80.11.2
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
```

```
 no vad
!
sip-ua
!
line con 0
 exec-timeout 0 0
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 login local
```

Configuring Cisco Unified Communications Manager

Cisco UCM Version

Figure 8: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System > Service Parameters

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

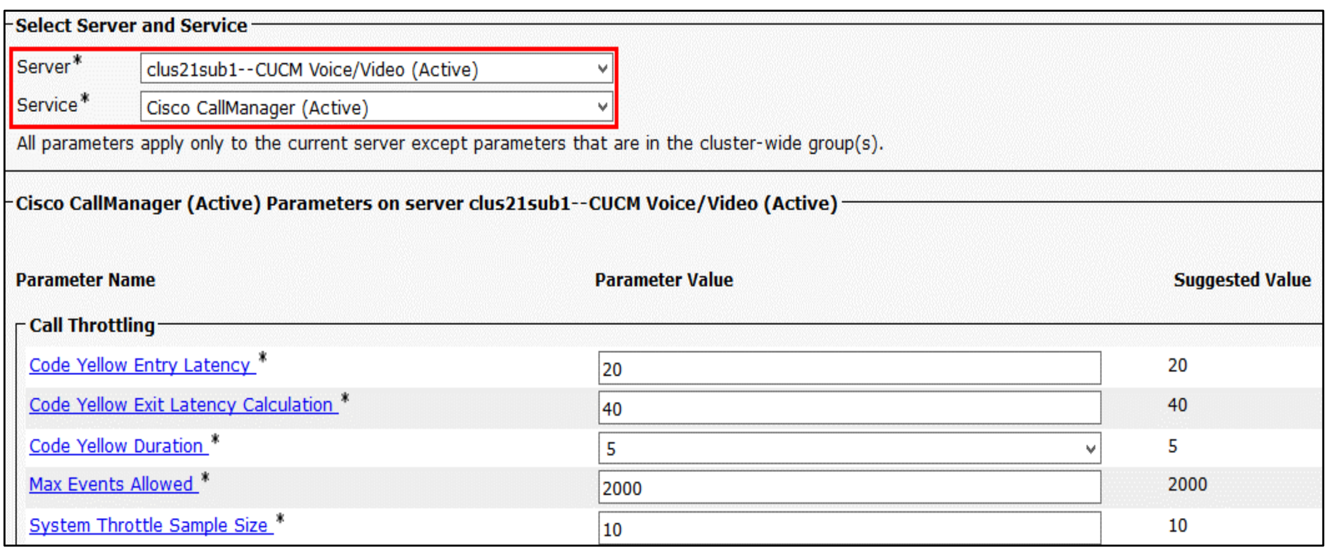


Figure 9: Service Parameters

Offnet Calls via Spectrum Enterprise SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and the Spectrum Enterprise network and calls are routed via Cisco UBE

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

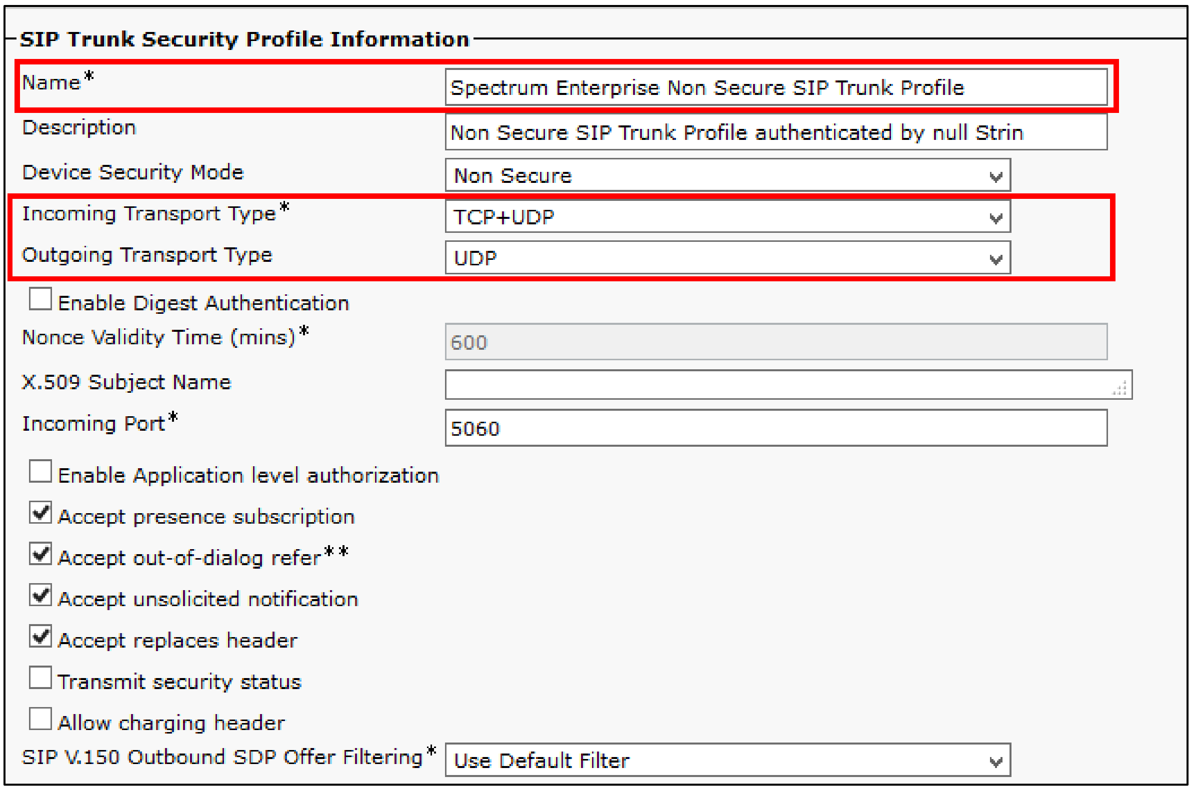


Figure 10: SIP Trunk Security Profile

Explanation

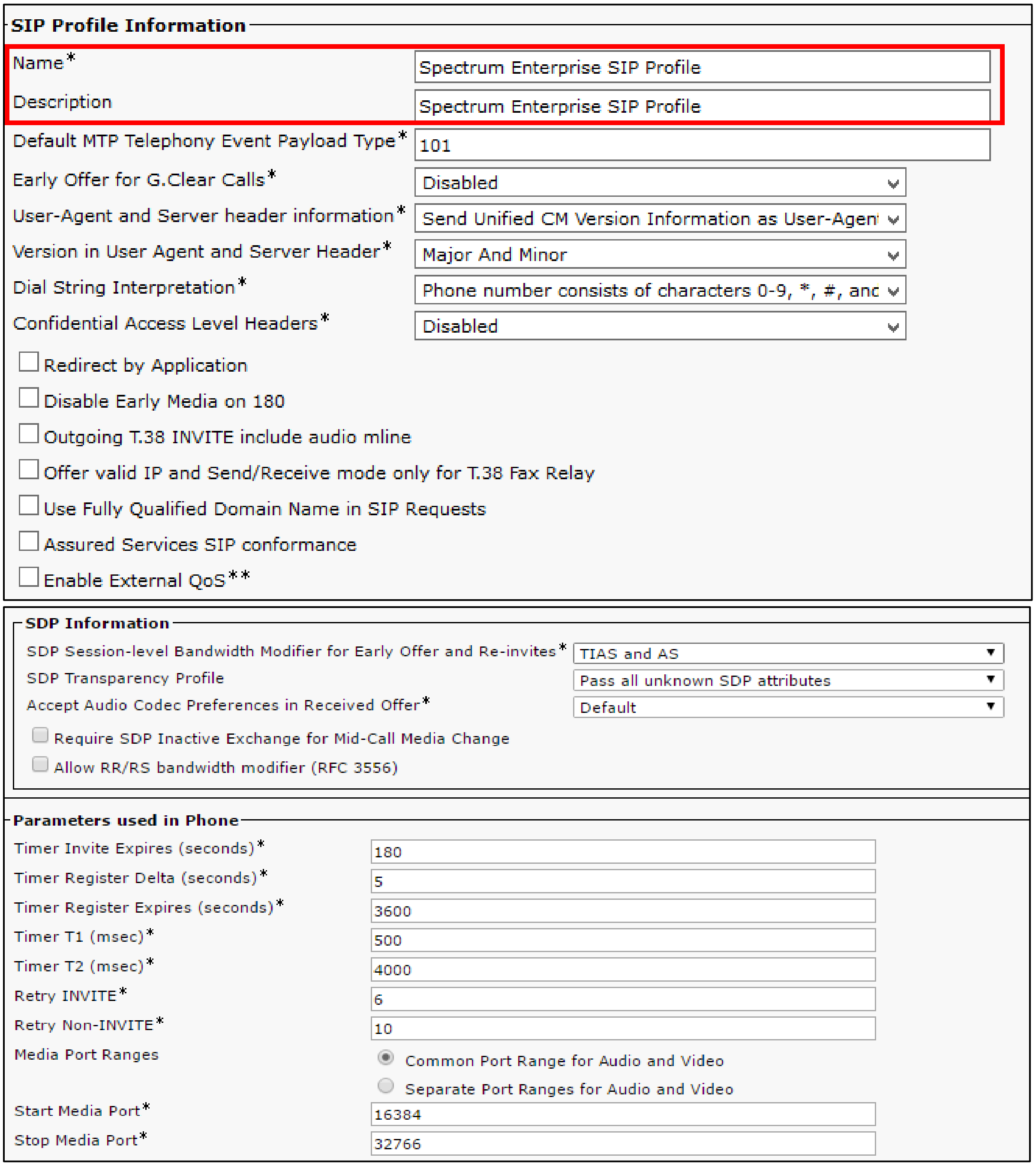
Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to Spectrum Enterprise SBC should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.

SIP Profile Configuration

SIP Profile will be later associated with the SIP trunk

Navigation: Device → Device Settings → SIP Profile

1. Name*= Spectrum Enterprise SIP Profile
2. Description = Spectrum Enterprise SIP Profile



SIP Profile Information

Name*	Spectrum Enterprise SIP Profile
Description	Spectrum Enterprise SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agen
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, anc
Confidential Access Level Headers*	Disabled

- [ ] Redirect by Application
- [ ] Disable Early Media on 180
- [ ] Outgoing T.38 INVITE include audio mline
- [ ] Offer valid IP and Send/Receive mode only for T.38 Fax Relay
- [ ] Use Fully Qualified Domain Name in SIP Requests
- [ ] Assured Services SIP conformance
- [ ] Enable External QoS**

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default

- [ ] Require SDP Inactive Exchange for Mid-Call Media Change
- [ ] Allow RR/RS bandwidth modifier (RFC 3556)

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

Figure 11: SIP Profile

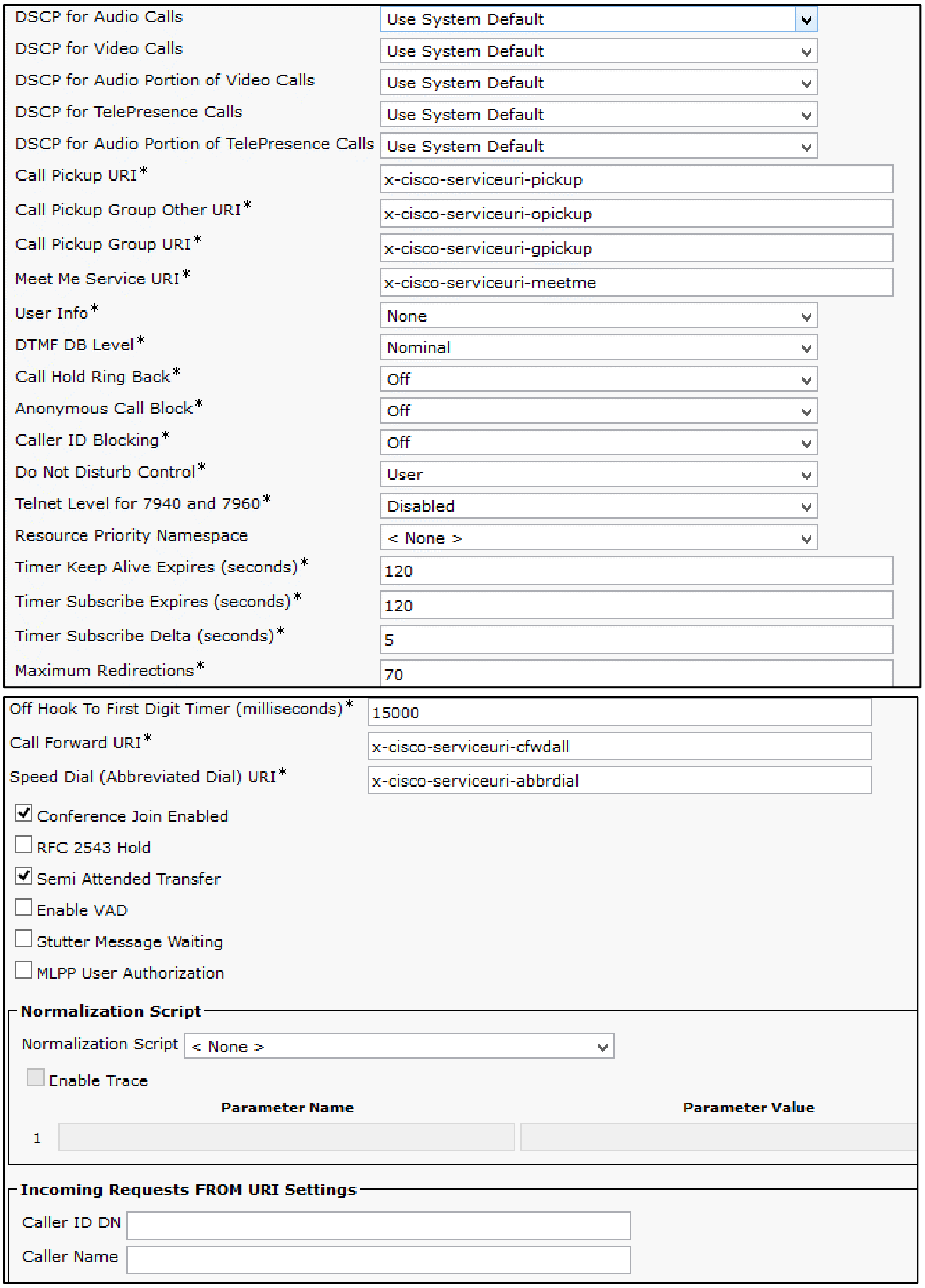


Figure 12: SIP Profile (Cont.)

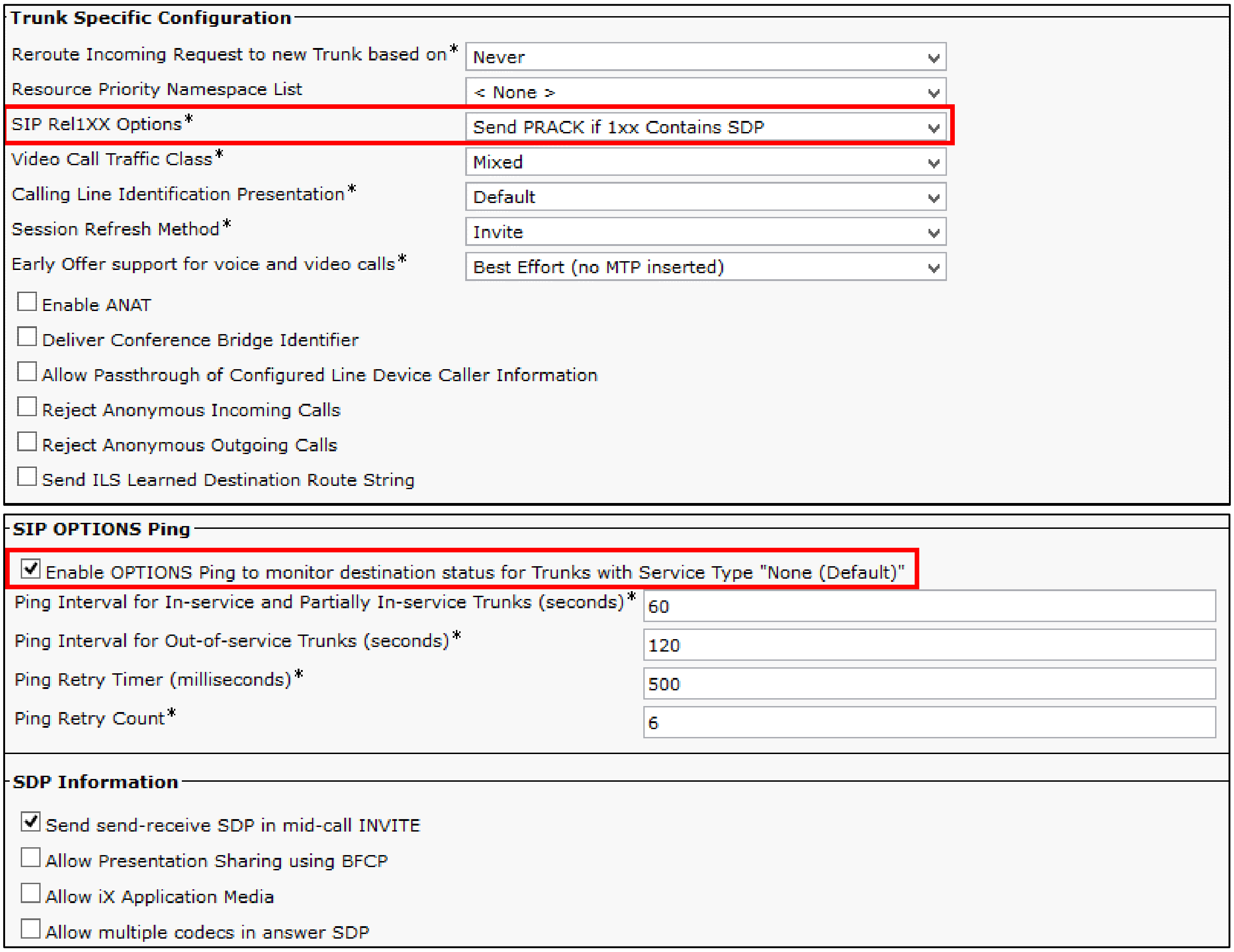


Figure 13: SIP Profile (Cont.)

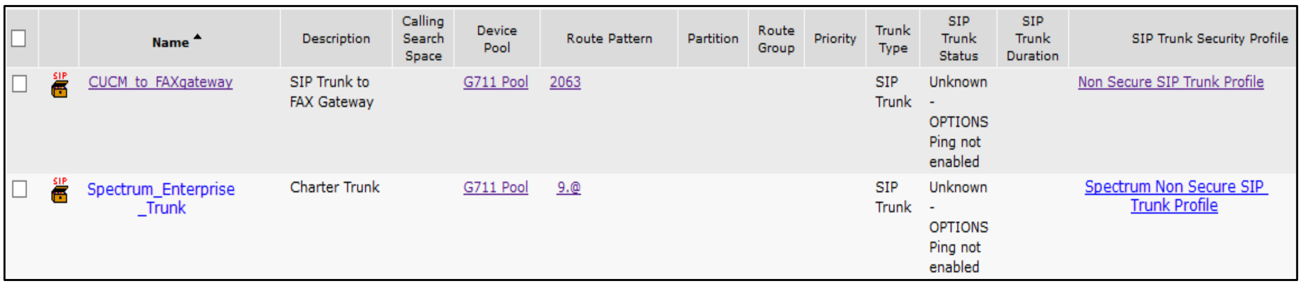
Explanation

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 Cisco UBE

Navigation: Device → Trunk



		Name ▲	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
☐	SIP	CUCM_to_FAXgateway	SIP Trunk to FAX Gateway		G711 Pool	2063				SIP Trunk	Unknown - OPTIONS Ping not enabled		Non Secure SIP Trunk Profile
☐	SIP	Spectrum_Enterprise_Trunk	Charter Trunk		G711 Pool	9.@				SIP Trunk	Unknown - OPTIONS Ping not enabled		Spectrum Non Secure SIP Trunk Profile

Figure 14: SIP Trunks List

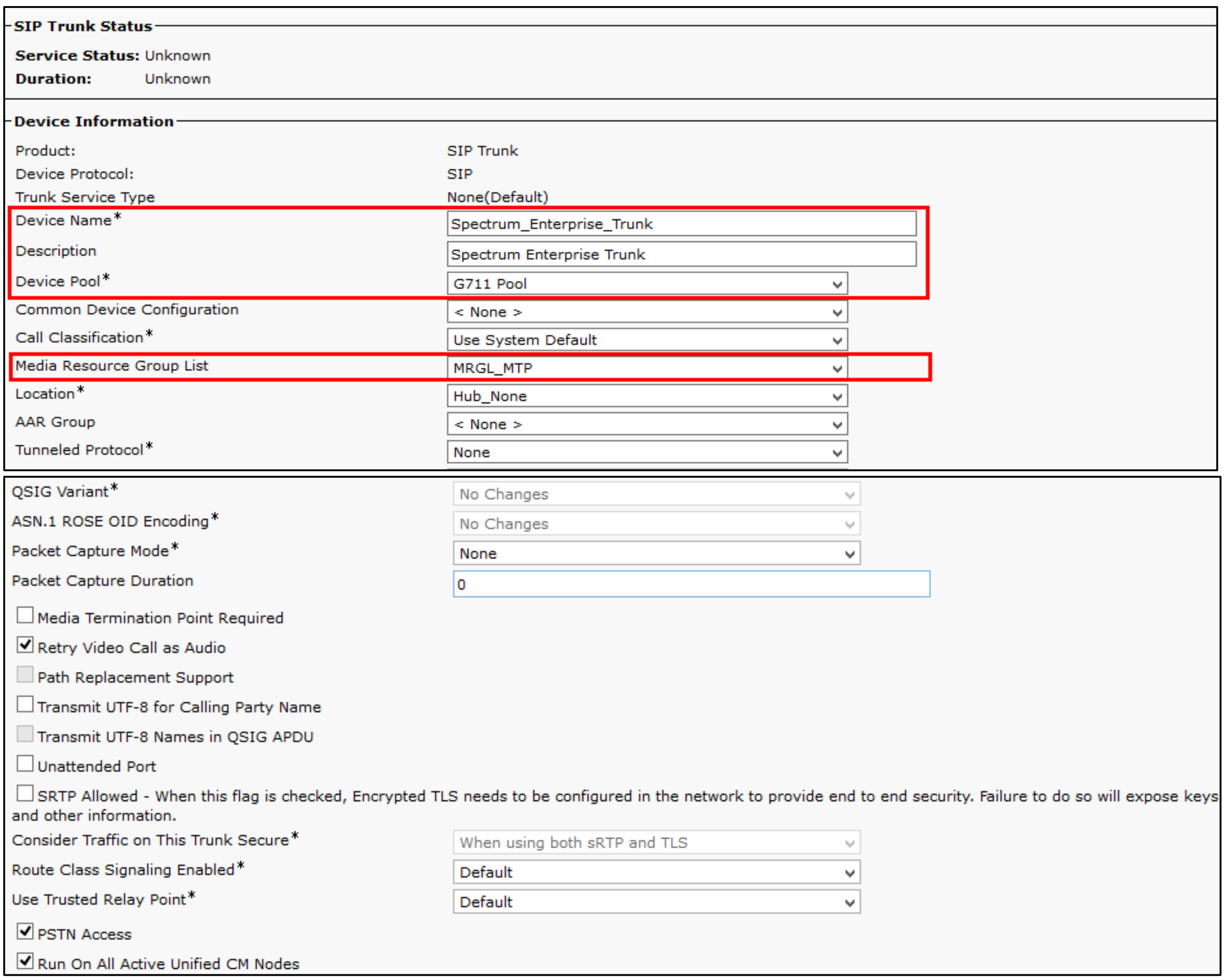


Figure 15: SIP Trunk to Cisco UBE

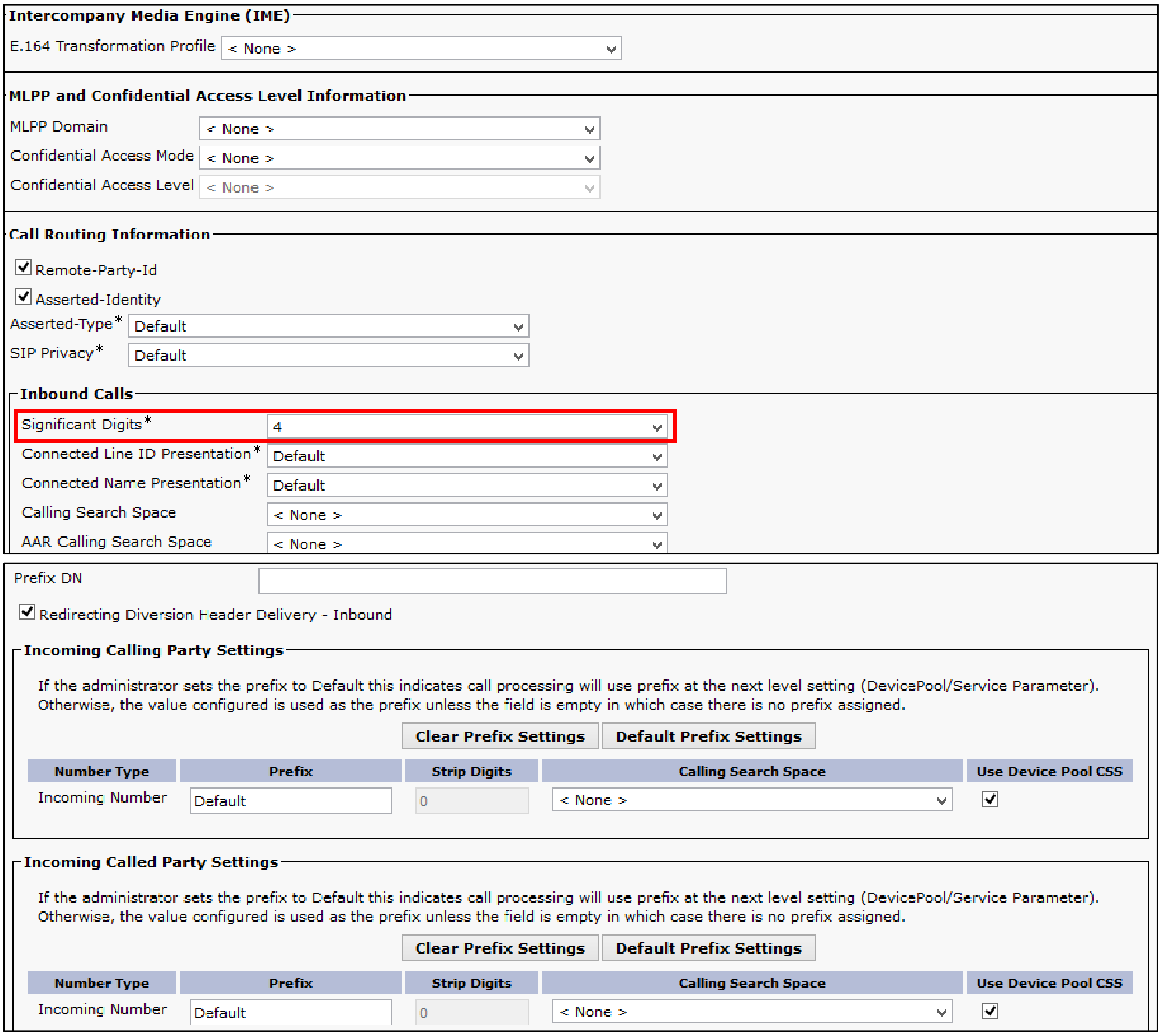


Figure 16: SIP Trunk to Cisco UBE (Cont.)

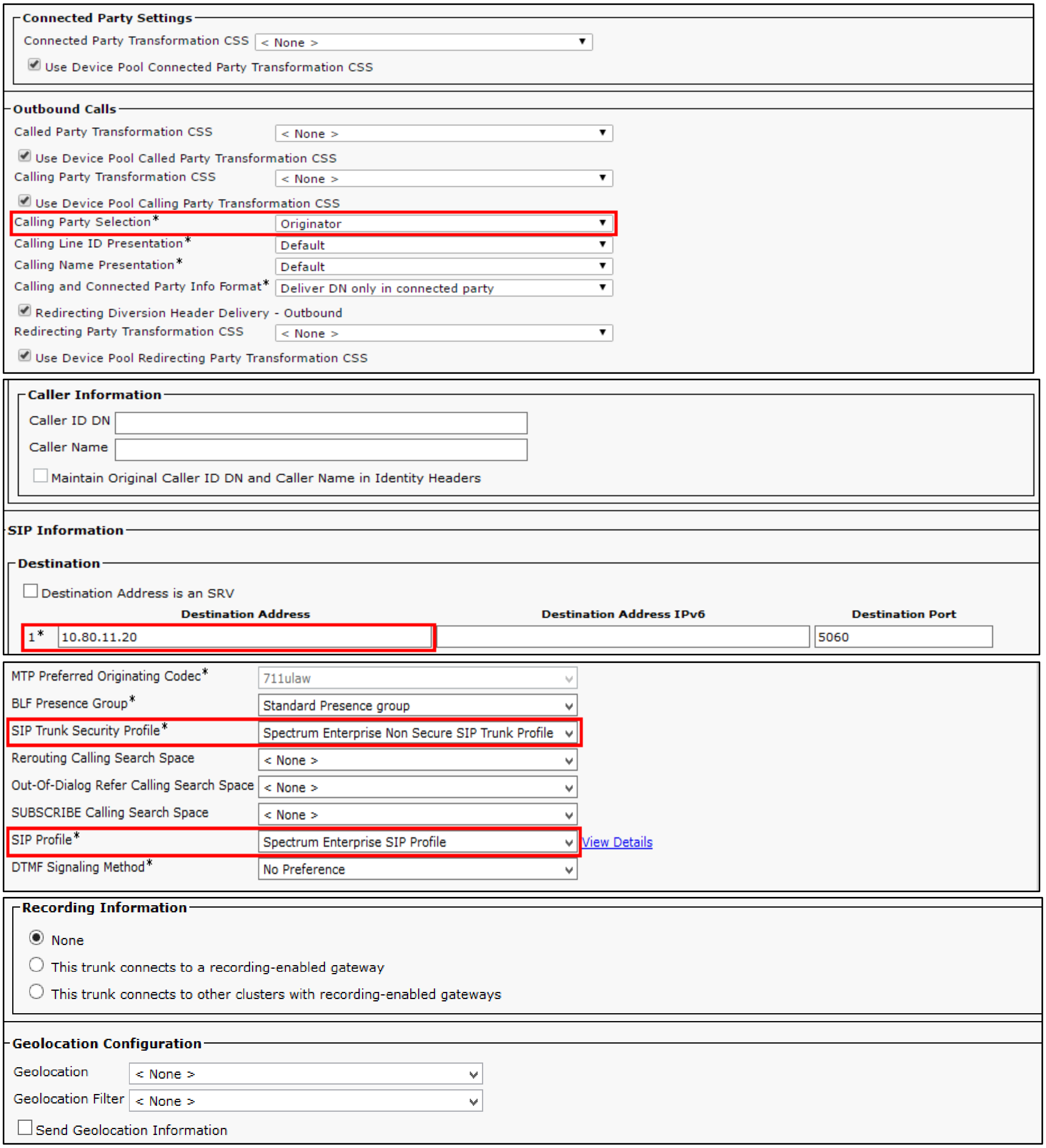


Figure 17: SIP Trunk to Cisco UBE (Cont.)

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	10.80.11.20	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

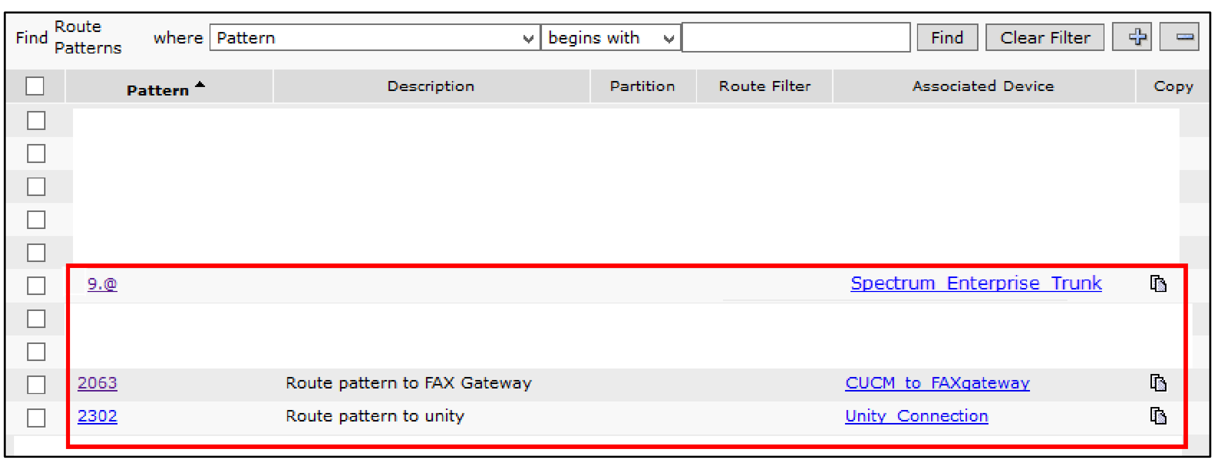


Figure 18: Route Patterns List

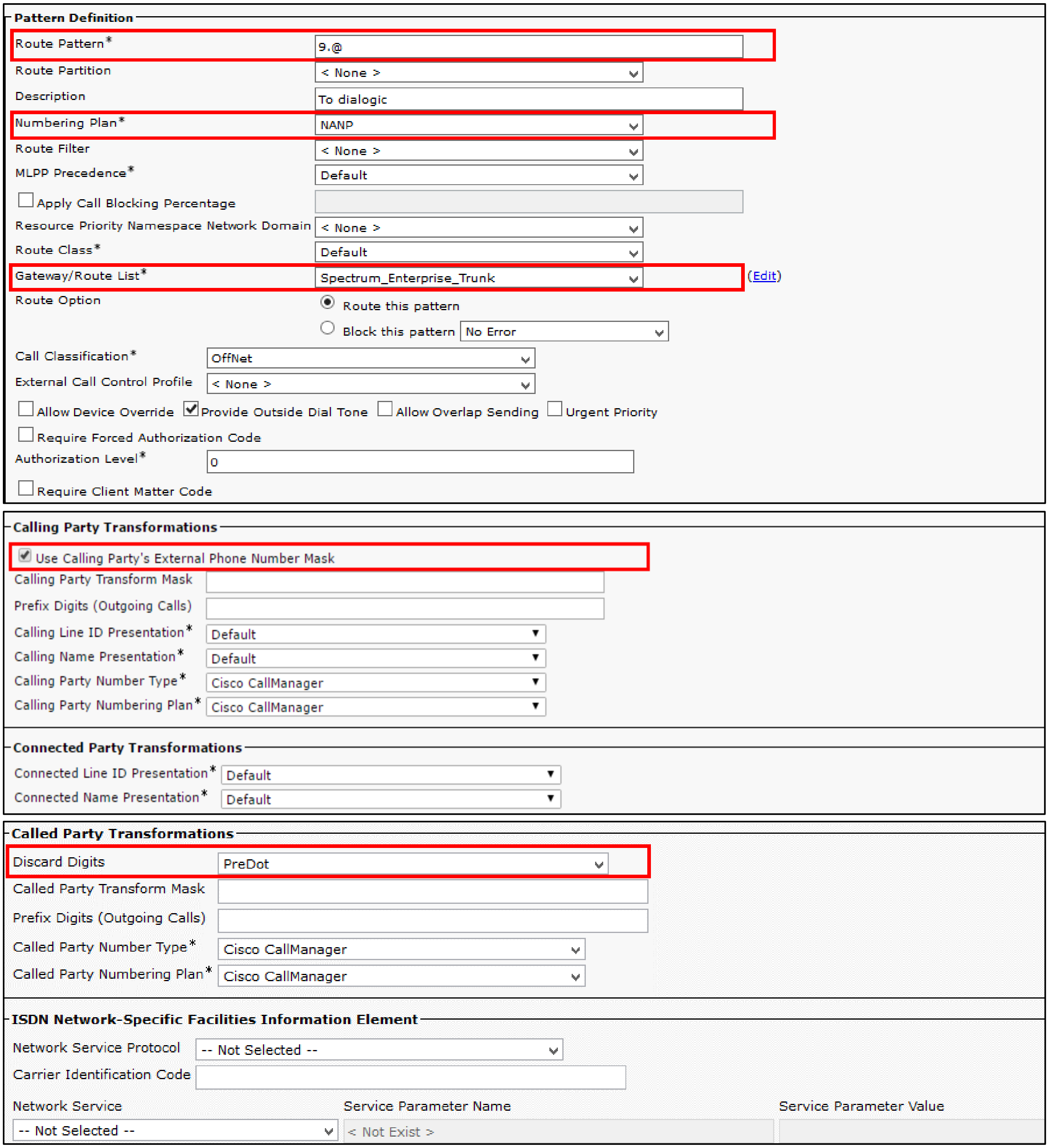


Figure 19: Route Pattern for Voice

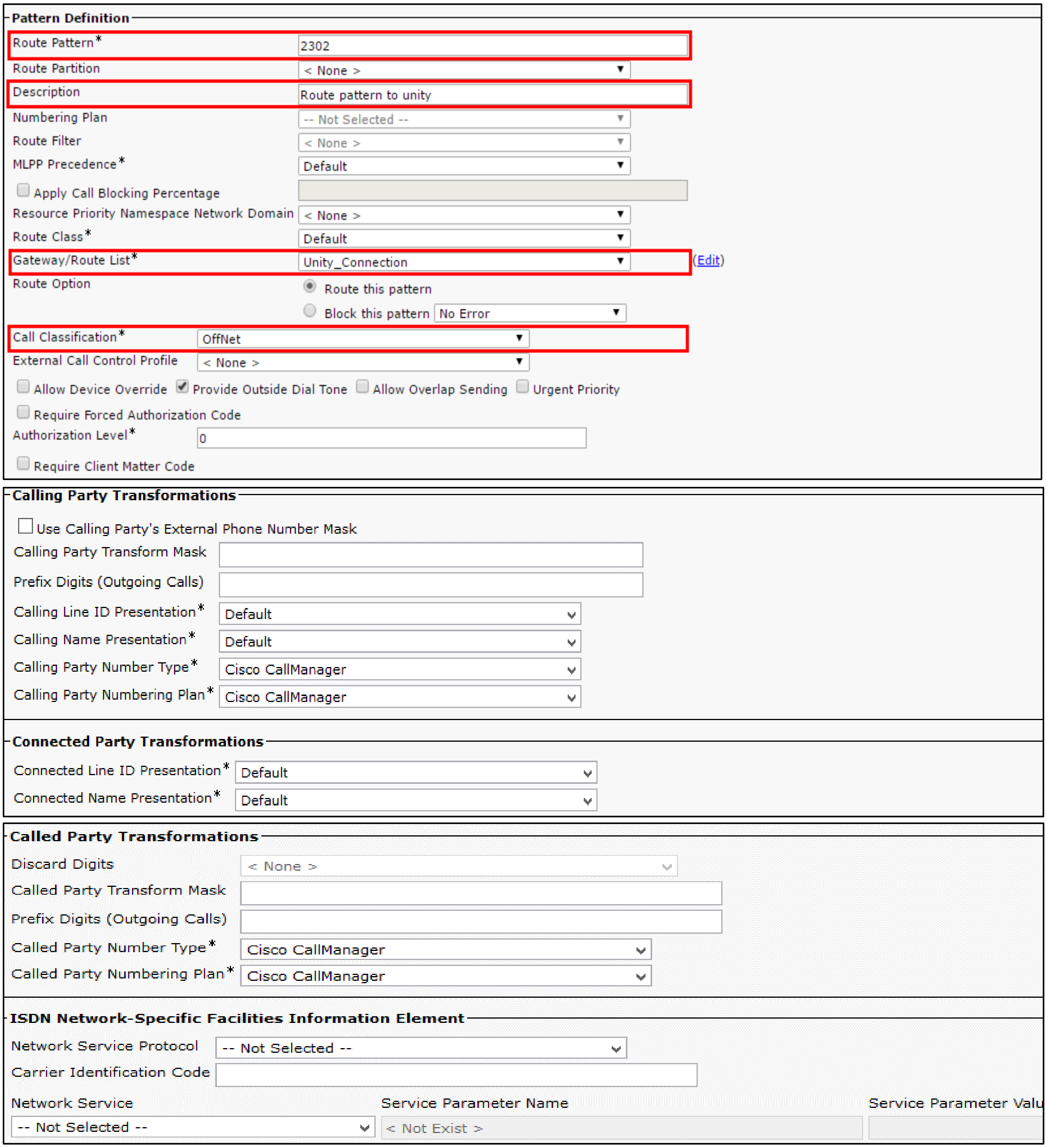


Figure 20: Route Pattern for Voice (Cont.)

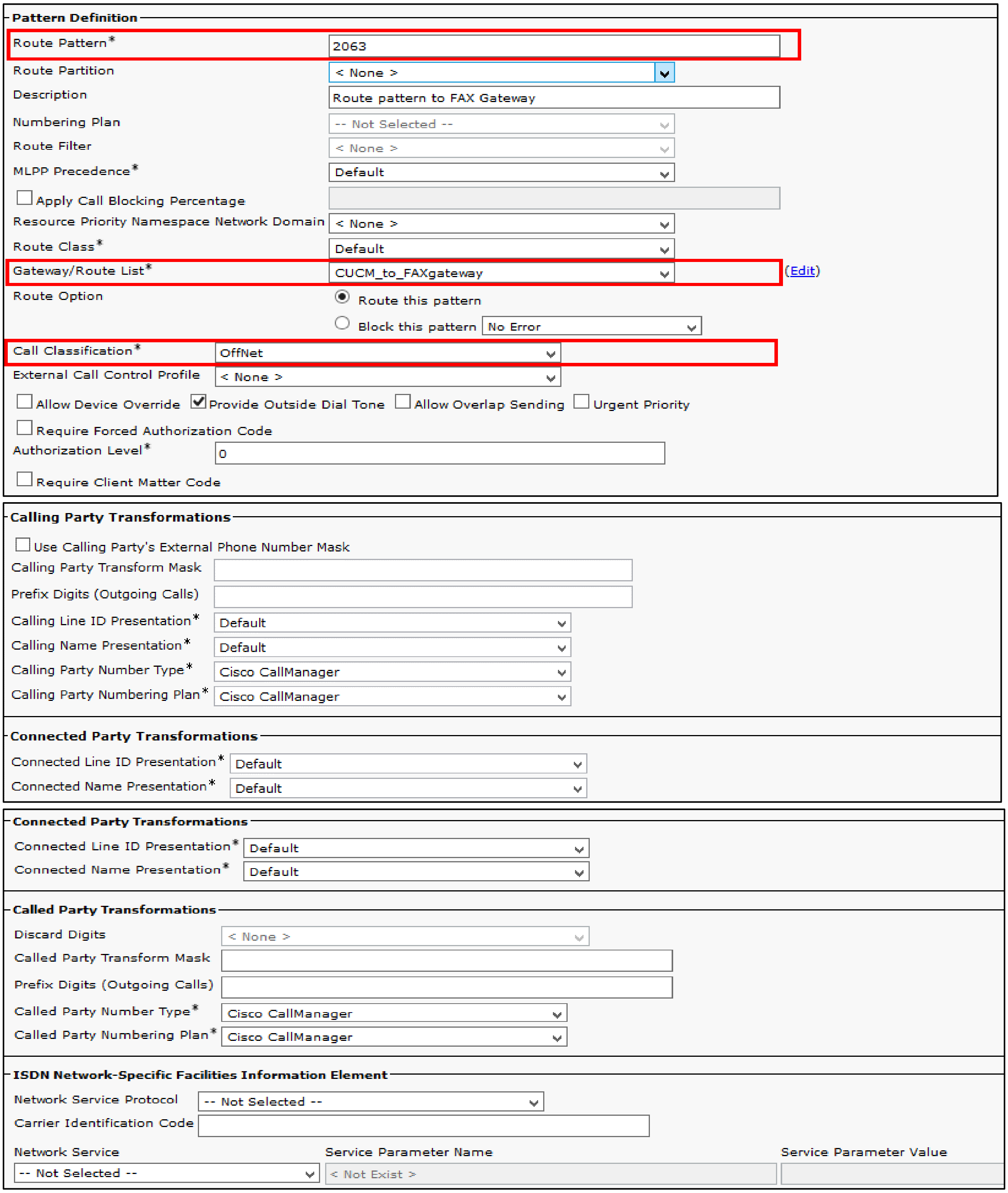


Figure 21: Route Pattern for Fax

Explanation

Setting	Value	Description
Route Pattern	9.@ for Voice & International Calls, 2063 for Fax Call and 2302 for Unity Connection	Specify appropriate Route Pattern
Gateway/Route List	Spectrum Enterprise for Route Pattern 9.@, 2063 for SIP Trunk To Fax Gateway and 2032 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.@, 2063 and 2032	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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