Application Note

Charter SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR4321/K9 [IOS-XE 3.17.1 – 15.6(1)S1] using SIP

July 25, 2016

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Introduction

Service Providers today, such as Charter, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and centralized IP to TDM POP gateways to provide on-net and off-net services.

Charter is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and Charter network, Cisco Unified Border Element (Cisco UBE 11.5.0) ISR 4321/K9 running IOS-XE 3.17.1 – 15.6(1) S1 can be used. The Cisco Unified Border Element provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to Charter 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 Charter 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) 11.0.1 and Cisco Unified Border Element (Cisco UBE 11.5.0) on ISR 4321/K9 [IOS-XE 3.17.1 - 15.6(1)S1] for connectivity to Charter SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (Charter).
- Testing was performed in accordance to Charter 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 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 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

Network Topology

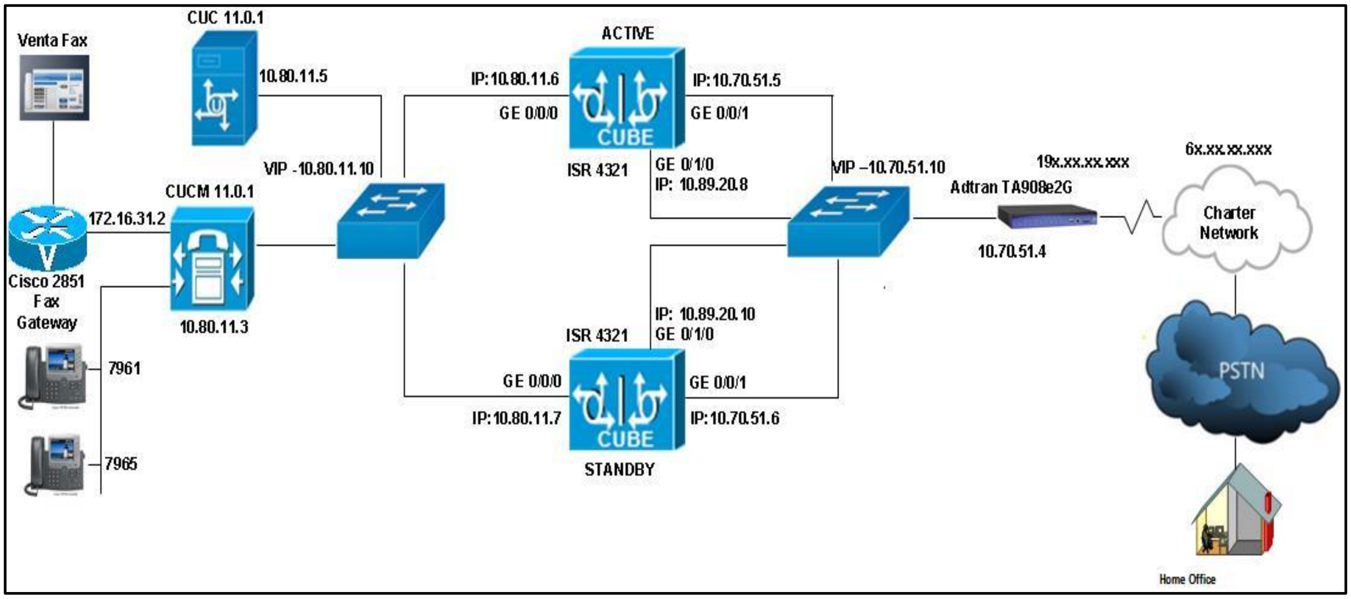

Figure 1: Network Topology

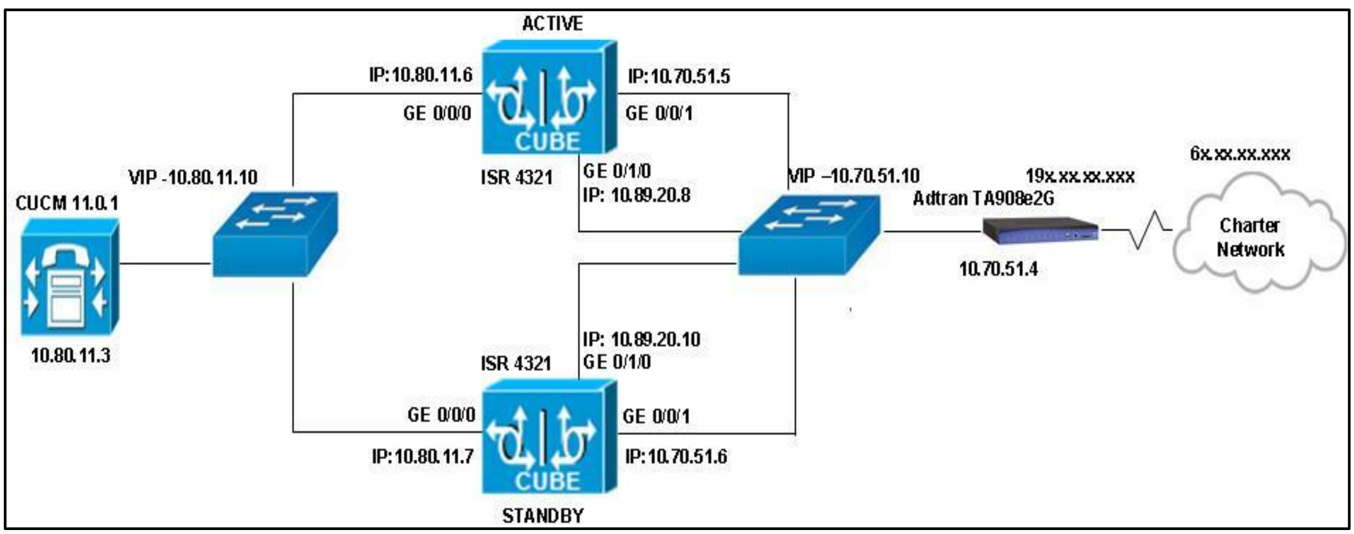

Figure 2: Cisco UBE High Availability

System Components

Hardware Requirements

- Cisco UCSC-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as CUBE
- Cisco ISR4321/K9 (1RU) processor with 1647061K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
- Processor board ID FLM1925W0X2
- Cisco 2851 Fax Gateway
- IP phones 7961 (SIP) and 7965 (SCCP)
- Adtran Total_Access_908e_2nd_Gen) – Provided and managed by Charter

Software Requirements

- Cisco Unified Communications Manager 11.0.1
- Cisco Unity Connection 11.0.1
- IOS-XE 3.17.1 - 15.6(1)S1 for ISR 4321/K9 Cisco Unified Border Element 11.5.0
- Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.6(1)S1, RELEASE SOFTWARE (fc3)
- Cisco IOS XE Software, Version 03.17.01.S
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway
- Adtran Total_Access_908e_2nd_Gen /R11.4.6.E - Provided and managed by Charter

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 Service Provider
- 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 Charter

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
 description Charter LAN MS4 1/0/7
 ip address 10.80.11.6 255.255.255.0
 media-type rj45
 negotiation auto
 no mop enabled
 redundancy rii 1
 redundancy group 1 ip 10.80.11.10 exclusive
!
interface GigabitEthernet0/0/1
 description Charter WAN MS4 1/0/8
 ip address 10.70.51.5 255.255.255.0
 negotiation auto
 no mop enabled
 redundancy rii 2
 redundancy group 1 ip 10.70.51.10 exclusive
!
```

Global Cisco UBE Settings

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

```
voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol pass-through g711ulaw
sip
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
session refresh
asserted-id pai
privacy pstn
early-offer forced
no silent-discard untrusted
midcall-signaling 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

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

```
dial-peer voice 200 voip
description Outbound-from IP PBX to PSTN - WAN facing
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip options-keepalive
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
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 210 voip
description outgoing call to Charter - LAN facing
```

```
huntstop
session protocol sipv2
session transport udp
incoming called-number .T
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
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 nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 500 voip
description cube-dp incoming call from PSTN
huntstop
session protocol sipv2
session transport udp
incoming called-number 303835....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip options-keepalive
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
```

```
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 100 voip
description Inbound-from PSTN to IP PBX - LAN facing
huntstop
destination-pattern 303835....
session protocol sipv2
session target ipv4:10.80.11.3:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
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 nsf 000000
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 "8" 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 "8". A "8.@" 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 Charter, Caller dial 8 prefix followed by the target 1+10Digit DID no for that extension number, 8 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 Charter network which will direct back to Cisco UBE and handled same as normal incoming PSTN call. For International calls same pattern 8.@ followed by 011, country code and calling no is used.

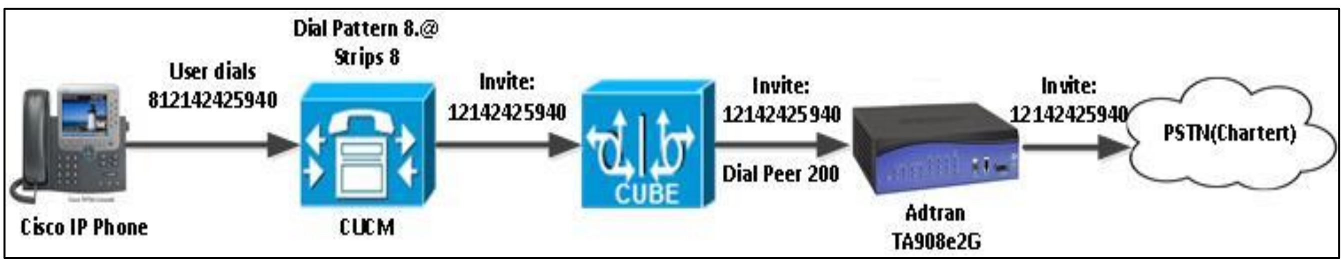

Figure 3: Outbound Voice Call

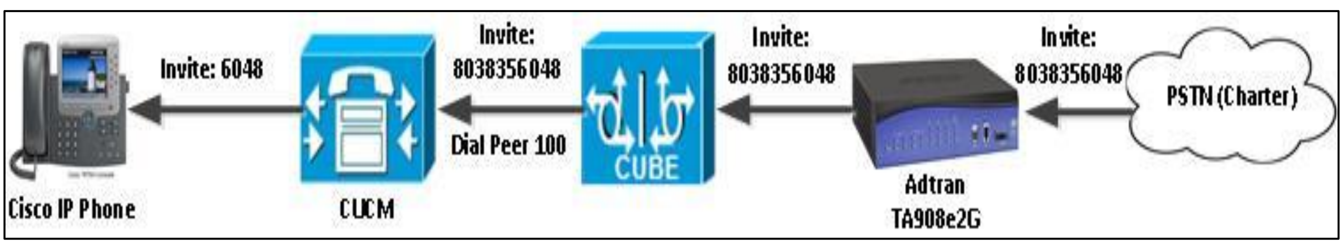

Figure 4: Inbound Voice Call

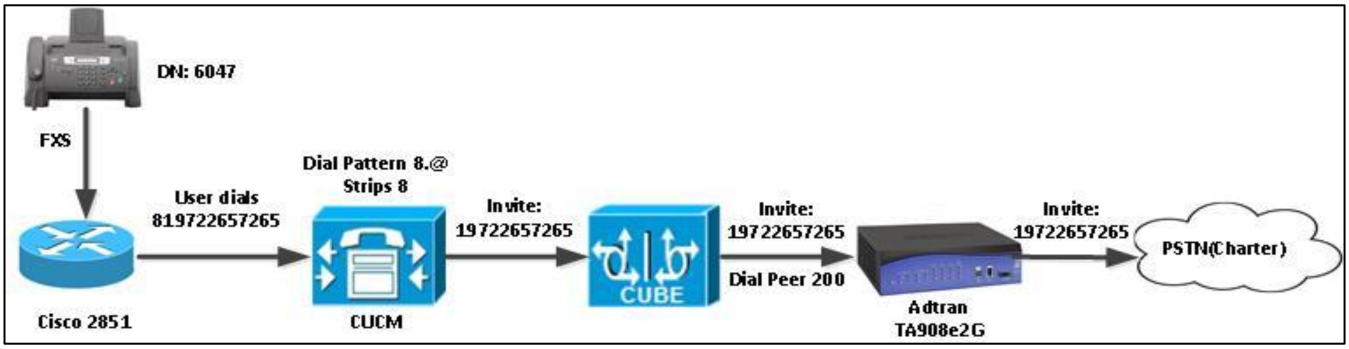

Figure 5: Outbound Fax Call

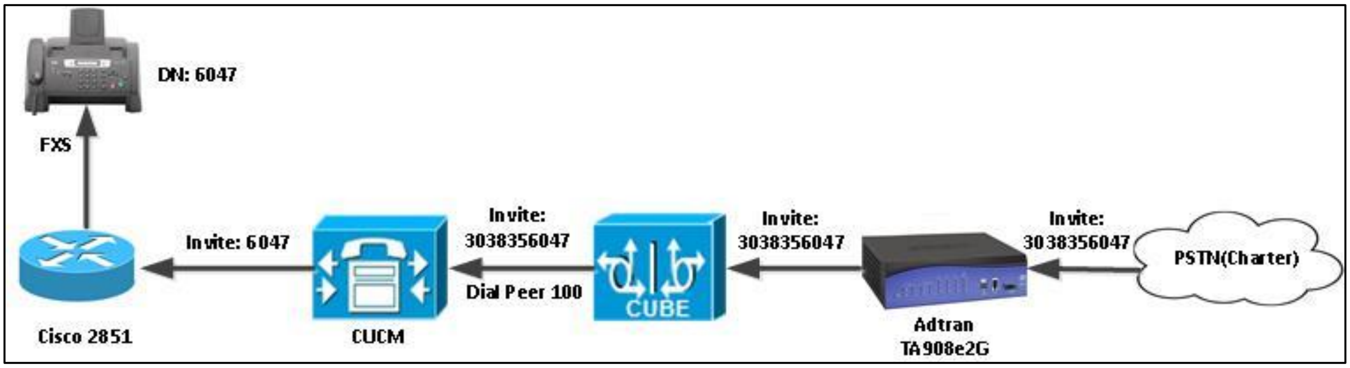

Figure 6: Inbound Fax Call

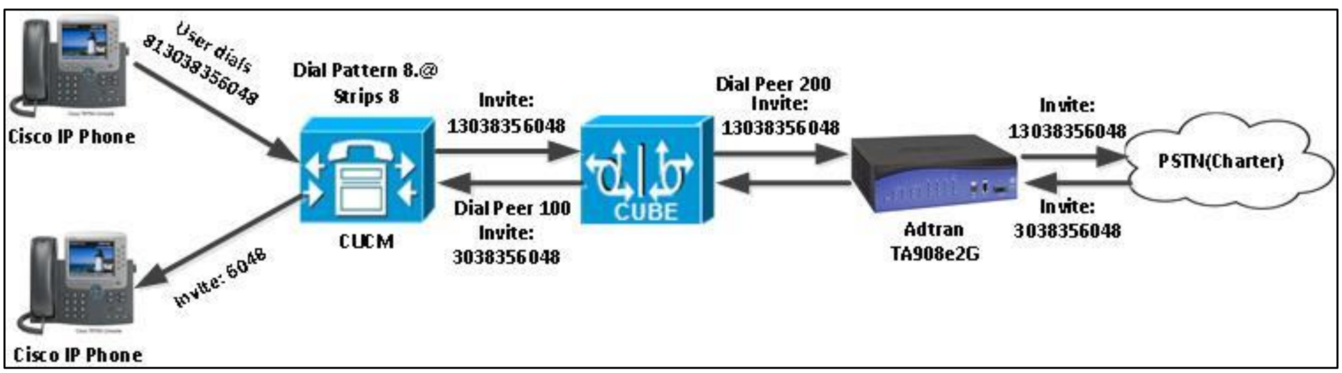

Figure 7: PBX to PBX via Charter Call

Configuration Example

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

Active Cisco UBE

```
version 15.6
service config
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keep alive disable-kernel-core
!
hostname CharterCube1
!
boot-start-marker
boot system bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
vrf definition mgmt-intf
!
no logging rate-limit
no aaa new-model
```

```
no ip domain lookup
!
subscriber templating
multilink bundle-name authenticated
!
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
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
g729 annexb-all
!
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 ISR4321/K9 sn FDO19220MW3
license boot level appxk9
license boot level uck9
```

```
!
spanning-tree extend system-id
!
redundancy
mode none
application redundancy
group 1
name b2bhaCharter
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
description Charter LAN MS4 1/0/7
ip address 10.80.11.6 255.255.255.0
media-type rj45
negotiation auto
no mop enabled
redundancy rii 1
redundancy group 1 ip 10.80.11.10 exclusive
!
```

```
interface GigabitEthernet0/0/1
description Charter WAN MS4 1/0/8
ip address 10.70.51.5 255.255.255.0
negotiation auto
no mop enabled
redundancy rii 2
redundancy group 1 ip 10.70.51.10 exclusive
!
interface GigabitEthernet0/1/0
description CUBE HA MS5 3/0/36
ip address 10.89.20.8 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
 no mop enabled
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0/0/0
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 10.70.0.0 255.255.0.0 10.70.51.1
ip route 10.80.11.0 255.255.255.0 10.80.11.1
ip route 172.16.0.0 255.255.0.0 10.80.11.1
```

```
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 200 voip
description Outbound-from IP PBX to PSTN - WAN facing
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip options-keepalive
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
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 210 voip
description outgoing call to Charter - LAN facing
```

```
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number .T
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 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 nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 500 voip
 description cube-dp incoming call from PSTN
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number 303835....
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 100
 voice-class sip options-keepalive
 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
 fax rate disable
```

```
 fax nsf 000000
 fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 100 voip
 description Inbound-from PSTN to IP PBX - LAN facing
 huntstop
 destination-pattern 303835....
 session protocol sipv2
 session target ipv4:10.80.11.3:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 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 nsf 000000
 fax protocol pass-through g711ulaw
no vad
!
!
sip-ua
 keepalive target ipv4:10.70.51.4:5060
 timers keepalive active 180
 sip-server ipv4:10.70.51.4:5060
!
!
line con 0
```

```
 stopbits 1
line aux 0
 stopbits 1
line vty 0 5
 exec-timeout 0 0
 password
 login
!
!
end
```

Standby Cisco UBE

CharterCube2#sh running-config

```
version 15.6
service config
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CharterCube2
!
boot-start-marker
boot system flash bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no logging rate-limit
no aaa new-model
no ip domain lookup
!
subscriber templating
!
multilink bundle-name authenticated
```

```
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
  asserted-id pai
  privacy pstn
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
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 ISR4321/K9 sn FDO19220MQ9
!
spanning-tree extend system-id
!
```

```
redundancy
 mode none
 application redundancy
  group 1
   name b2bhaCharter
   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
 description Charter CUBE1 LAN MS4 1/0/11
 ip address 10.80.11.7 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.11.10 exclusive
!
interface GigabitEthernet0/0/1
 description Charter CUBE1 WAN MS4 1/0/12
 ip address 10.70.51.6 255.255.255.0
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 10.70.51.10 exclusive
```

```
!
interface GigabitEthernet0/1/0
 description CUBE HA MS5 3/0/38
 ip address 10.89.20.10 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
interface Vlan1
 no ip address
 shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 10.70.51.1
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 10.80.11.0 255.255.255.0 10.80.11.1
ip route 172.16.0.0 255.255.0.0 10.80.11.1
!
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 200 voip
 description Outbound-from IP PBX to PSTN - WAN facing
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 100
 voice-class sip options-keepalive
 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
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 210 voip
 description outgoing call to Charter - LAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number .T
 voice-class codec 1
```

```
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 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 nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 500 voip
 description cube-dp incoming call from PSTN
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 303835....
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 100
 voice-class sip options-keepalive
 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
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
```

```
!
dial-peer voice 100 voip
 description Inbound-from PSTN to IP PBX - LAN facing
 huntstop
 destination-pattern 303835....
 session protocol sipv2
 session target ipv4:10.80.11.3:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 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 nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
!
sip-ua
 keepalive target ipv4:10.70.51.4:5060
 timers keepalive active 180
 sip-server ipv4:10.70.51.4:5060
!
!
line con 0
 stopbits 1
line aux 0
```

```
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password
 login
!
!
End
```

Configuring Cisco Unified Communications Manager

Cisco UCM Version

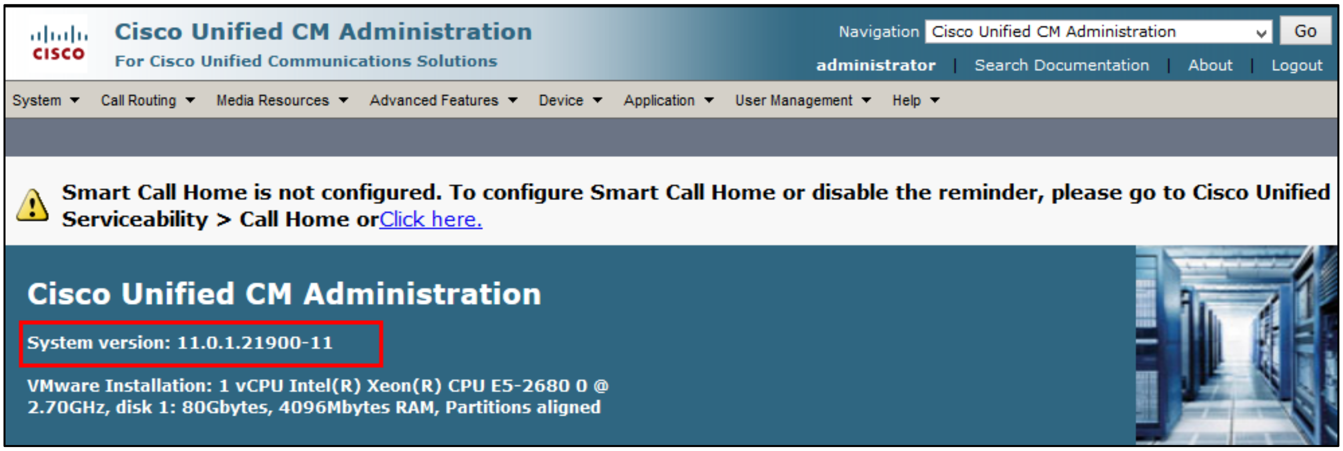

Figure 8: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System > Service Parameters

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

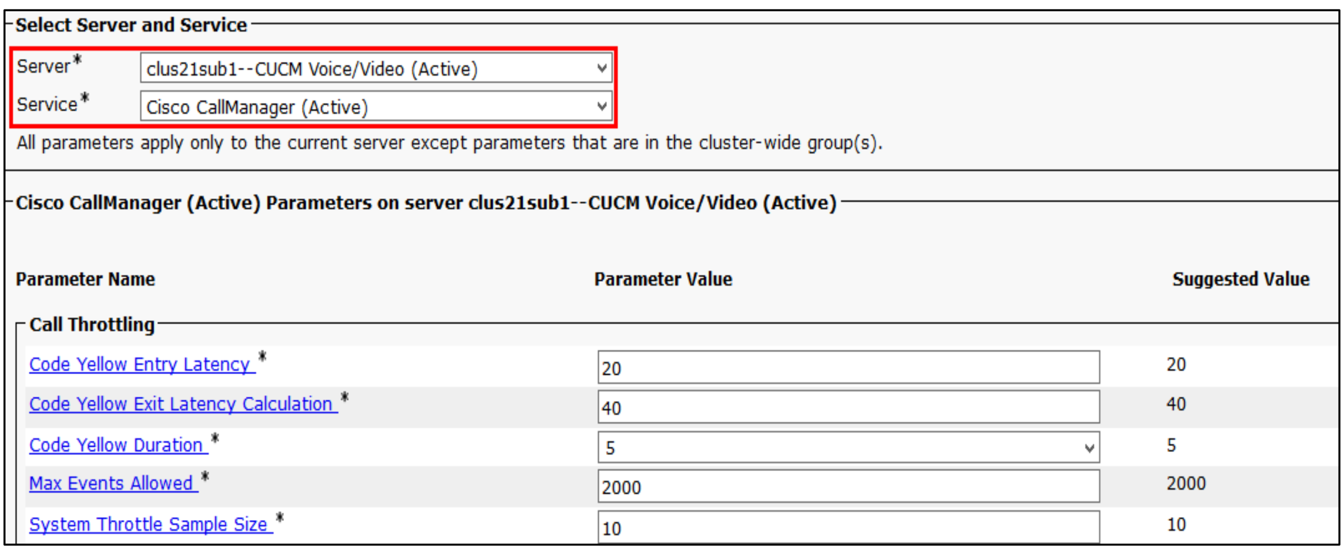

Figure 9: Service Parameters

Offnet Calls via Charter SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and Charter Network and calls are routed via Cisco UBE

SIP Trunk Security Profile

Navigation: System > Security > SIP Trunk Security Profile

1. Name*= Charter Non Secure SIP Trunk Profile
2. Description = non Secure SIP Trunk Profile authenticated by null String

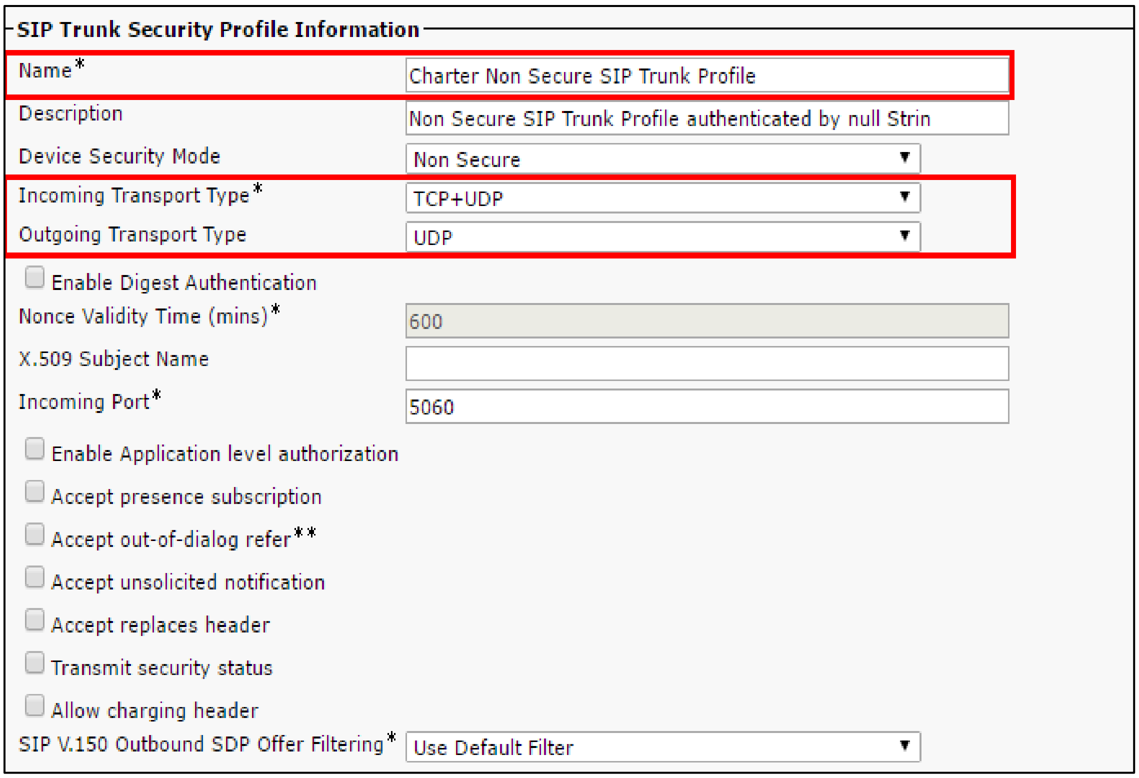

Figure 10: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to Charter 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*= Charter SIP Profile
2. Description = Default SIP Profile

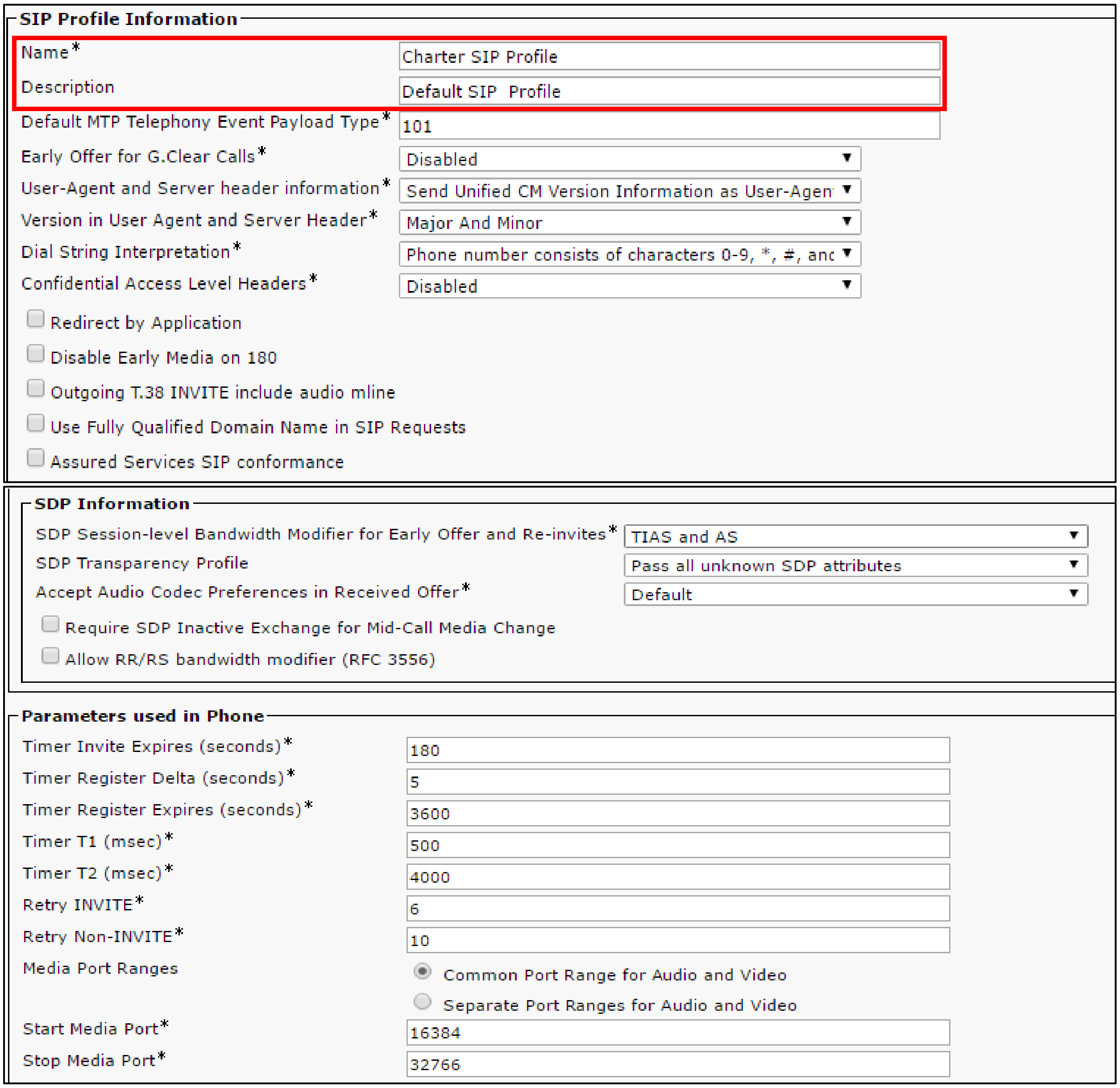

Figure 11: SIP Profile

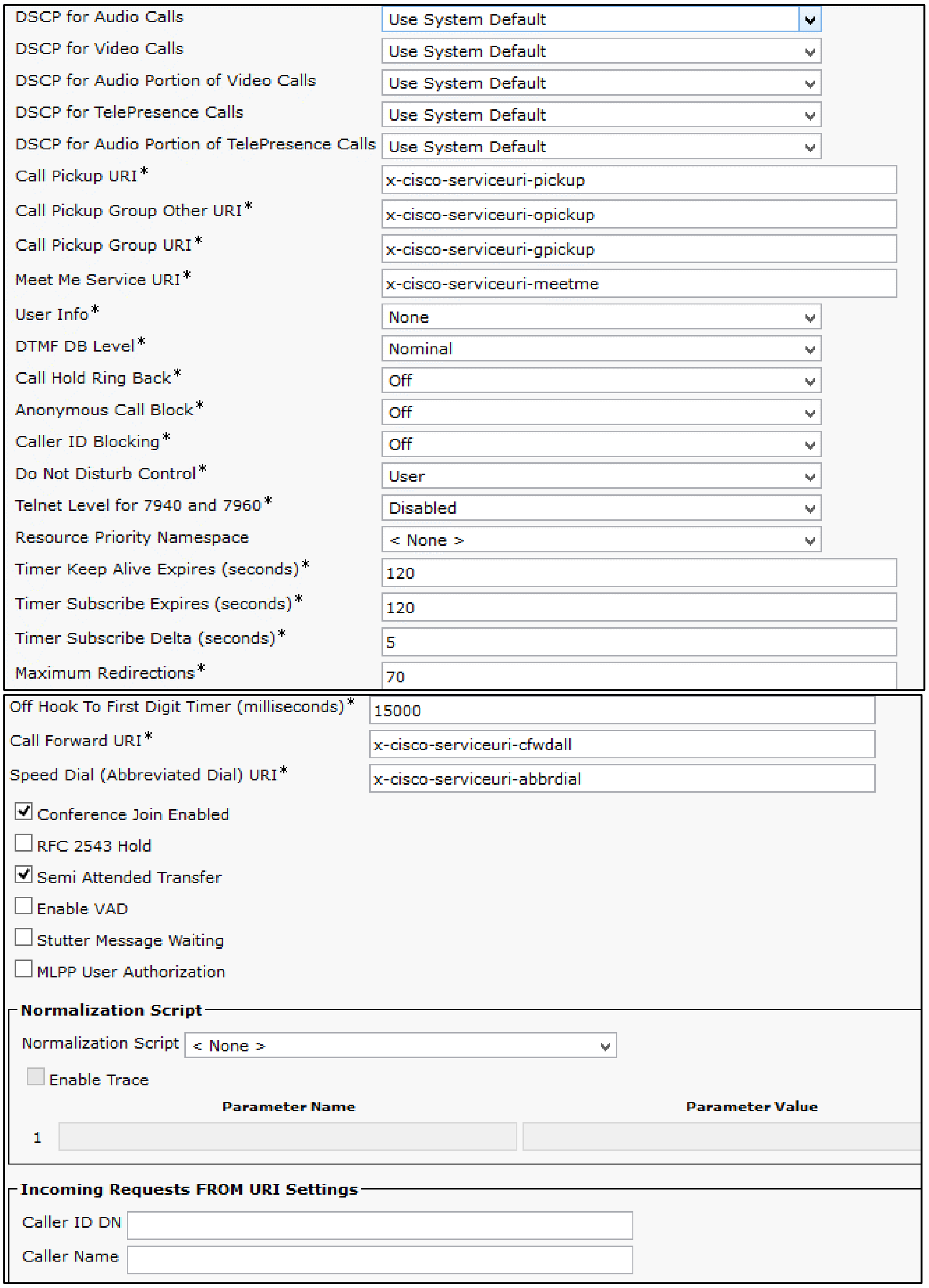

Figure 12: SIP Profile (Cont.)

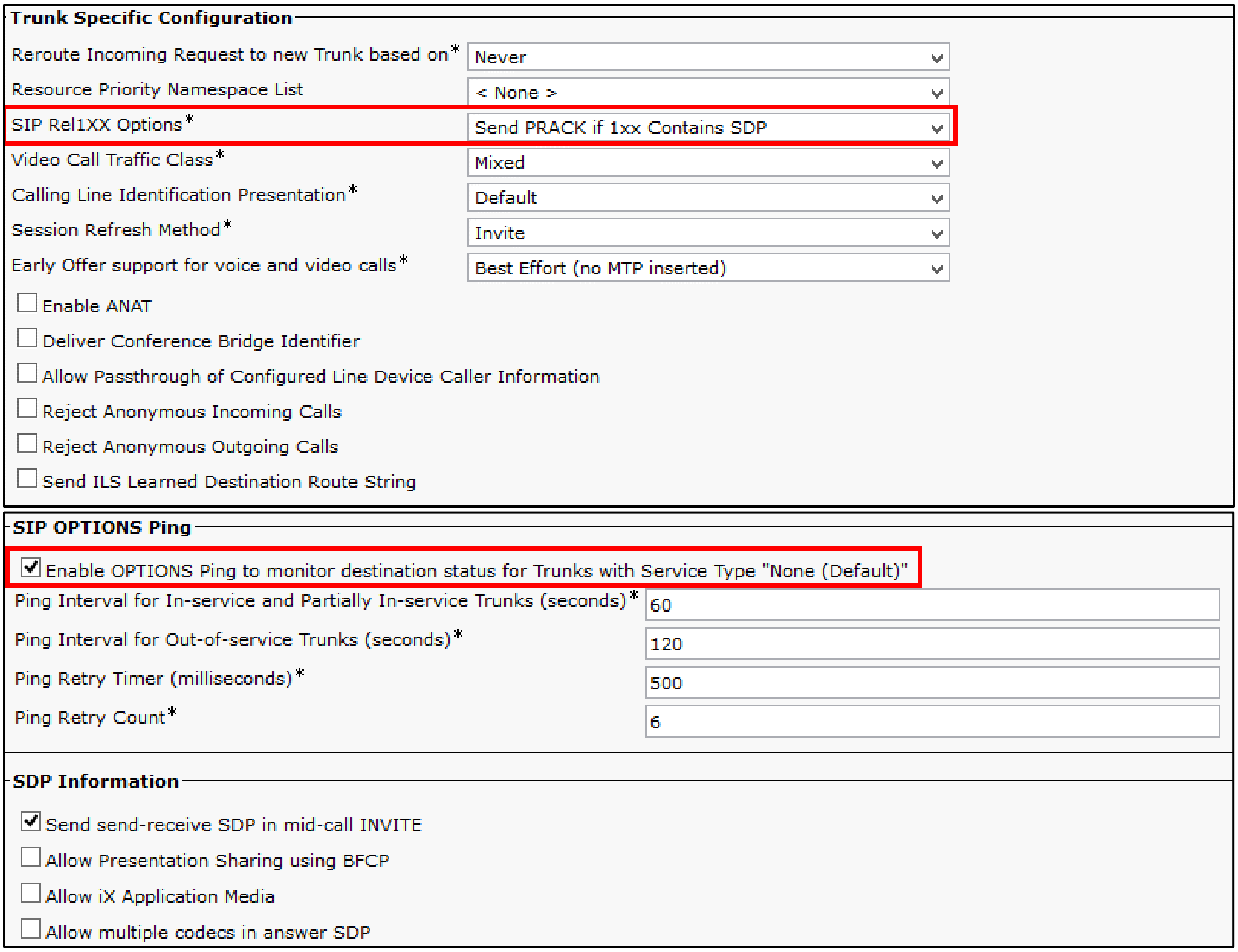

Figure 13: SIP Profile (Cont.)

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 Cisco UBE

Navigation: Device > Trunk

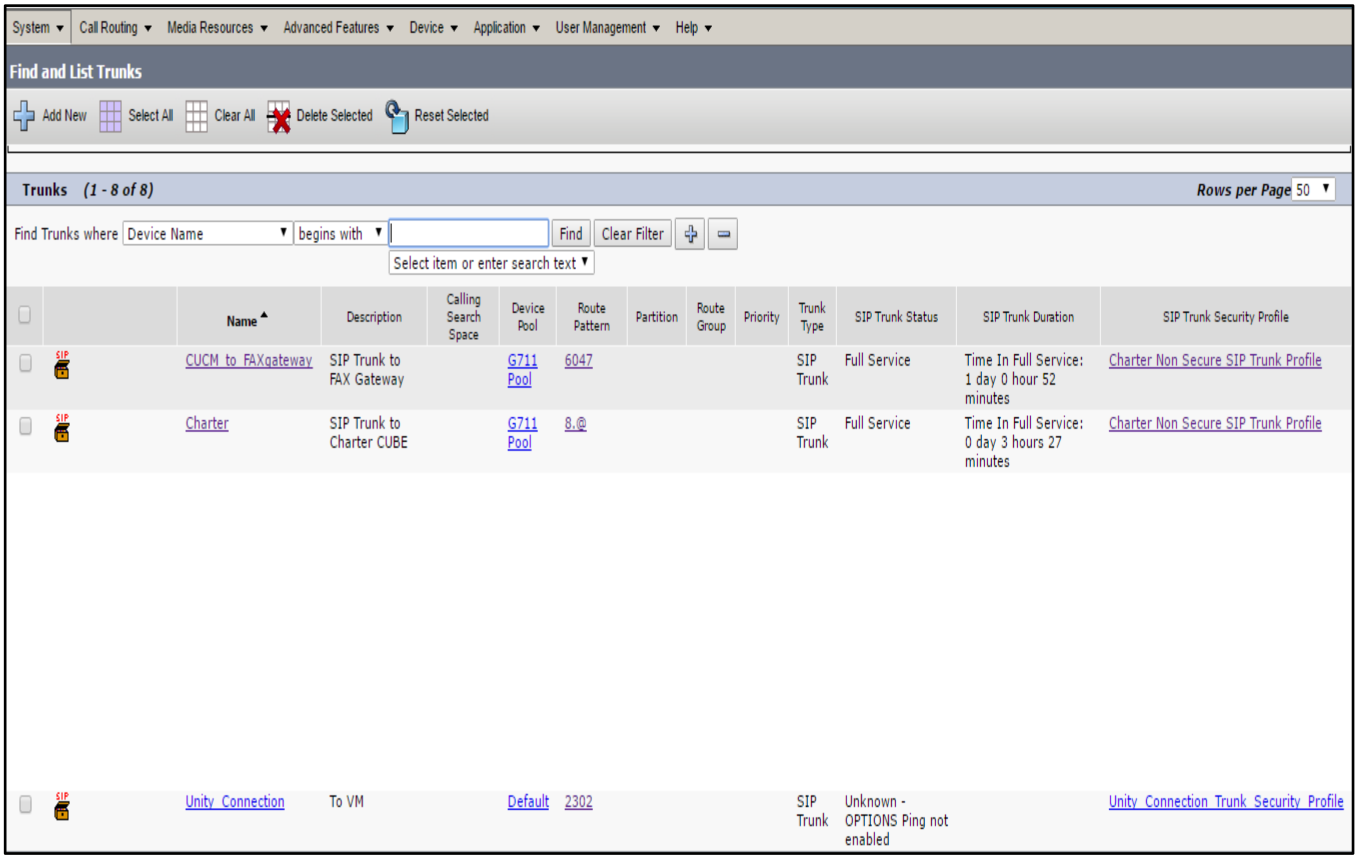

Figure 14: SIP Trunks List

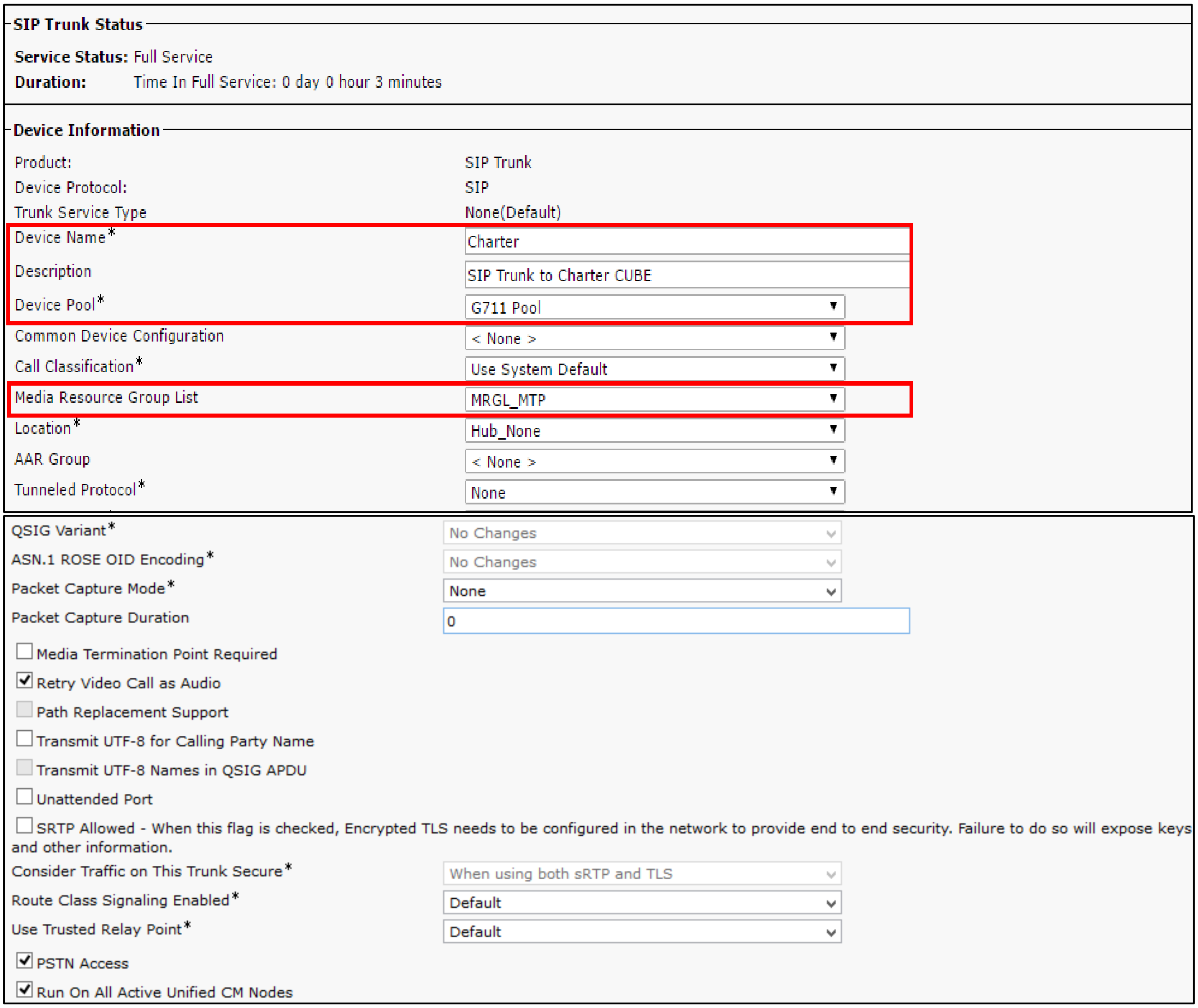

Figure 15: SIP Trunk to Cisco UBE

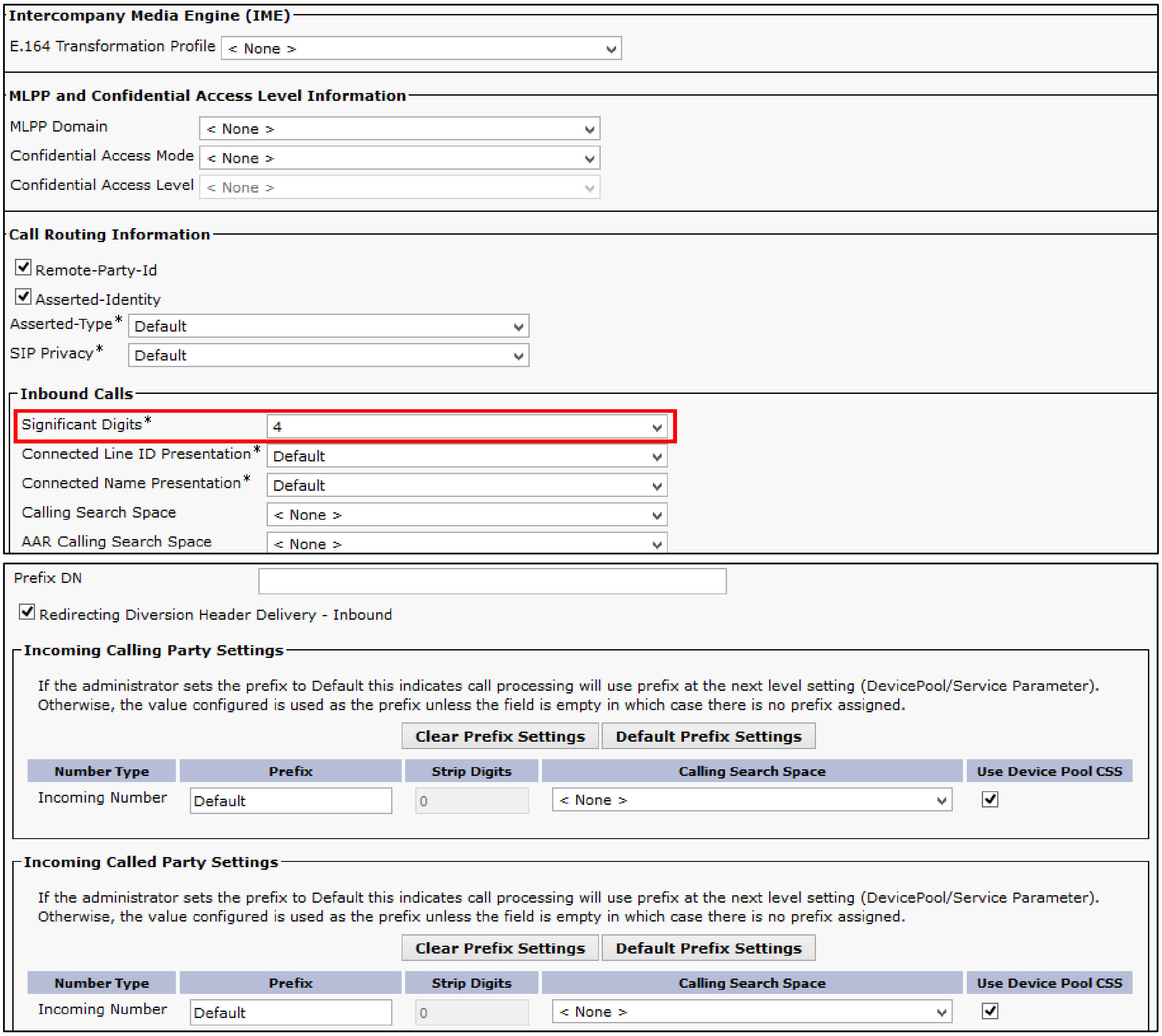

Figure 16: SIP Trunk to Cisco UBE (Cont.)

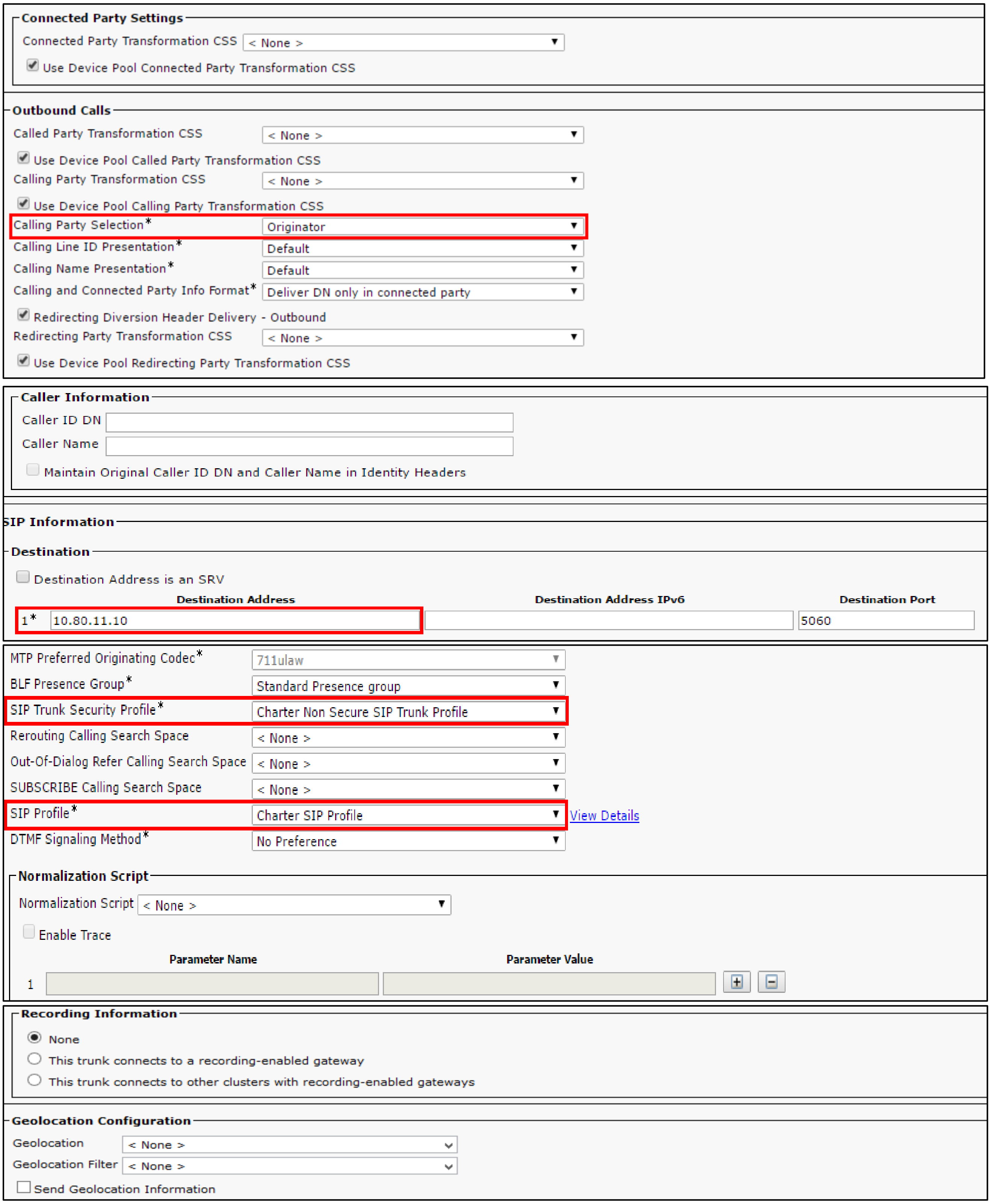

Figure 17: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	Charter	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	10.80.11.10	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	Charter Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Charter 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 “8”.1+10 digits number to access PSTN via Cisco UBE
    - “8” is removed before sending to Cisco UBE
- For FAX call, Access Code “8”+ 1+10 digits number is used at Cisco Fax gateway
    - “8” is removed at Cisco UCM
    - The rest of the number is sent to Cisco UBE to Charter network
- Incoming fax call to 6047 will be sent to Cisco Fax gateway

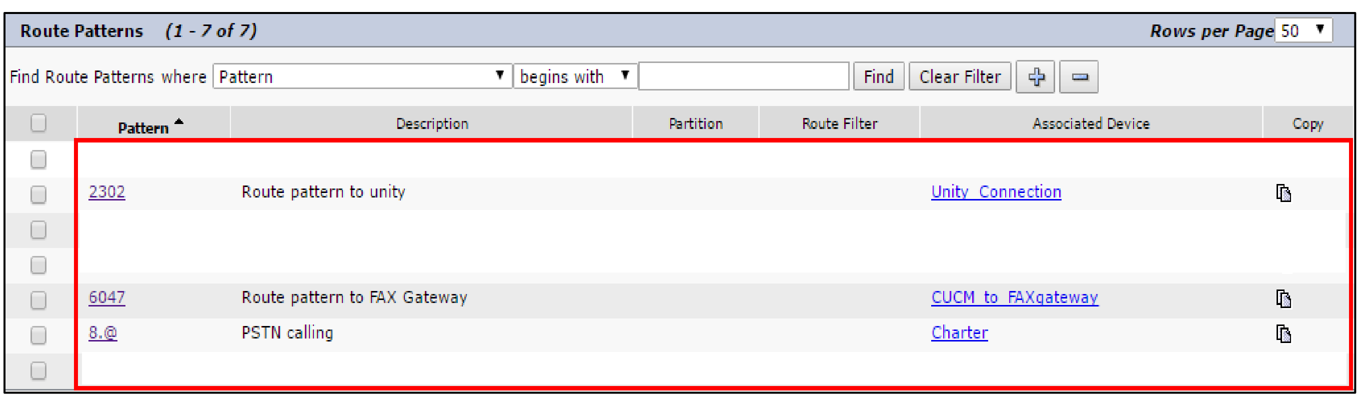

Figure 18: Route Patterns List

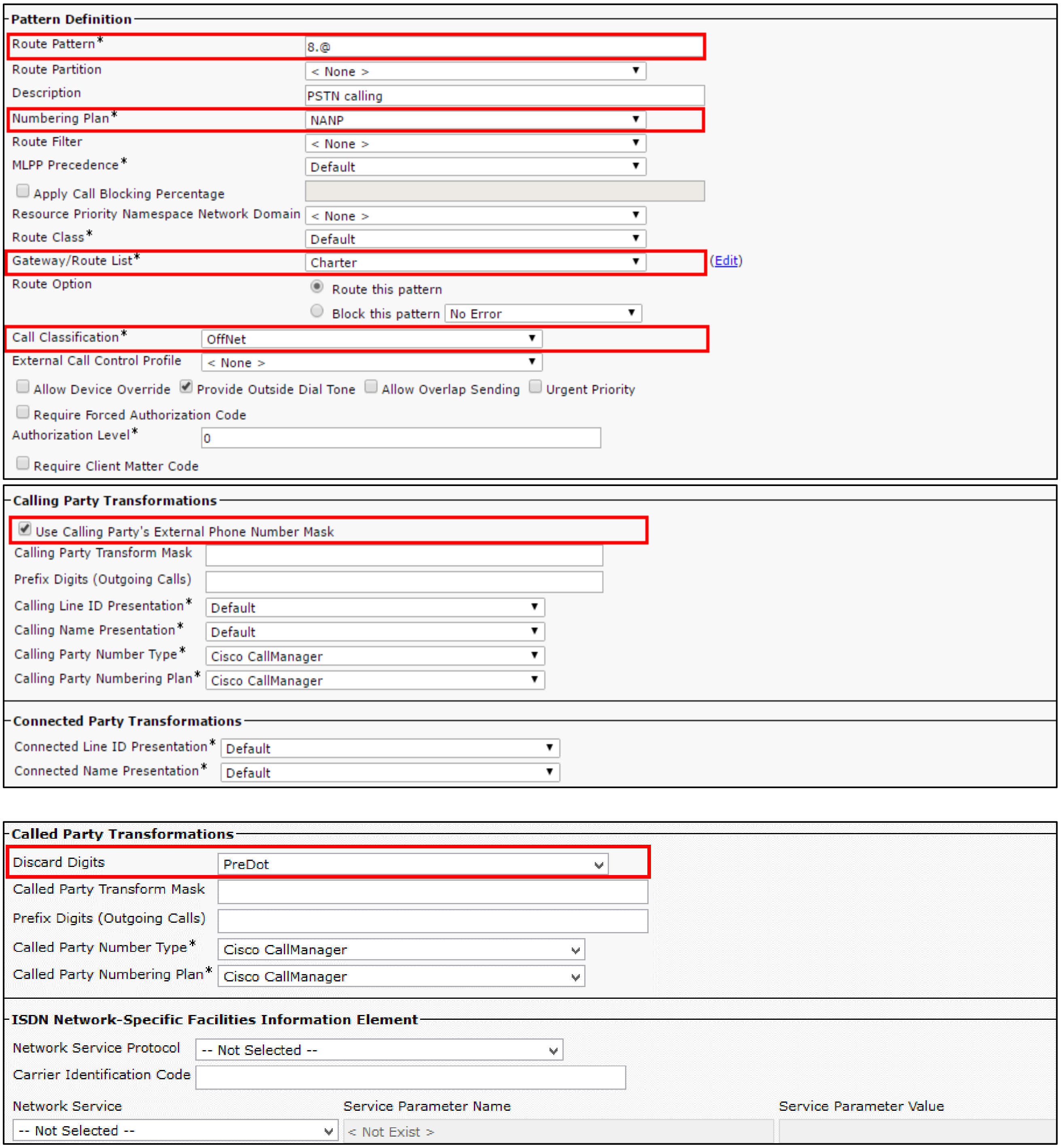

Figure 19: Route Pattern for Voice

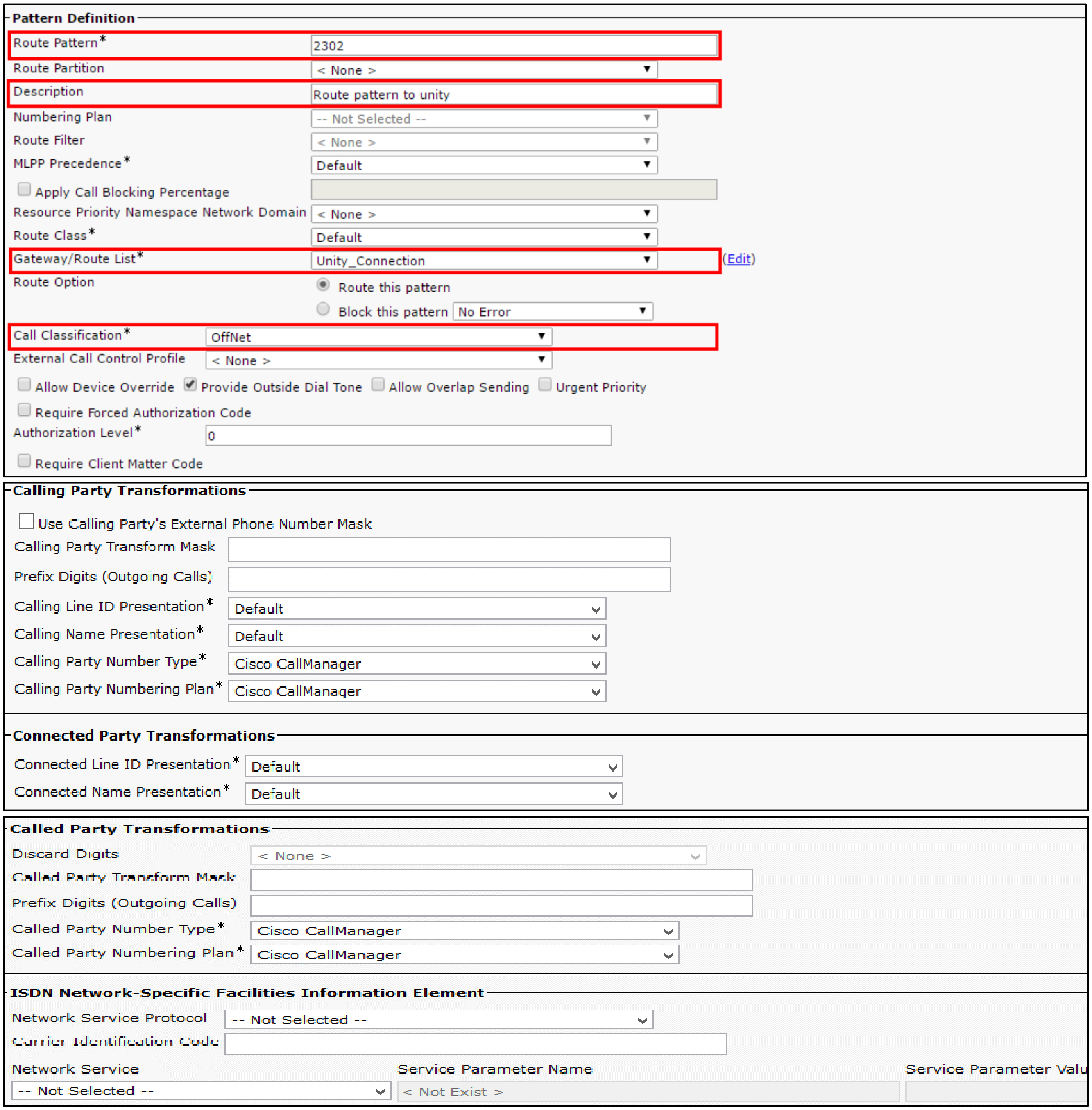

Figure 20: Route Pattern for Voice (Cont.)

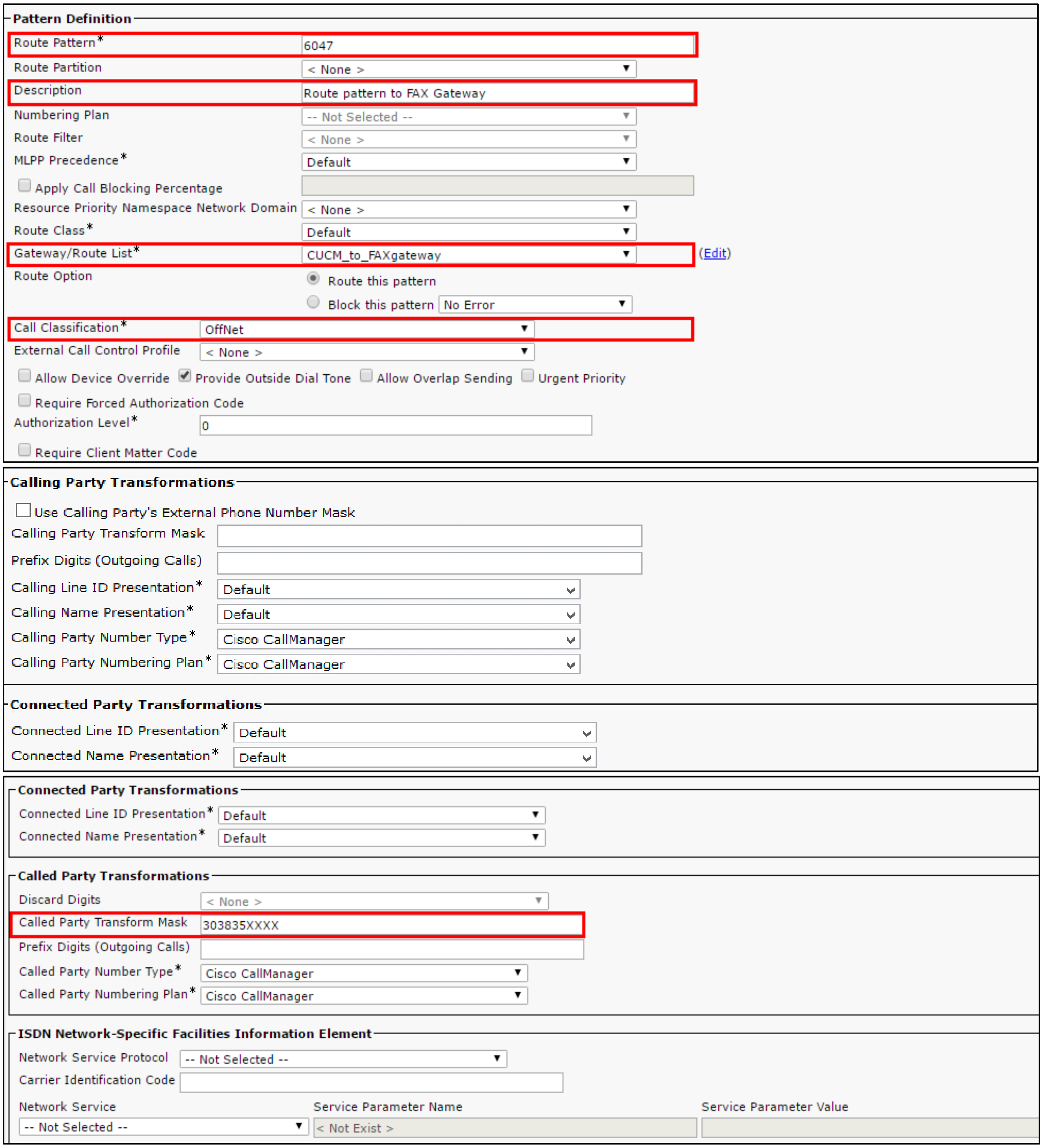

Figure 21: Route Pattern for Fax

Explanation

Setting	Value	Description
Route Pattern	8.@ for Voice & International Calls, 6047 for Fax Call and 2302 for Unity Connection	Specify appropriate Route Pattern
Gateway/Route List	Charter for Route Pattern 8. @, 6047 for SIP Trunk To Fax Gateway and 2032 for Unity Connection.	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 8.@	North American Numbering Plan
Call Classification	OffNet for Route Pattern 8.@, 6047 and 2032	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 8.@	Specifies how to modify digit before they are sent to Charter 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

Important Information

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