Application Note

Spectrum Enterprise SIP Trunking:

Cisco Unified Communications Manager 12.0.x with Cisco Unified Border Element (CUBE 12.0.0) on ISR 4321/K9 [IOS-XE – 16.6.1] using SIP

January 22, 2018

Table of Contents

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 12.0.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.0.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) 12.0.x and Cisco Unified Border Element (CUBE) on ISR 4321/K9 [IOS-XE – 16.6.1] for connectivity to Spectrum Enterprise SIP Trunking services. The deployment model covered in this application note is CPE (CUCM 12.0.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 (TWC) and acquisition of Bright House Networks.

Network Topology

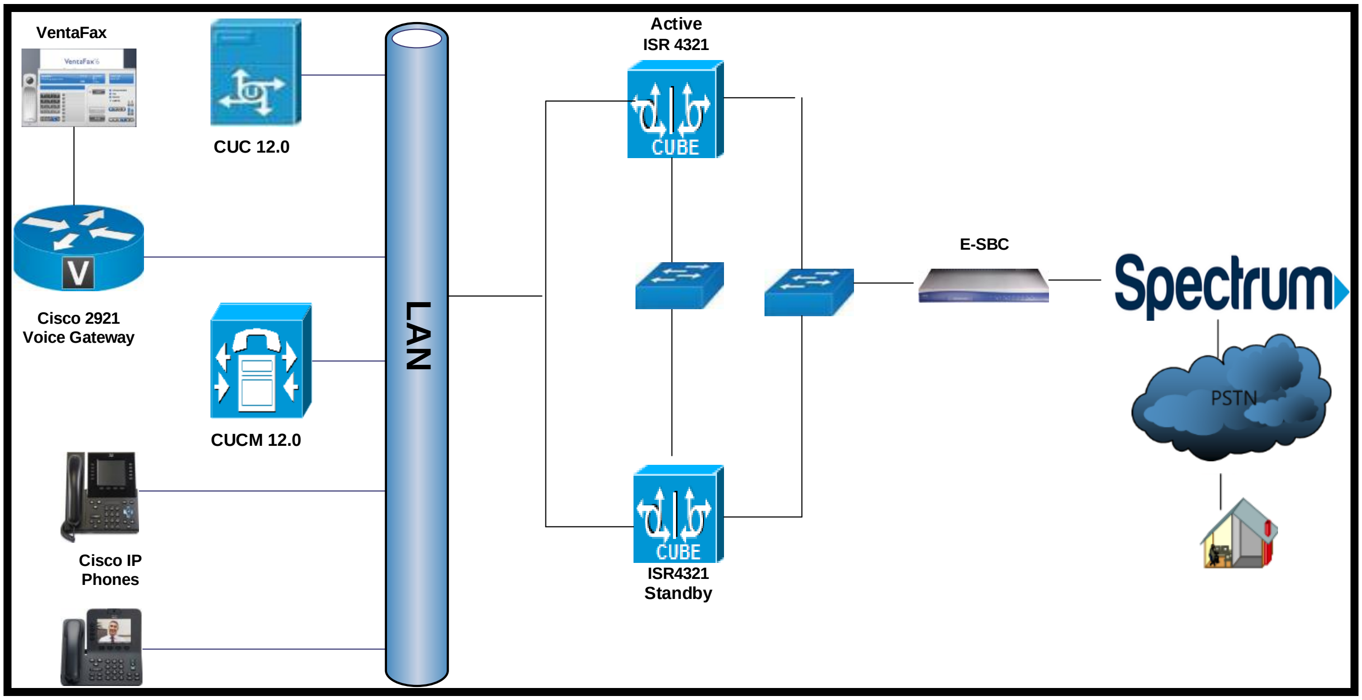

Figure 1: Network Topology

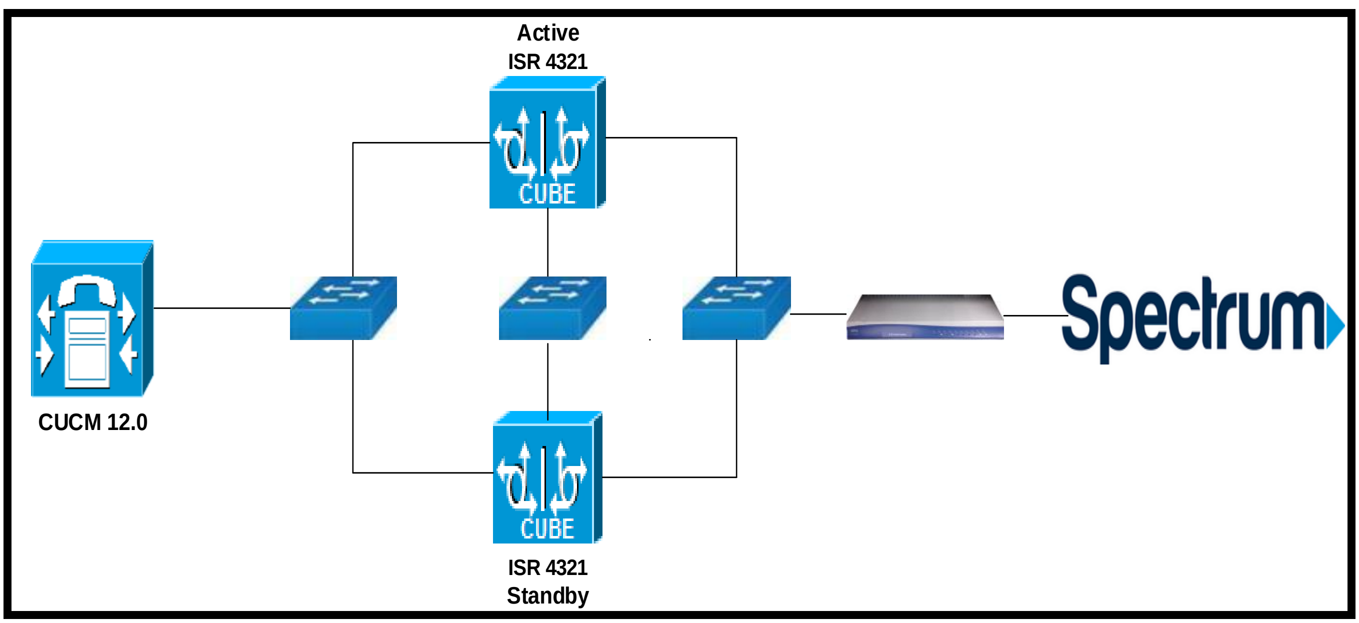

Figure 2: Cisco UBE High Availability

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 2921 Fax Gateway
- IP phones 99X1 (SIP), 8945 (SIP) and 7965 (SCCP)
- Spectrum eSBC – Provided and Managed by Spectrum Enterprise

Software Requirements

- Cisco Unified Communications Manager 12.0.1.10000-10
- Cisco Unity Connection 12.0.1.10000-10
- IOS-XE 16.6.1 for ISR 4321/K9 Cisco Unified Border Element
- Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.6.1, RELEASE SOFTWARE (fc4)
- Cisco IOS XE Software, Version 16.06.01
- IOS 15.1(4)M5 for Cisco 2921 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)
- T38 fax. Note T38 Fax was not available for testing with the Spectrum Enterprise L-Charter SIP trunk variant.

Features Not Supported

- Cisco IP phones used in this test do not support Blind Transfer
- G729 is not supported by Service Provider
- 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.

- T38 fax loopback call via Spectrum Enterprise (L-TWC variant) does not work,

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 10.64.5.8 255.255.0.0
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 10.64.5.10 exclusive
!
interface GigabitEthernet0/0/1
 ip address 10.80.10.48 255.255.255.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.10.50 exclusive
```

Global CUBE Settings

In order to enable CUBE 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
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  rel1xx supported "rel100"
  session refresh
  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 g711alaw
```

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 bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 20 voip
 description Outgoing to Spectrum SIP trunk
 destination-pattern [0-9]T
 session protocol sipv2
 session target ipv4:10.64.5.1 ---(Note: 10.64.5.11 for L-Charter variant)
 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 sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
```

```
!
dial-peer voice 30 voip
 description Outgoing from Spectrum SIP trunk
 huntstop
 session protocol sipv2
 incoming called-number 469573....
 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 sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 40 voip
 description Outgoing to CUCM from Spectrum SIP trunk
 huntstop
 destination-pattern 469573....
 session protocol sipv2
 session target ipv4:10.80.10.2
 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 sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 50 voip
 description Incoming from Spectrum SIP trunk
 huntstop
 session protocol sipv2
 incoming called-number 303547....
 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 sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 60 voip
 description Outgoing to CUCM from Spectrum SIP trunk
 huntstop
 destination-pattern 303547....
 session protocol sipv2
 session target ipv4:10.80.10.2
 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 sg3-to-g3
 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 “5” 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 “5”. A “5.@” 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 5 prefix followed by the target 1+10Digit DID for that extension number, 5 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 to Spectrum network which will direct back to Cisco UBE and handled same as normal incoming PSTN call. For International calls same pattern 5 followed by 011, country code and calling number is used.

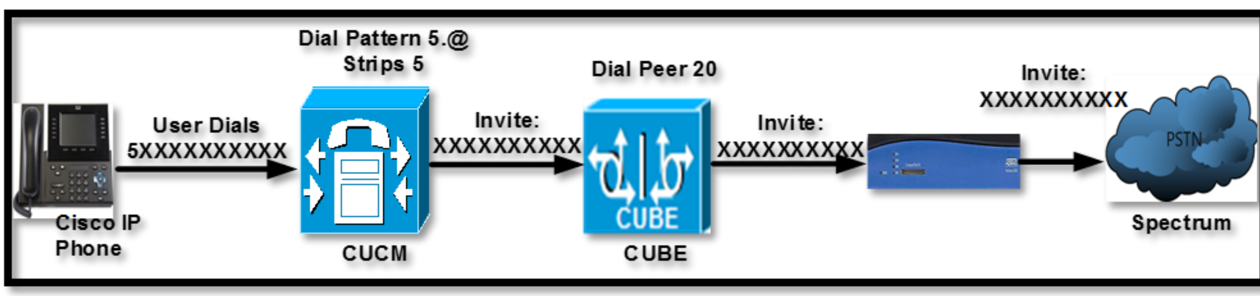

Figure 3: Outbound Voice Call

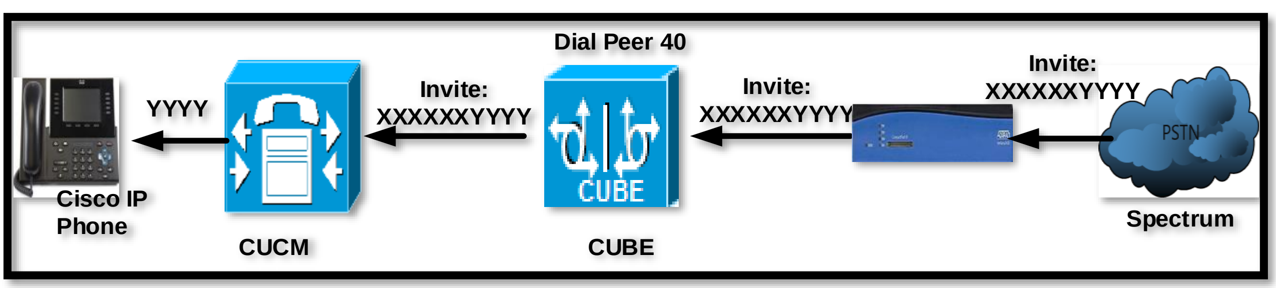

Figure 4: Inbound Voice Call

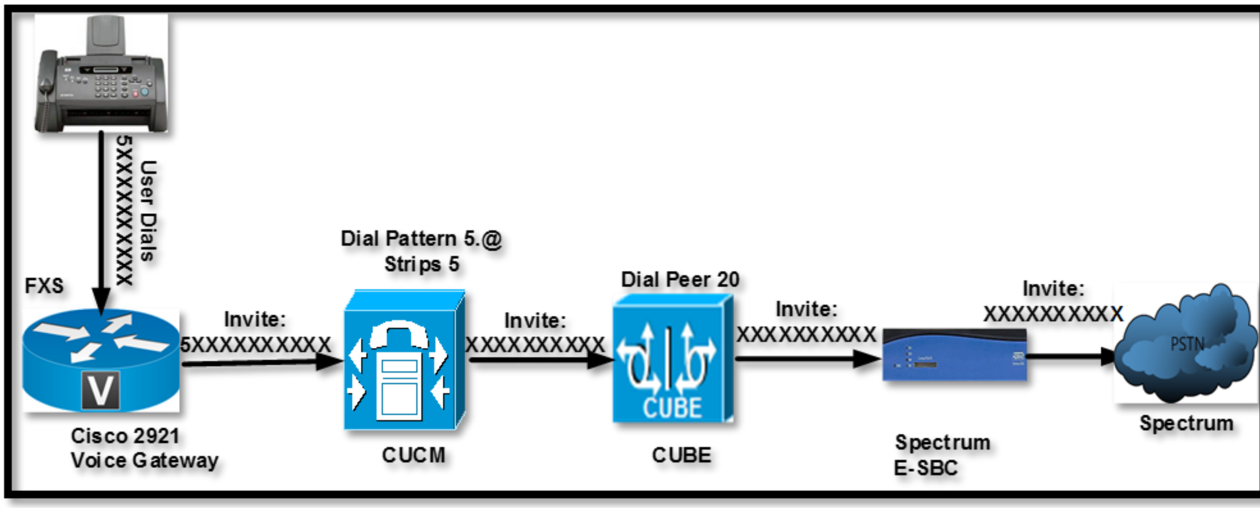

Figure 5: Outbound Fax Call

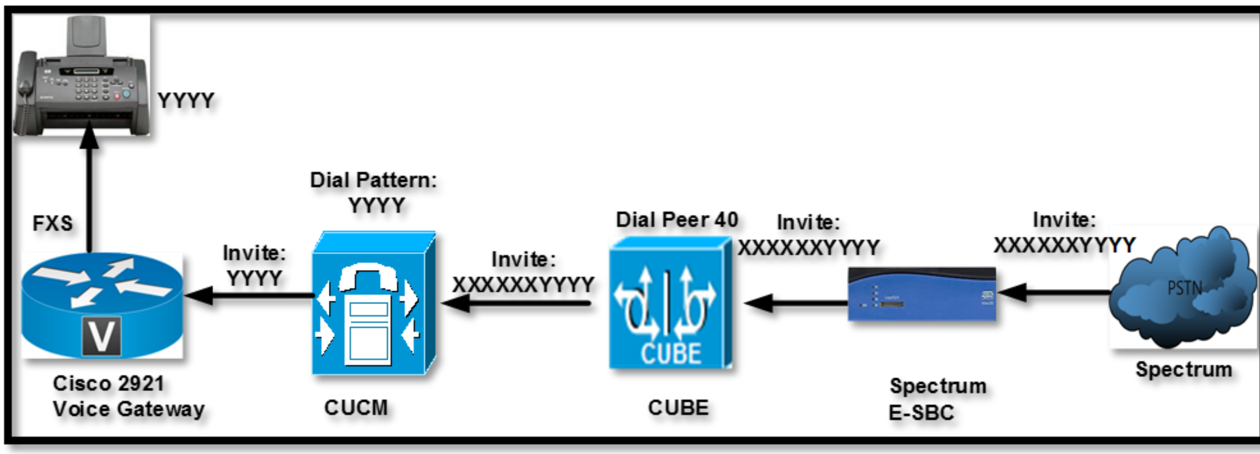

Figure 6: Inbound Fax Call

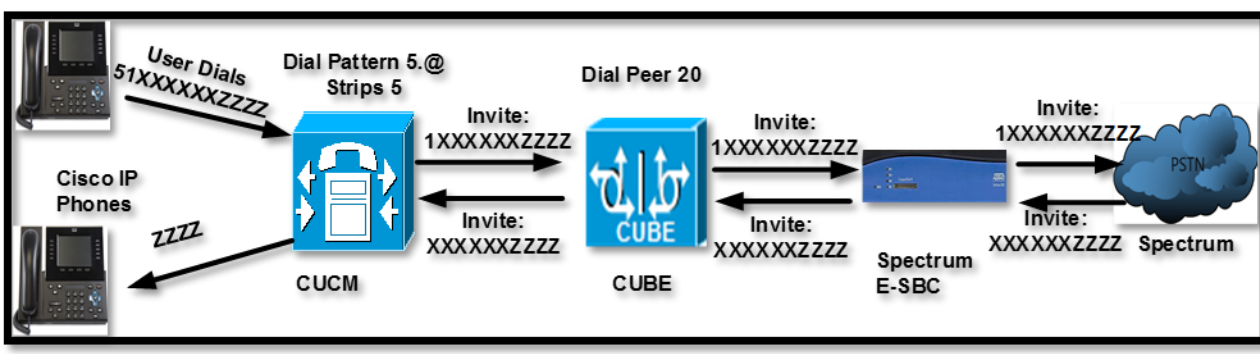

Figure 7: PBX to PBX via Spectrum Call

Configuration Example

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

Active Cisco UBE

Cube3SpecCube1#show run

```
version 16.6
service config
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 Cube3SpecCube1
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
enable secret 5 $1$GDSM$JeD6RWIHx4hcko/eAmvcEXc.
```

```
!
no aaa new-model
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-1582728230
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-1582728230
 revocation-check none
 rsakeypair TP-self-signed-1582728230
!
crypto pki certificate chain TP-self-signed-1582728230
!
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
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  rel1xx supported "rel100"
  session refresh
  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 g711alaw
!
license udi pid ISR4321/K9 sn FDO19220XSQ
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level securityk9
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 15060E0732FB3175783D
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   timers delay 30 reload 60
   control GigabitEthernet0/1/0 protocol 1
   data GigabitEthernet0/1/0
   track 1 shutdown
   track 2 shutdown
!
!
```

```
track 1 interface GigabitEthernet0/0/1 line-protocol
track 2 interface GigabitEthernet0/0/0 line-protocol
!
interface GigabitEthernet0/0/0
 ip address 10.64.5.8 255.255.0.0
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 10.64.5.10 exclusive
!
interface GigabitEthernet0/0/1
 ip address 10.80.10.48 255.255.255.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.10.50 exclusive
!
interface GigabitEthernet0/1/0
 ip address 1.1.1.1 255.255.255.0
 negotiation auto
!
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 GigabitEthernet0/0/1
```

```
ip route 0.0.0.0 0.0.0.0 10.64.5.1
ip route 10.80.19.0 255.255.255.0 10.80.10.1
ip route 172.16.24.0 255.255.248.0 10.80.10.1
!
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 GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax protocol pass-through g711ulaw
```

```
 no vad
!
dial-peer voice 20 voip
 description Outgoing to Spectrum SIP trunk
 destination-pattern [0-9]T
 session protocol sipv2
 session target ipv4:10.64.5.1 (Change to 10.64.5.11 for SIP trunk L-Charter variant)
 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
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dial-peer voice 30 voip
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dial-peer voice 40 voip
```

```
 description Outgoing to CUCM
 huntstop
 destination-pattern 469573....
 session protocol sipv2
 session target ipv4:10.80.10.2
 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 sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
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dial-peer voice 50 voip
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 huntstop
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 incoming called-number 303547....
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 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 sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 60 voip
 description Outgoing to CUCM from Spectrum SIP trunk
 huntstop
```

```
 destination-pattern 303547....
 session protocol sipv2
 session target ipv4:10.80.10.2
 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 sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
!
line con 0
 exec-timeout 0 0
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 login local
 transport input telnet
!
ntp server 34.202.215.187
ntp server pool.ntp.org
!
end
```

Standby Cisco UBE

```
Cube4SpecCube2#show run

version 16.6
service config
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 Cube4SpecCube2
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
enable secret 5 $1$JlS4$FtJ1pipGXrkwyKV3i/1c5.
!
no aaa new-model
```

```
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-1017057749
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-1017057749
 revocation-check none
 rsakeypair TP-self-signed-1017057749
!
crypto pki certificate chain TP-self-signed-1017057749
!
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
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  rel1xx supported "rel100"
  session refresh
  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 g711alaw
!
!
license udi pid ISR4321/K9 sn FDO19220XQ9
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 071B244778X80354471C
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   timers delay 30 reload 60
   control GigabitEthernet0/1/0 protocol 1
   data GigabitEthernet0/1/0
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/1 line-protocol
!
track 2 interface GigabitEthernet0/0/0 line-protocol
```

```
!
interface GigabitEthernet0/0/0
 ip address 10.64.5.9 255.255.0.0
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 10.64.5.10 exclusive
!
interface GigabitEthernet0/0/1
 ip address 10.80.10.49 255.255.255.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.10.50 exclusive
!
interface GigabitEthernet0/1/0
 ip address 1.1.1.2 255.255.255.0
 negotiation auto
!
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 GigabitEthernet0/0/1
ip route 0.0.0.0 0.0.0.0 10.64.5.1
ip route 10.80.19.0 255.255.255.0 10.80.10.1
```

```
ip route 172.16.24.0 255.255.248.0 10.80.10.1
!
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 GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 20 voip
 description Outgoing to Spectrum SIP trunk
```

```
 destination-pattern [0-9]T
 session protocol sipv2
 session target ipv4:10.64.5.1 (Change to 10.64.5.11 for SIP trunk L-Charter variant)
 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 sg3-to-g3
 fax protocol pass-through g711ulaw
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dial-peer voice 30 voip
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 incoming called-number 469573....
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 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
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 fax protocol pass-through g711ulaw
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 description Outgoing to CUCM
 huntstop
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```

```
 session target ipv4:10.80.10.2
 voice-class codec 1
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 fax-relay sg3-to-g3
 fax protocol pass-through g711ulaw
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dial-peer voice 60 voip
 description Outgoing to CUCM from Spectrum SIP trunk
 huntstop
 destination-pattern 303547....
 session protocol sipv2
 session target ipv4:10.80.10.2
 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 sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad!
line con 0
 exec-timeout 0 0
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 login local
 transport input telnet
!
ntp server 34.208.249.133
!
end
```

Configuring Cisco Unified Communications Manager

CUCM Version

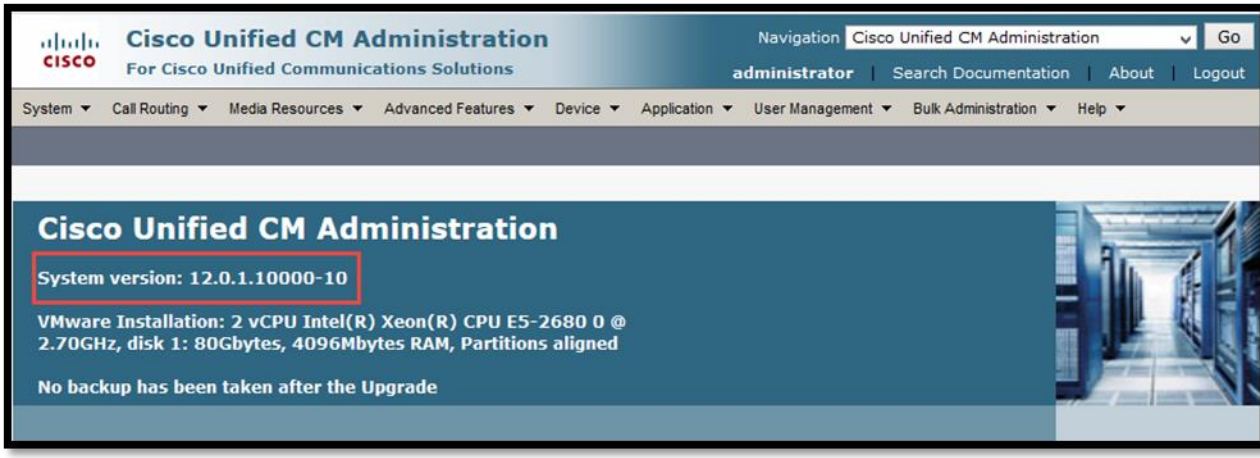


Figure 8: CUCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

1. Select **Server***: clus20pubSub1--CUCM Voice/Video (Active)
2. Select **Service***: Cisco Call Manager (Active)
3. All other fields are set to default values

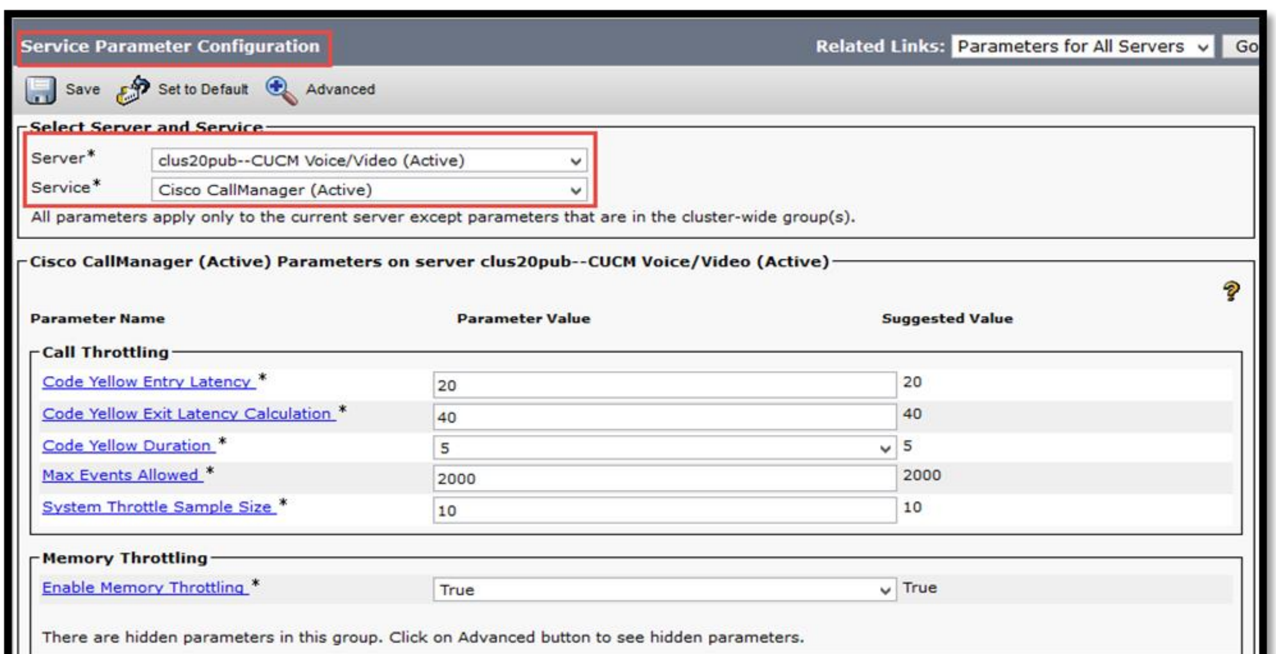


Figure 9: Service Parameters

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 SIP Trunk Security Profile**
2. **Description**: **Non Secure Profile for Spectrum**

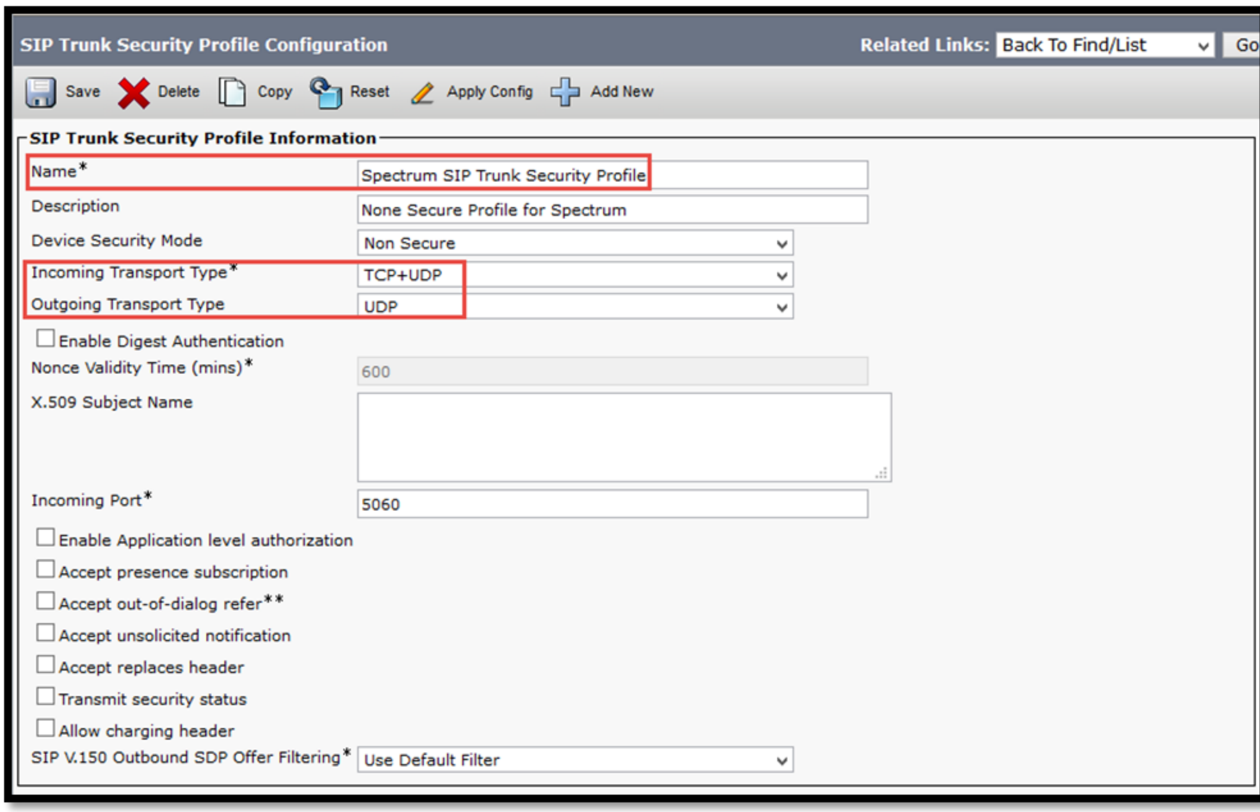


Figure 10: SIP Trunk Security Profile

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 Standard SIP Profile**
2. **Description: Default SIP Profile**

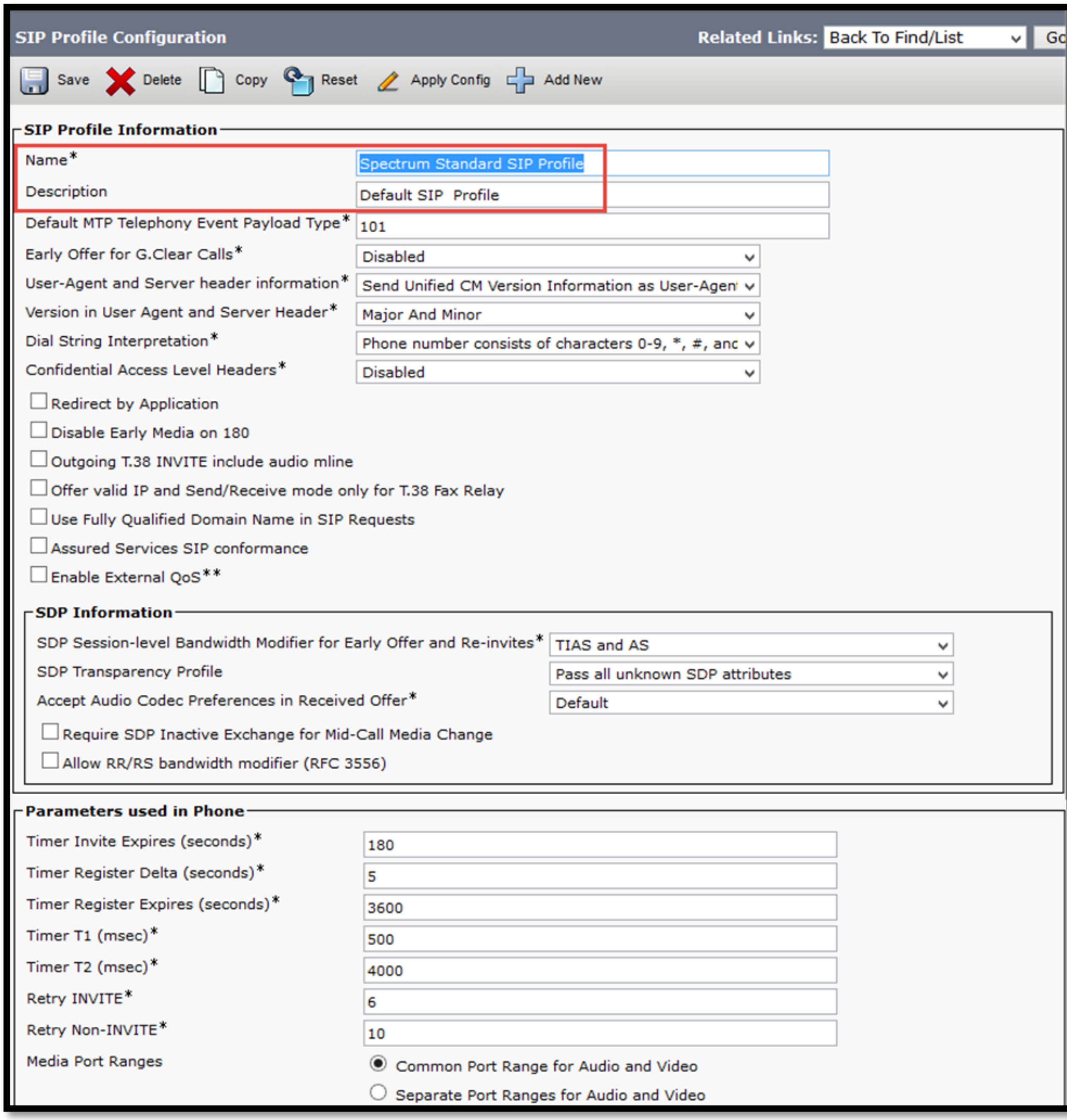
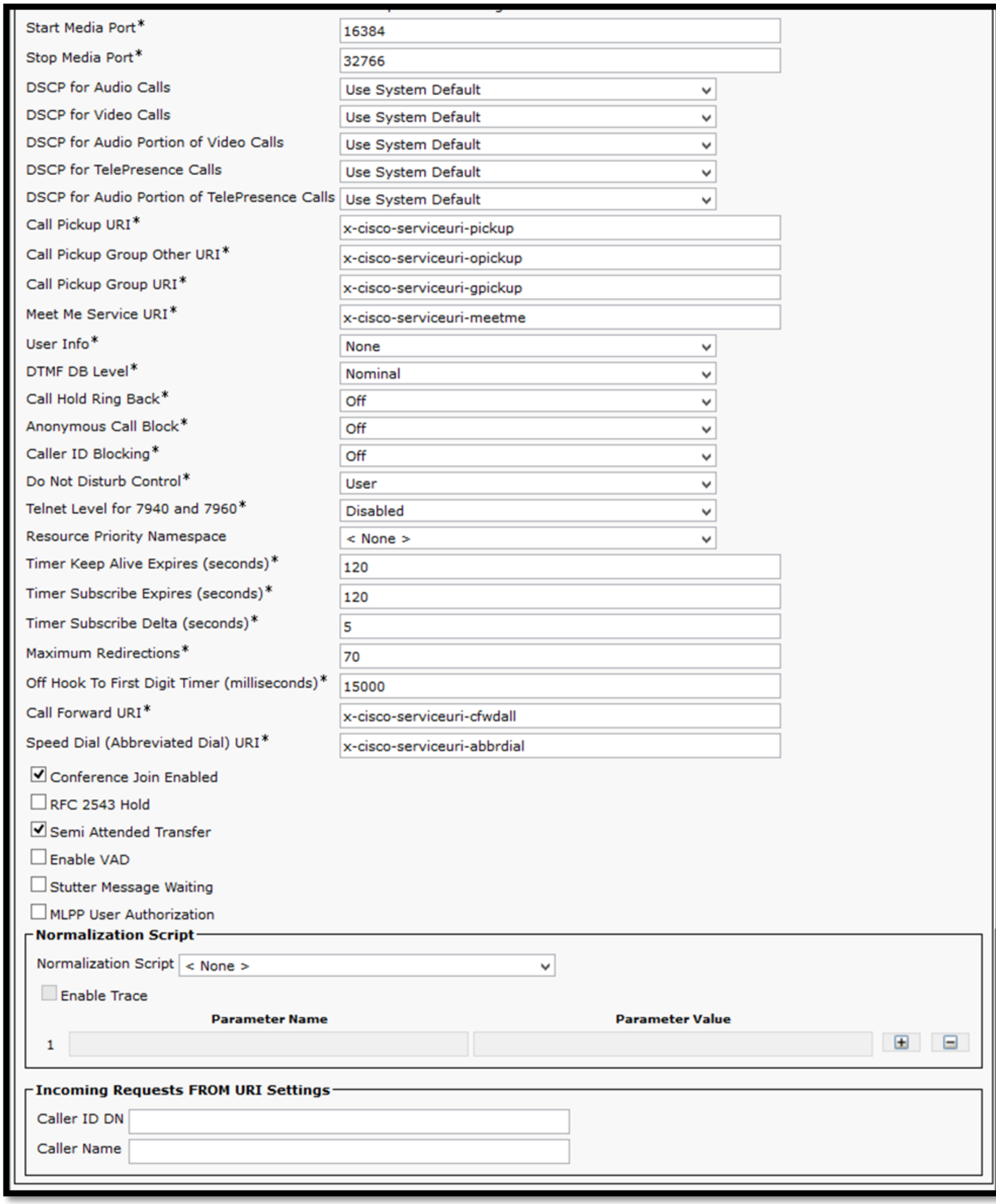


Figure 11: SIP Profile



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

Normalization Script

Normalization Script: < None >

- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

Caller ID DN:

Caller Name:

Figure 12: SIP Profile (Cont.)

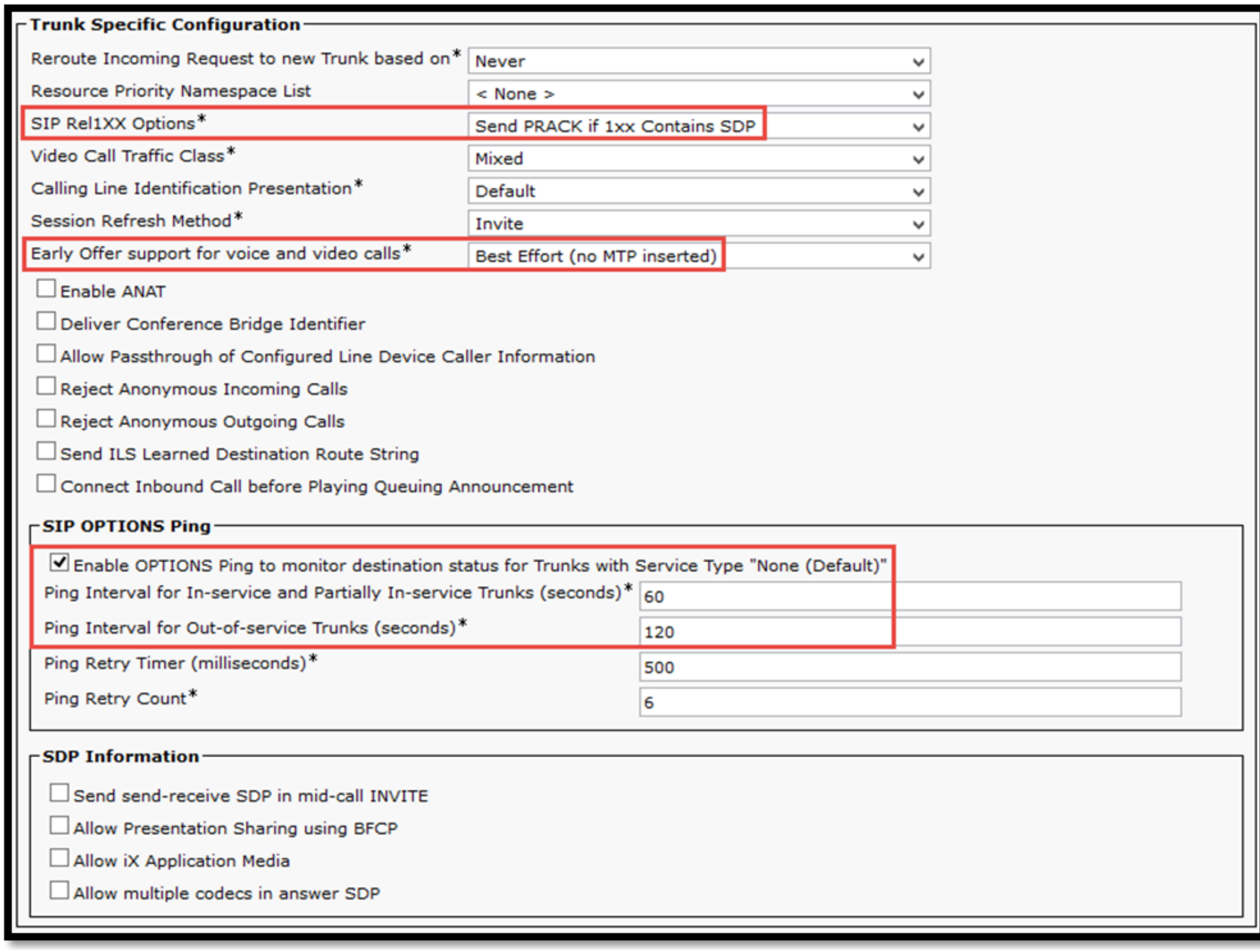


Figure 13: SIP Profile (Cont.)

Explanation

Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
SIP RelXX Options	Send PRACK if 1xx Contains SDP	Enable Provisional Acknowledgements(Reliable 100 messages)
Early Offer support for voice and video calls	Best Effort (no MTP inserted)	Enable early media call
Enable OPTIONS Ping to monitor destination status	Checked	SIP OPTIONS Ping enabled between CUCM and CUBE
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

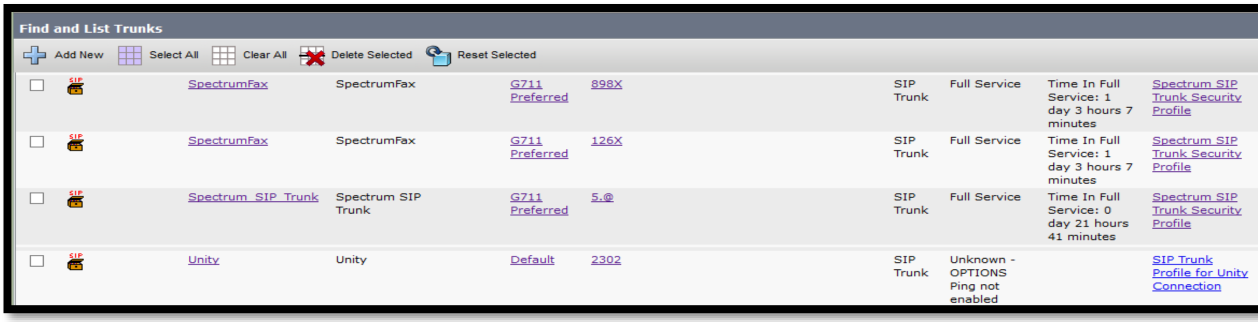


Figure 14: SIP Trunks List

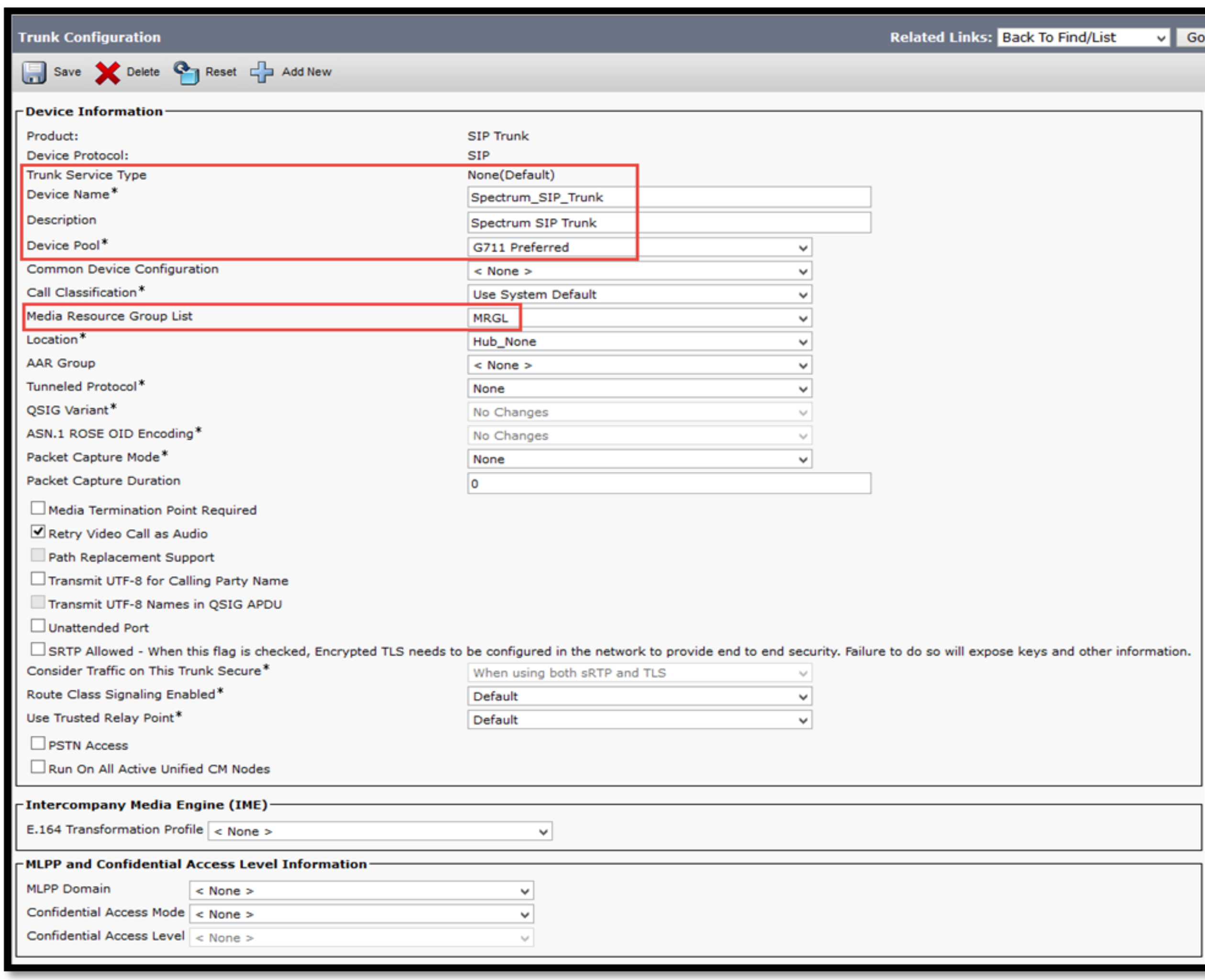
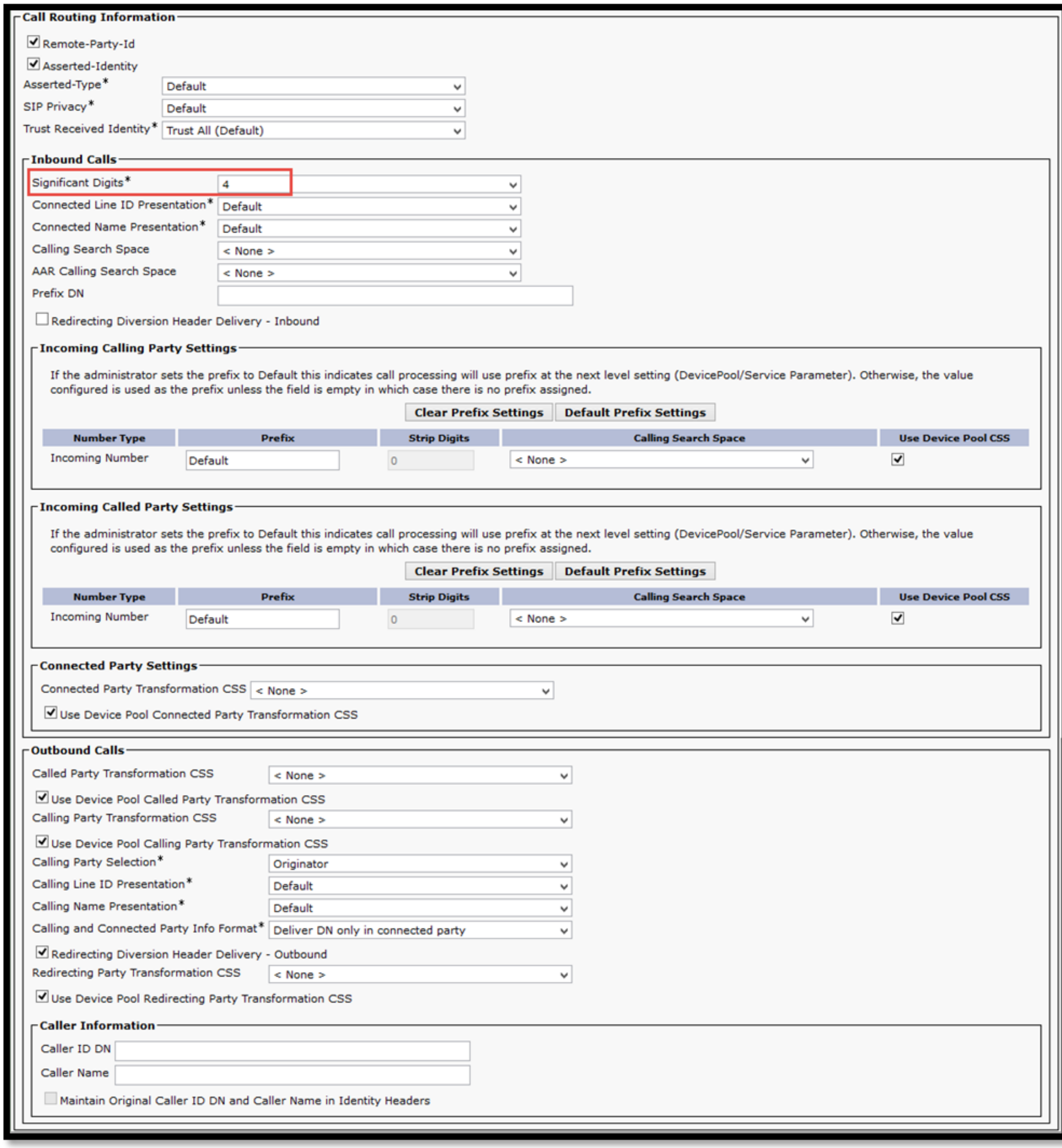


Figure 15: SIP Trunk to CUBE


Call Routing Information

☑ Remote-Party-Id
☑ Asserted-Identity

Asserted-Type*	Default
SIP Privacy*	Default
Trust Received Identity*	Trust All (Default)

Inbound Calls

Significant Digits*	4
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	

☐ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☑

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☑

Connected Party Settings

Connected Party Transformation CSS: < None >

☑ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS: < None >

☑ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS: < None >

☑ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling and Connected Party Info Format*	Deliver DN only in connected party

☑ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS: < None >

☑ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN	
Caller Name	

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

Figure 16: SIP Trunk to CUBE (Cont.)

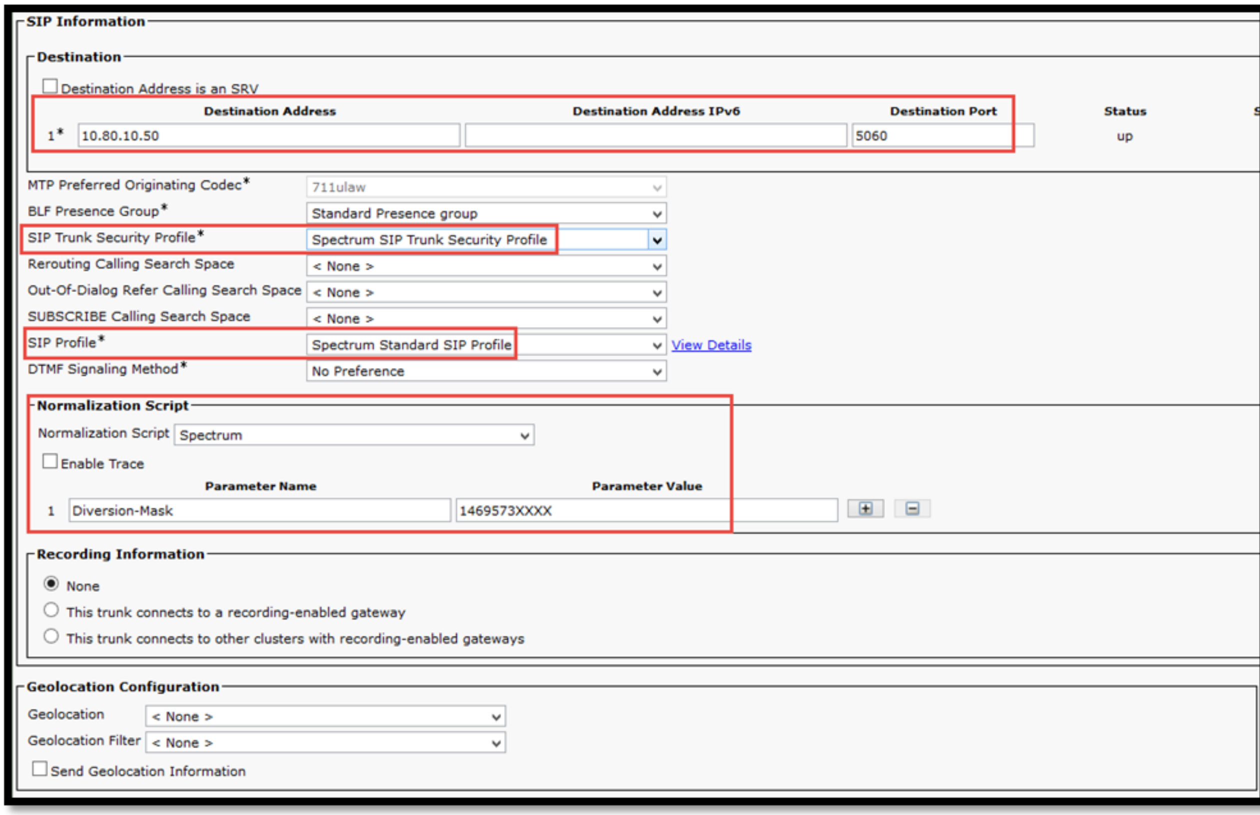


Figure 17: SIP Trunk to CUBE (Cont.)

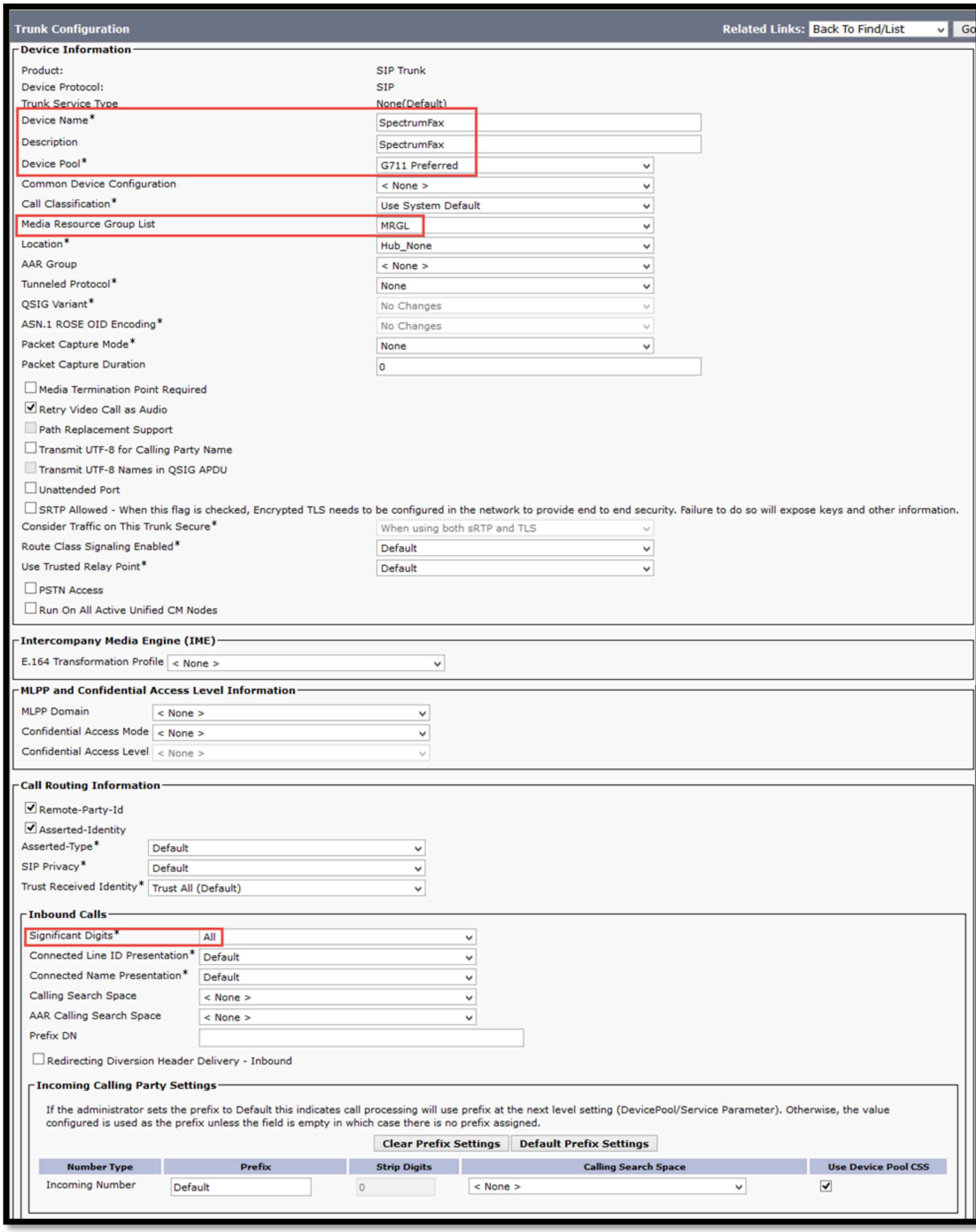


Figure 18: SIP Trunk to Fax Gateway

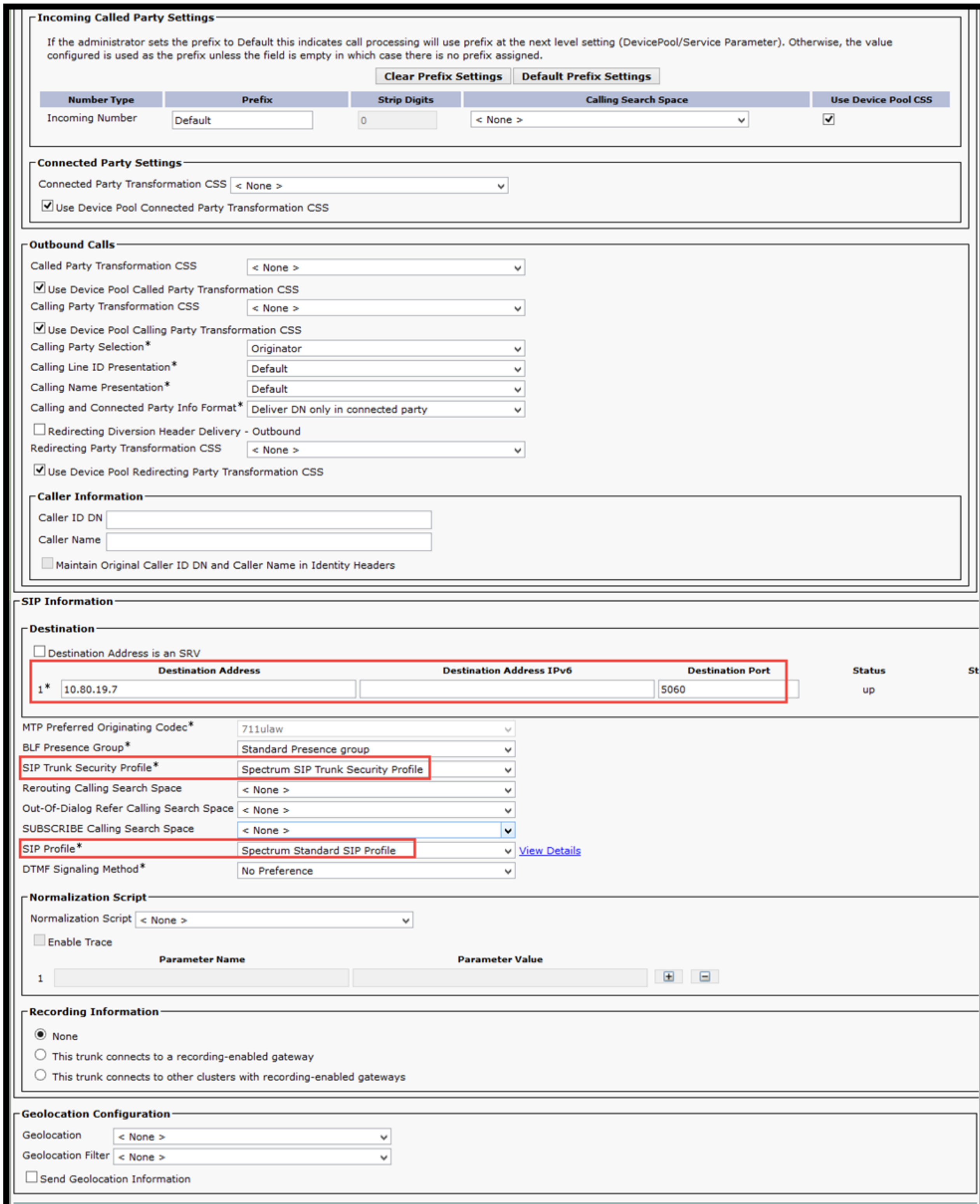


Figure 19: SIP Trunk to Fax gateway Cont.

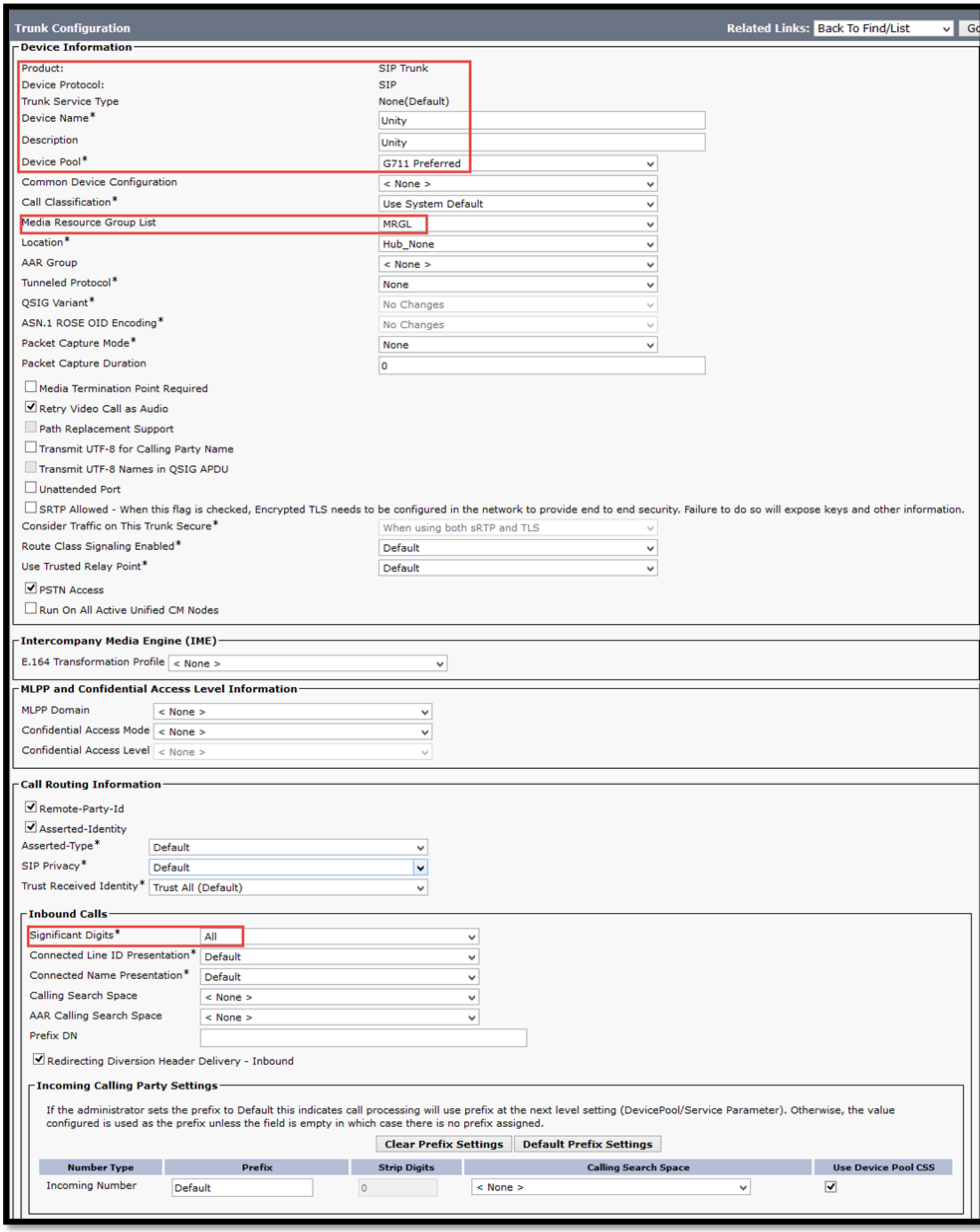


Figure 20: SIP trunk to Cisco Unity Connection

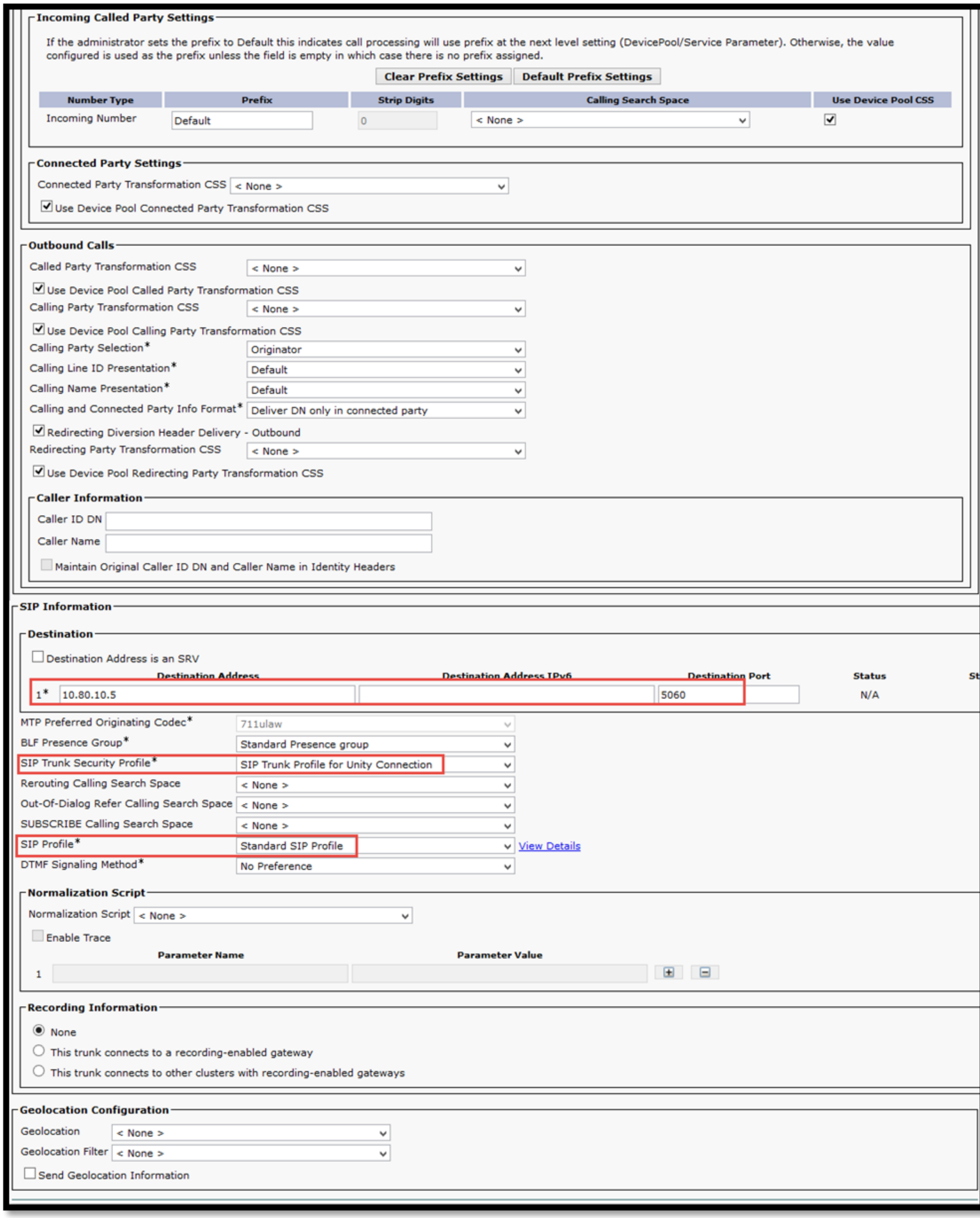


Figure 21: SIP Trunk to Unity Connection Cont.

Explanation

Parameter	Value	Description
Device Name	Spectrum_SIP_Trunk for SIP Trunk to CUBE; SpectrumFax for SIP trunk to Fax Gateway; Unity for SIP trunk to Cisco Unity Connection	Name for the trunk
Device Pool	G711 Preferred	G711 Device Pool is used for this trunk
Media Resource Group List	MRGL	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4 for SIP Trunk to CUBE; All for SIP trunk to Fax gateway and Cisco Unity Connection	4 digits Extension for all CPE phones/Fax/Unity Connection Pilot number
Destination Address	10.80.10.50/10.80.19.7/10.80.10.5 for SIP trunk to CUBE/Fax Gateway/Cisco Unity Connection	IP address of the Cisco UBE Virtual LAN/Fax Gateway/Cisco Unity Connection
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
Normalization Script	Spectrum for SIP trunk to CUBE only	Convert 4 digits EXT to 11 digits DID for Diversion header
Diversion-Mask	1469573XXXX/1303547XXXX	Used in the Manipulation Script for Timer Warner Cable/L-Charter

Normalization Script

A SIP Normalization Script is used to convert SIP Diversion Headers from 4-digit EXT to the full 11-digit telephone number, this is required for call redirecting over Spectrum SIP network.
Navigate to **Device**>**Device Settings**>**SIP Normalization Script**

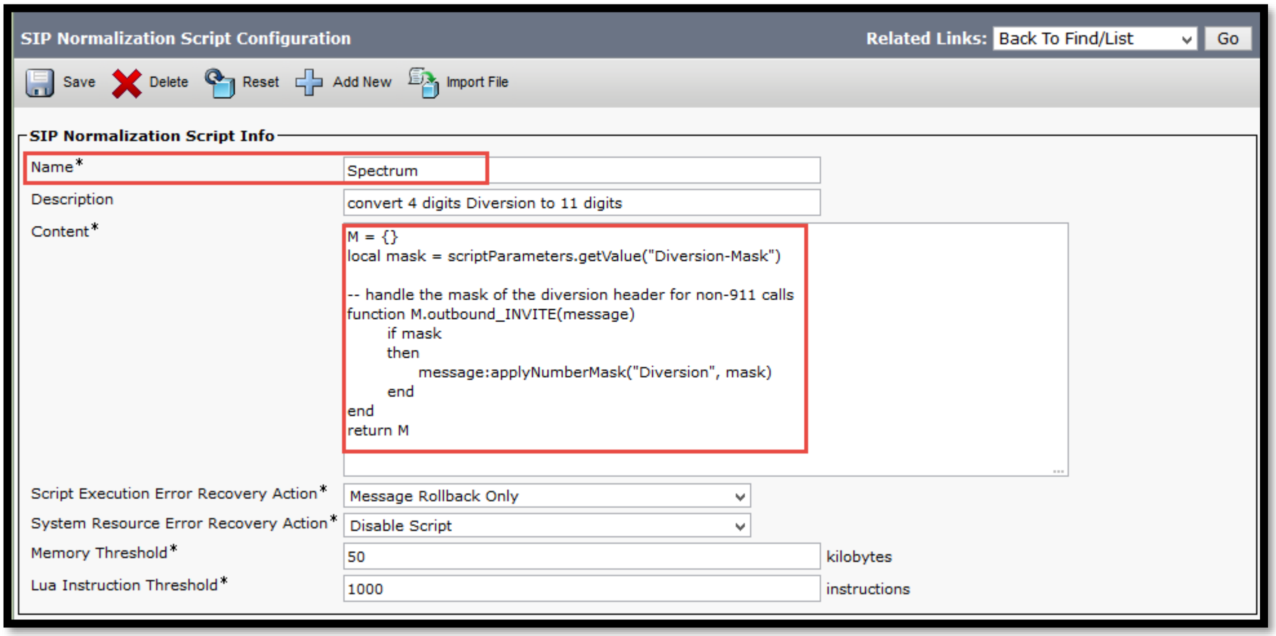


Figure 22: SIP Normalization Script

SIP Normalization Script (Text):

```
M = {}
local mask = scriptParameters.getValue("Diversion-Mask")

-- handle the mask of the diversion header for non-911 calls
function M.outbound_INVITE(message)
        if mask
        then
            message:applyNumberMask("Diversion", mask)
        end
end
return M
```

Dial Plan

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial "5"+1+10 digits number to access PSTN via Cisco UBE
  - "5" is removed before sending to Cisco UBE
- For FAX call, Access Code "5"+ 1+10 digits number is used at Cisco Fax gateway
  - "5" is removed at Cisco UCM
  - The rest of the number is sent to Cisco UBE to Spectrum Enterprise network
- Incoming fax call to 898X (for L-TWC) or 126X (for L-Charter) will be sent to Cisco Fax gateway
- Call to Voice Mail 2302 will be send to Cisco Unity Connection

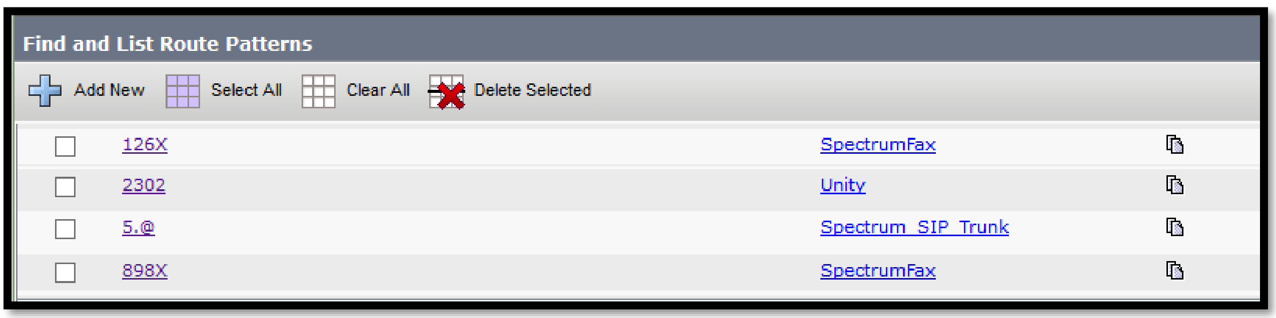


Figure 23: Route Patterns List

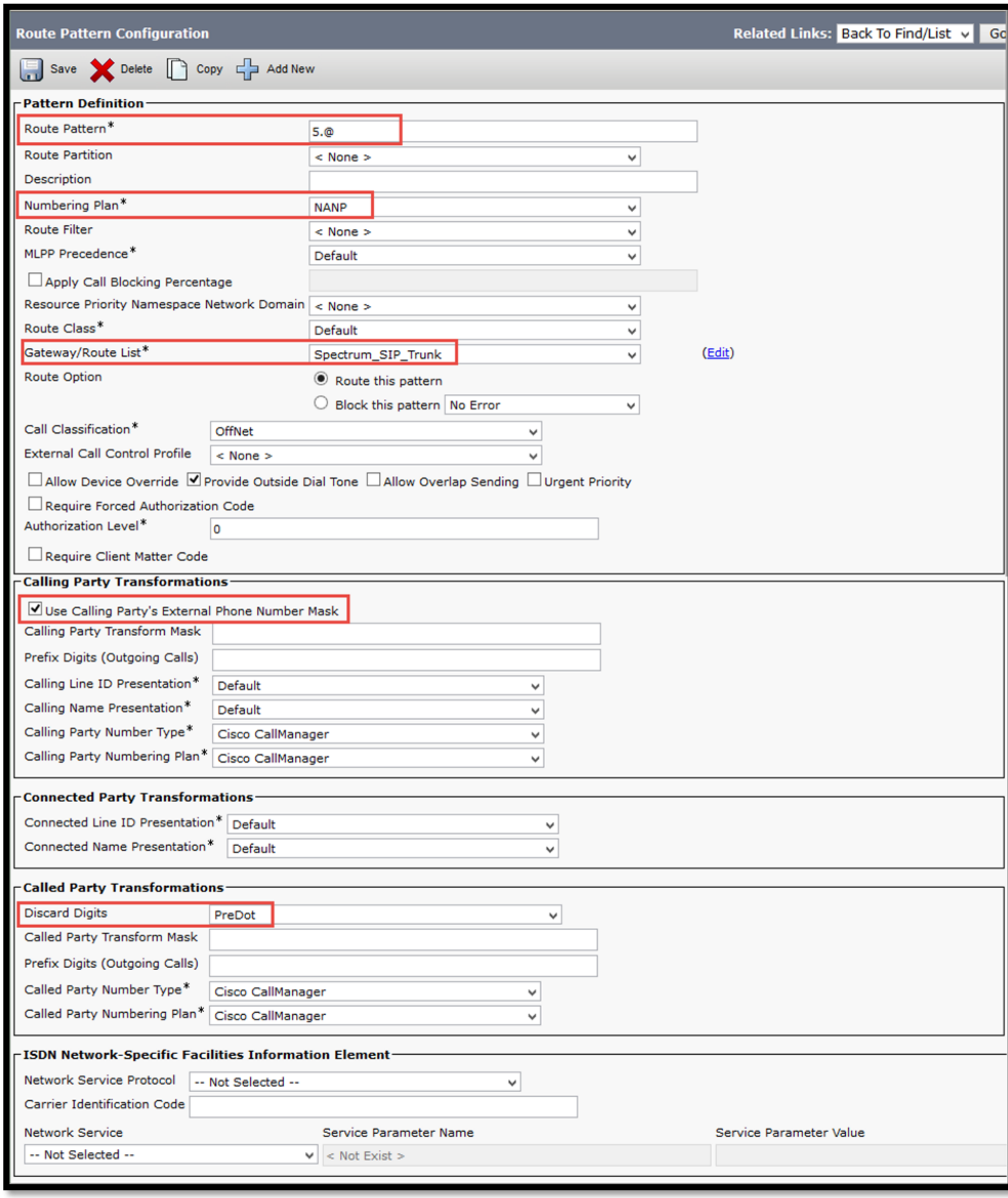


Figure 24: Route Pattern for Voice

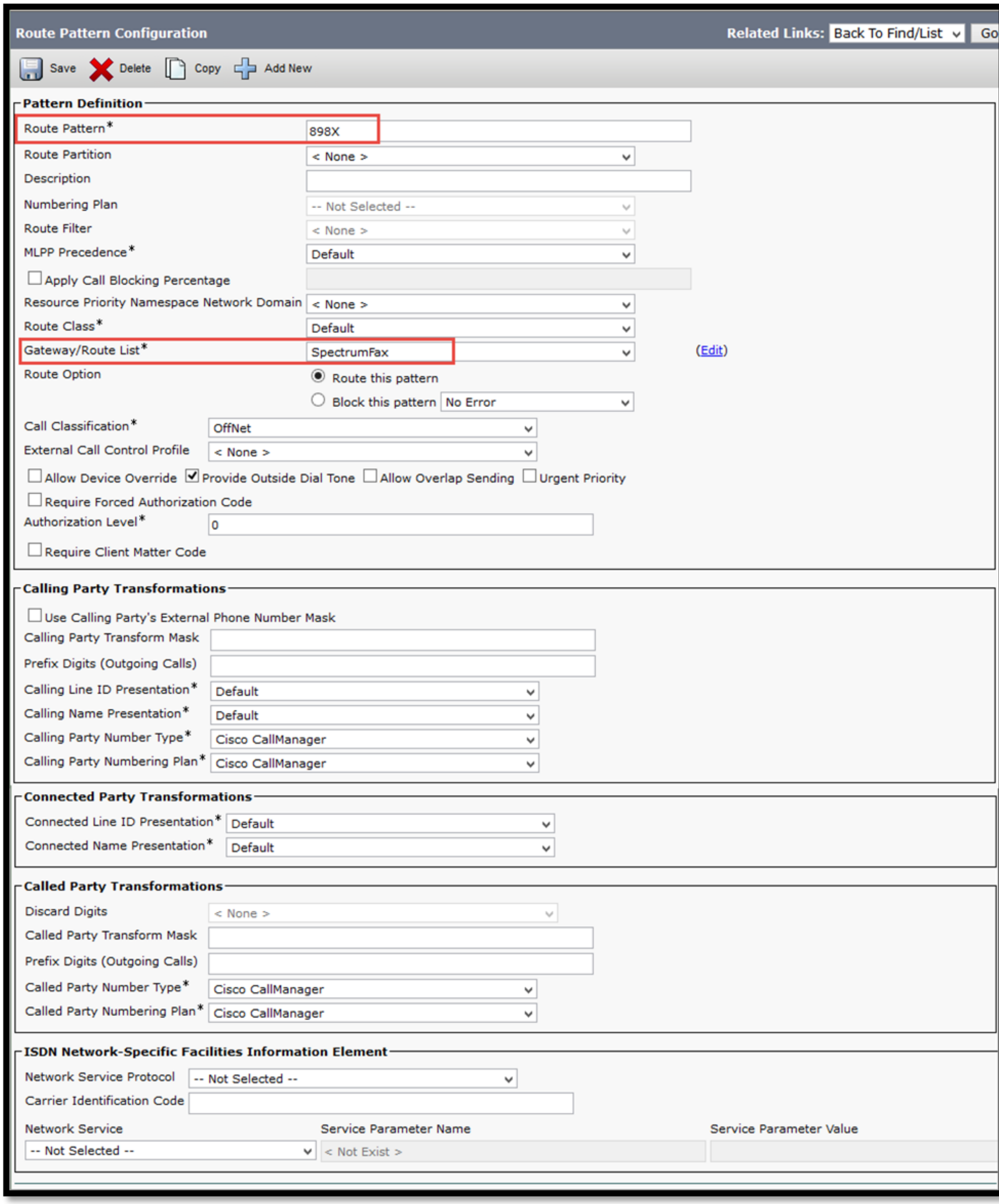


Figure 25: Route Pattern for Fax (L-TWC)

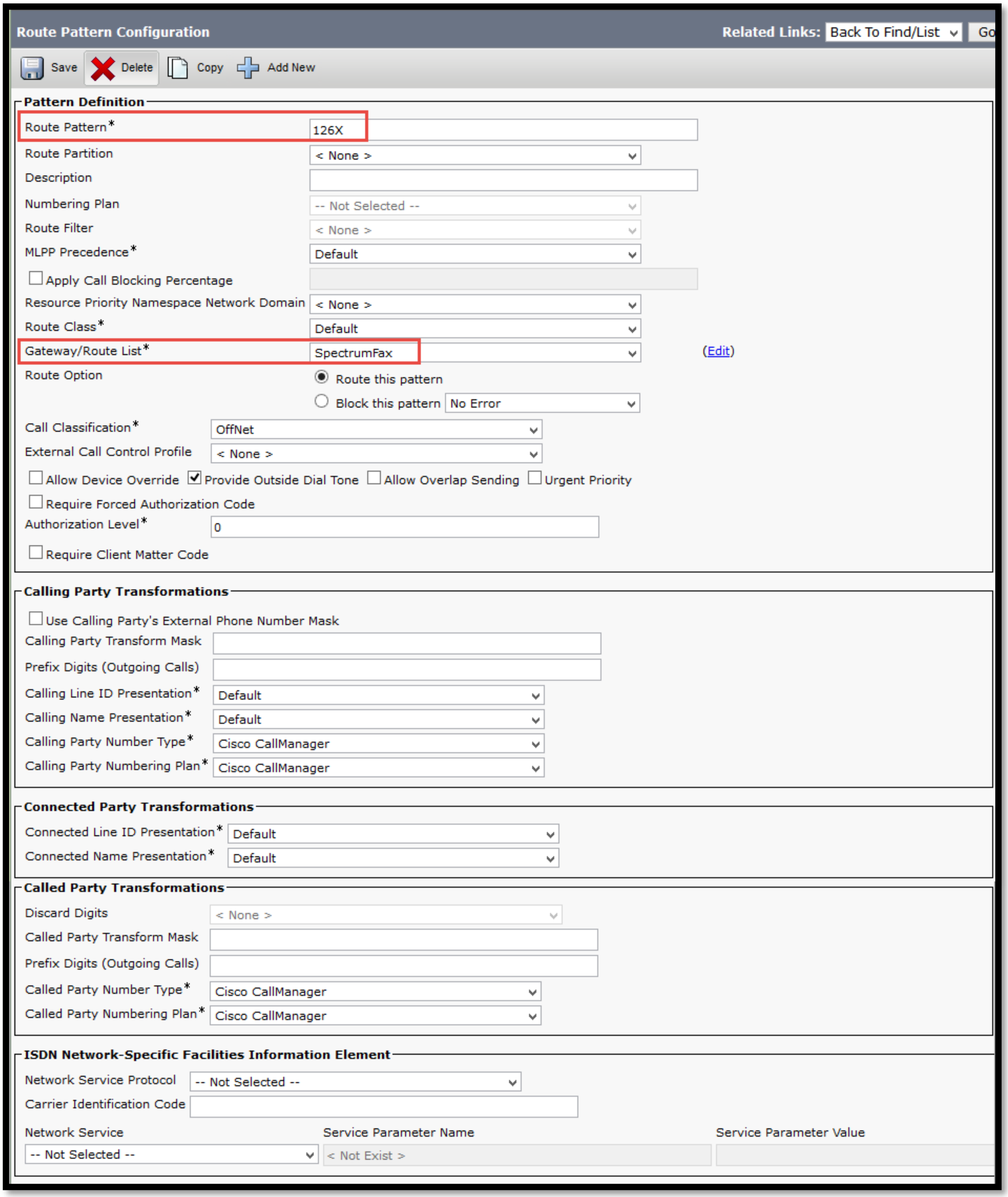


Figure 26: Route Pattern for Fax (L-Charter)

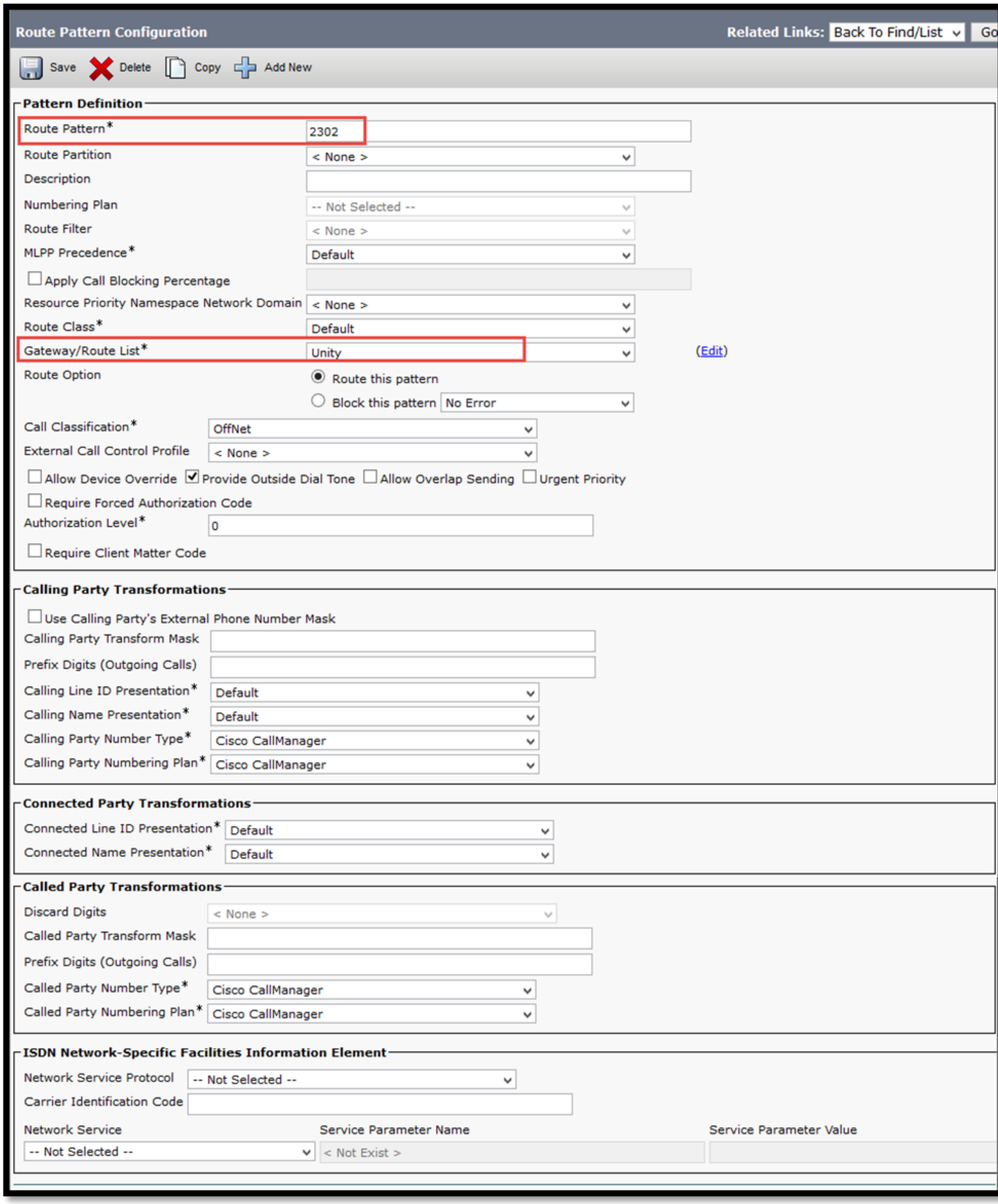


Figure 27: Route Pattern to Cisco Unity Connection

Explanation

Setting	Value	Description
Route Pattern	5.@ for Voice & International Calls, 898X for Fax Call and 2302 for Unity Connection	Specify appropriate Route Pattern
Gateway/Route List	Spectrum _SIP_Trunk for Route Pattern 5.@, SpectrumFax for Pattern 898X and 126X; and Cisco Unity Connection for Pattern 2302	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 5.@	North American Numbering Plan
Discard Digits	PreDot for Route Pattern 5.@	Specifies how to modify digit before they are sent to Spectrum network

Fax Gateway Example Configuration

FAX-GATEWAY#show run

version 15.4

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

service sequence-numbers

!

hostname FAX-GATEWAY

!

boot-start-marker

boot-end-marker

!

aqm-register-fnf

!

logging queue-limit 1000000000

logging buffered 30000000

logging rate-limit 10000

no logging console

```
enable secret 4 iR3uUX3Bo6oYbT6ajhFwJx39FR4g.1QCmm7yYduKGZI
!
no aaa new-model
clock timezone CST -6 0
clock summer-time CDT recurring
!
ip domain name lab.tekvizion.com
ip name-server 10.64.1.3
ip cef
no ipv6 cef
multilink bundle-name authenticated
!
stcapp register capability 0/0/0 both
stcapp ccm-group 15
!
stcapp feature access-code
!
stcapp feature speed-dial
!
stcapp supplementary-services
 port 0/0/0
  fallback-dn 4037
!
cts logging verbose
!
crypto pki trustpoint TP-self-signed-1120430079
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-1120430079
 revocation-check none
```

```
 rsakeypair TP-self-signed-1120430079
!
!
crypto pki certificate chain TP-self-signed-1120430079
 certificate self-signed 01
  3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 31313230 34333030 3739301E 170D3133 31303031 32313434
  33305A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 31323034
  33303037 3930819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
  8100D201 333D5645 360F1451 B7BC6EBB 7FFC6715 F1658731 5ABE4743 6B6EA3B0
  1C7D1D76 17B5BD08 BA68F94B 80F790EA 8C46036E 7FDE5827 3BF9B45F 7B71D959
  B6943C33 A4781184 7DA425F8 81EC15A8 1AFC1E03 3690A9E4 DF1C9AF1 974151B1
  15742950 E0ABEBBB DDE57C06 97D3C3A4 4205ADE2 36CB165E 47967DC4 8F41F25B
  17430203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
  551D2304 18301680 14107F60 ACB1F013 843D556C F6C614CF 491E69DA E5301D06
  03551D0E 04160414 107F60AC B1F01384 3D556CF6 C614CF49 1E69DAE5 300D0609
  2A864886 F70D0101 05050003 81810029 2333B82C 9C91341D 6921E674 E365F73A
  840DB5A3 54C20F8B E004BE5E 741B5B53 3978D629 DBE53B11 51C2F563 9A82DCC4
  37D269C6 B8A5F80D 969C90E1 DA963C31 D49B3F9C B7A1E484 C792A0E7 1BEA5D90
  0A501640 BD4045A8 649E269C B4891C0D FF317730 DD446398 2E0130DB 6DCD39EA
  8C68EC80 308E03E1 80C6C503 CD45A4
        quit
voice-card 0
!
voice service voip
 no ip address trusted authenticate
 allow-connections sip to sip
```

```
 no supplementary-service sip handle-replaces
 redirect ip2ip
 fax protocol pass-through g711ulaw
 sip
  rel1xx disable
  header-passing
  error-passthru
  midcall-signaling passthru
!
license udi pid CISCO2901/K9 sn FTX174081TS
hw-module pvdm 0/0
!
username cisco privilege 15 secret 5 $1$vLg0$o8vBEuXW0zO4WrX6xkgOY/
!
redundancy
!
translation-rule 898
 Rule 1 8985 4695738985
 Rule 2 8986 4695738986
!
translation-rule 126
 Rule 1 1267 3035471267
 Rule 2 1268 3035471268
!
interface Embedded-Service-Engine0/0
 no ip address
 shutdown
!
interface GigabitEthernet0/0
```

```
 description FAX Gateway for IP Toll Free
 ip address 10.80.19.7 255.255.255.0
 duplex auto
 speed auto
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 10.80.19.1
!
!
control-plane
!
!
voice-port 0/0/0
 no echo-cancel enable
 no vad
 cptone IN
```

```
 caller-id enable
!
voice-port 0/0/1
 no echo-cancel enable
 no vad
 cptone IN
 caller-id enable
 !
 !
no mgcp timer receive-rtcp
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
!
no ccm-manager fax protocol cisco
!
!
dial-peer voice 203 voip
 description Gateway to CUCM
 service session
 destination-pattern 5T
 translate-outgoing calling 898 (898 for L-TWC and 126 for L-Charter)
 session protocol sipv2
 session target ipv4:10.80.10.2
 session transport udp
 codec g711ulaw
```

```
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 204 voip
 description CUCM to Gateway
 service session
 session protocol sipv2
 session transport udp
 incoming called-number 898. (898. for L-TWC and 126. for L-Charter)
 codec g711ulaw
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 8985 pots
 service session
 destination-pattern 8985
 no digit-strip
 port 0/0/0
 forward-digits 0
!
dial-peer voice 1268 pots
 service session
 destination-pattern 1268
 no digit-strip
 port 0/0/1
 forward-digits 0
!
```

```
!
gateway
 timer receive-rtp 1200
!
sip-ua
 no timers hold
!
!
!
gatekeeper
 shutdown
!
!
telephony-service
 max-conferences 8 gain -6
 transfer-system full-consult
!
!
line con 0
 exec-timeout 0 0
 login local
line aux 0
line vty 0 4
 session-timeout 180
 exec-timeout 0 0
 login local
 transport input telnet ssh
line vty 5 15
 privilege level 15
```

login local

transport input telnet ssh

!

scheduler allocate 20000 1000

!

End

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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