Application Note

IntelePeer SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR 4321 [IOS - 15.6(1) S] using SIP

April 21, 2016

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Introduction

Service Providers today, such as IntelePeer, 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.

IntelePeer 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 IntelePeer network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS 15.6(1)S can be used. The Cisco Unified Border Element 15.6(1)S provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to IntelePeer IP 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 IntelePeer 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) on ISR 4321/K9 [IOS - 15.6(1)S] for connectivity to IntelePeer SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (IntelePeer).
- Testing was performed in accordance to IntelePeer 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 IntelePeer 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 IntelePeer 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

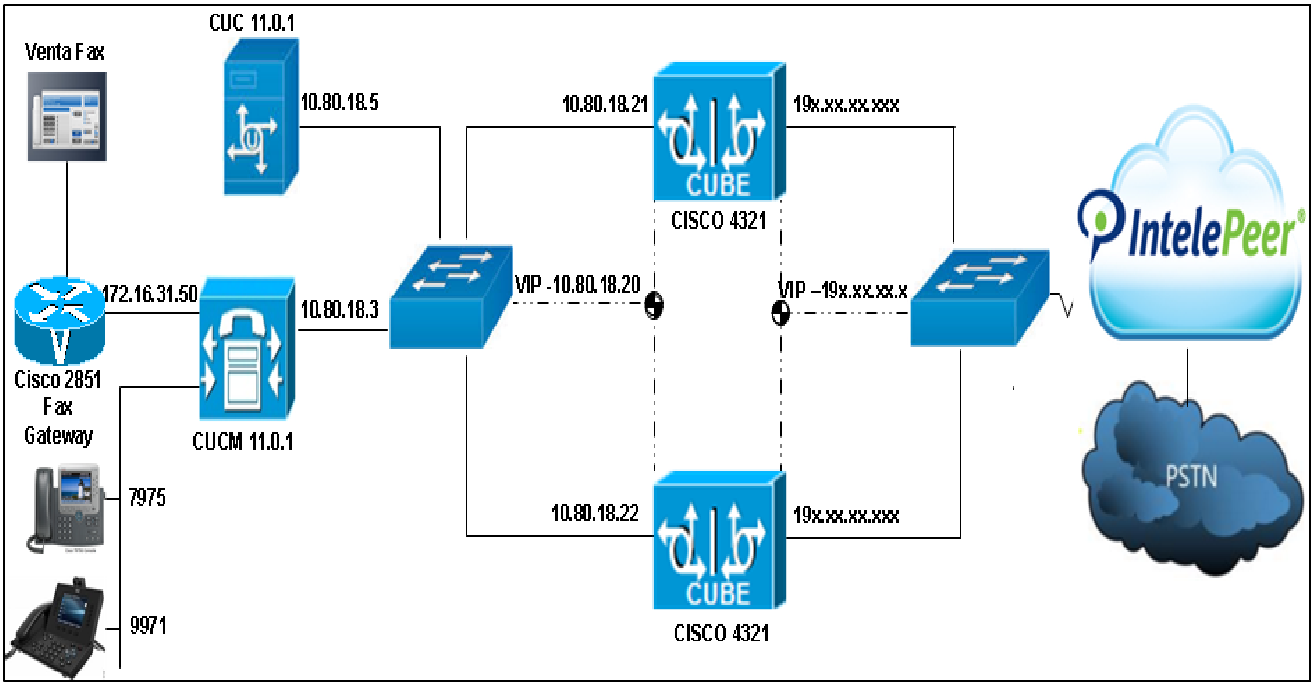

Figure 1: Network Topology

- Cisco IP Phones 7975,7965 and 9971 phones are the devices primarily used throughout the testing to place or receive calls
- VentaFax Soft Client is used to perform all fax related scenarios. The fax client is connected to SIP Gateway via FXS port which in turn communicates with Cisco UCM over SIP.

System Components

Hardware Requirements

- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 Router as CUBE
- Cisco 2851 Fax Gateway
- IP Phones 9971(SIP),7965 (SCCP) and 7975 (SIP)

Software Requirements

- Cisco Unified Communications Manager 11.0.1
- Cisco Unity Connection 11.0.1
- IOS 15.6(1)S for ISR 4321/K9 Cisco Unified Border Element
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway

Features

Features Supported

- Incoming and outgoing off-net calls using G711ULaw and G729
- Call Hold
- Call Transfer (unattended and attended)
- Call Forward (all, busy and no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF (RFC2833)
- Media flow-through on Cisco UBE
- Fax (G.711 pass-through and T.38)

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer
- Fax re-invite is not supported by Service Provider

Caveats

- CLID is not updated on PSTN phones for transfer (attended and unattended) OffNet PSTN scenarios. Caller ID is not updated at PSTN once transfer is completed by PBX. Cisco UBE modify PAI/PPI header and forward to network in the tested release. CISCO BUG ID: CSCuv04539
- Transfer scenario from CPE observed no way speech path issue between PSTN phones while mid call codec negotiated from g729 to g711ulaw
- Privacy: ID sent from PBX has stripped from IntelePeer while make call from PBX Phone A to PBX phone B. The caller ID shows as anonymous when making call from PBX to PSTN.
- Intermittent no way speech path observed for an inbound calls

Configuration

Configuring Cisco Unified Border Element

Network Interface

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

```
interface GigabitEthernet0/0/0
 ip address 10.80.18.21 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.20 exclusive
!
interface GigabitEthernet0/0/1
 ip address 192.65.x.x 255.255.255.128
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.x.x 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
 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

privacy-policy passthru

g729 annexb-all

!

Explanation

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

Media Passing through Cisco UBE (Media flow-through vs. Media flow-around)

Default Cisco UBE configuration enables Cisco UBE to work in flow-through mode (this test uses the flow-through mode). In order to enable flow-around mode, perform the following actions:

voice service voip

media flow-around

Codecs

G729 is used as the preferred codec for this testing and changed the codecs according to the test plan description

voice class codec 1

codec preference 1 g711ulaw

codec preference 2 g729r8

!

voice class codec 2

codec preference 1 g729r8

codec preference 2 g729br8

codec preference 3 g711ulaw

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
 translation-profile outgoing e16four
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 session transport tcp
 voice-class codec 2
 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 14400
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 210 voip
 description outgoing call to intelepeer - LAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 incoming called-number .T
 voice-class codec 2
 voice-class sip asserted-id pai
```

```
 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 14400
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 500 voip
 description cube-dp incomming call from PSTN
 huntstop
 session protocol sipv2
 session transport tcp
 incoming called-number +1949204....
 voice-class codec 2
 voice-class sip asserted-id pai
 voice-class sip profiles 100
 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 14400
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 100 voip
 description Inbound-from PSTN to IP PBX - LAN facing
 translation-profile outgoing noe16four
```

```
huntstop
destination-pattern +1949204....
session protocol sipv2
session target ipv4:10.80.18.3:5060
voice-class codec 2
voice-class sip asserted-id pai
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 14400
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 “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 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 IntelePeer, Caller dial 9 prefix followed by the 10 digit DID provided by IntelePeer, 9 was stripped and the ten-digit number was send to Cisco UBE, Cisco UBE translate the 10 digits extension number to its full ten-digits DID with E.164 format under Dial Peer 200 and send to IntelePeer network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

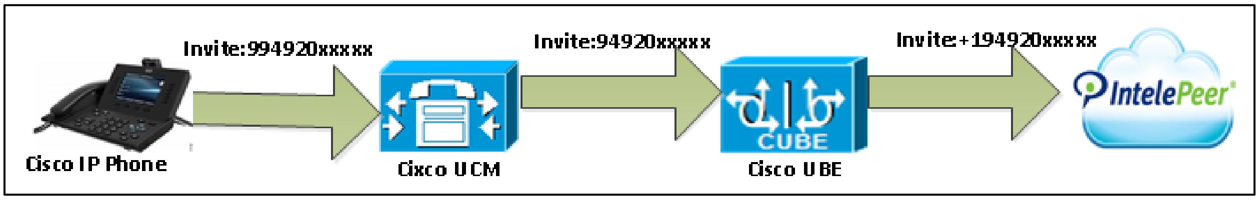

Figure 2: Outbound Voice Call

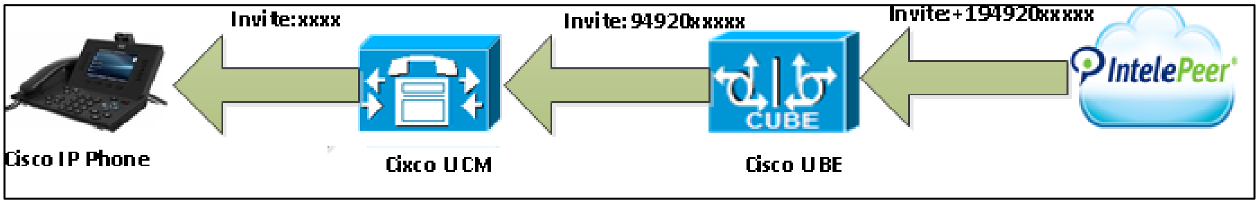

Figure 3: Inbound Voice Call

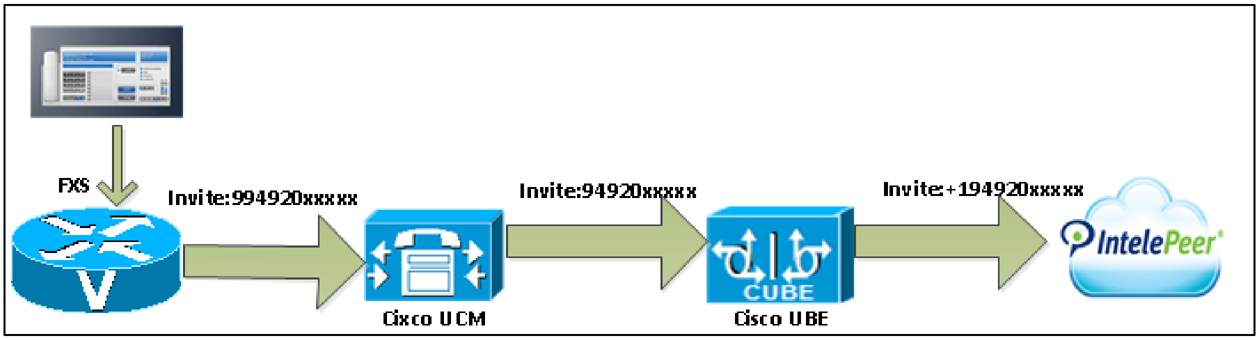

Figure 4: Outbound Fax Call

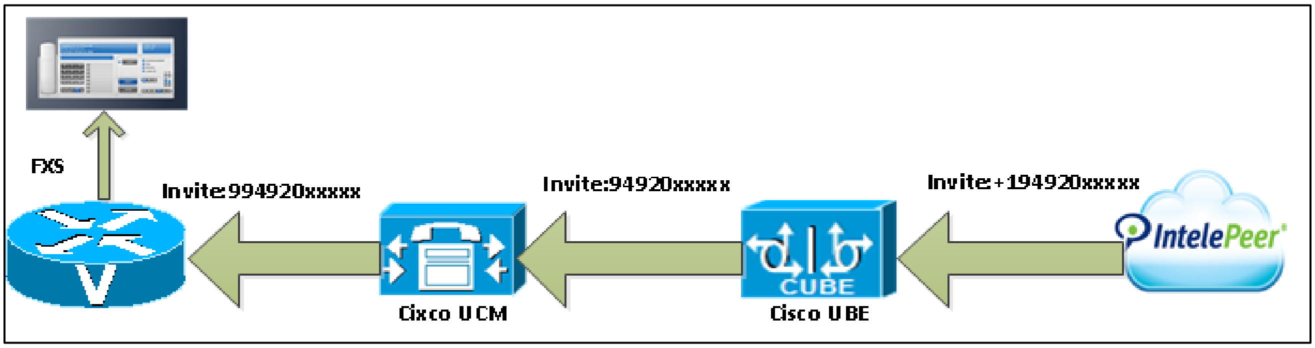

Figure 5: Inbound Fax Call

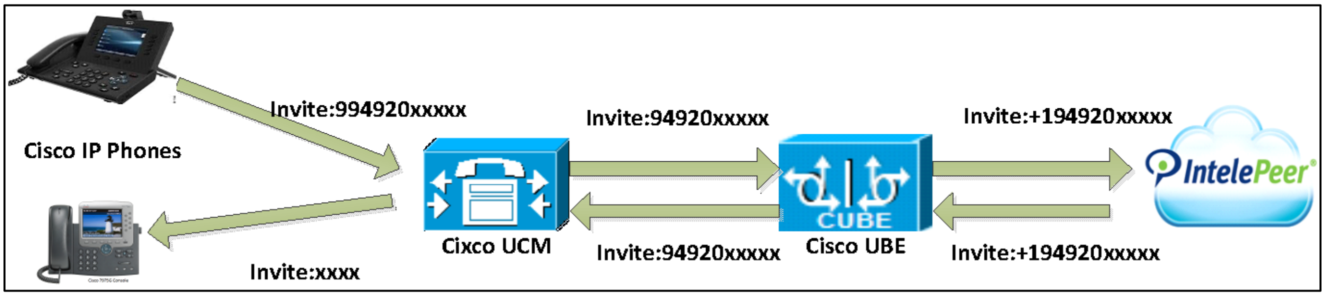

Figure 6: PBX to PBX via IntelePeer Call

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: cisco

Password:

IN_CUBE1#sh running

Building configuration...

Current configuration : 6624 bytes

!

! Last configuration change at 07:45:49 UTC Fri Apr 15 2016 by cisco

!

version 15.6

service timestamps debug datetime msec

```
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname IN_CUBE1
!
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
!
enable secret 5
enable password
!
no aaa new-model
!
!
!
!
!
!
!
```

```
!
!
no ip domain lookup
ip domain name tekvizion.com
!
!
!
!
!
!
!
!
!
!
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 codec 2
 codec preference 1 g729r8
 codec preference 2 g729br8
 codec preference 3 g711ulaw
!
!
voice class sip-profiles 100
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:+1949204\1@\2"
!
!
```

```
!
!
!
voice translation-rule 1
 rule 1 /\(^..........$\)/ /+1\1/
!
voice translation-rule 2
 rule 1 /\(^..........$\)/ /+1\1/
 rule 2 /\(^............$\)/ /+1\1/
 rule 3 /\(^442037……\)/ /+\1/
!
voice translation-rule 3
 rule 1 /\+1\(..........\)/ /\1/
!
voice translation-rule 4
 rule 1 /\+1\(..........\)/ /\1/
!
!
voice translation-profile e16four
translate calling 1
translate called 2
!
voice translation-profile noe16four
translate calling 3
translate called 4
!
!
!
license udi pid ISR4321/K9 sn FDO19220MSQ
```

```
!
spanning-tree extend system-id
!
username cisco privilege 15 password 0 tekV1z10n
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bhaIntelePeer
   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
!
!
translation-rule 1
```

```
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
interface GigabitEthernet0/0/0
 ip address 10.80.18.21 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.20 exclusive
!
```

```
interface GigabitEthernet0/0/1
 ip address 192.65.XX.XXX 255.255.255.128
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.x.x exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA MS5 3/0/37
 ip address 10.89.20.9 255.255.255.0
 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
ip route 0.0.0.0 0.0.0.0 192.65.XX.XXX
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
ip route 172.16.31.0 255.255.255.0 10.80.18.1
```

```
!
!
!
!
!
!
control-plane
!
 !
 !
 !
 !
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
!
!
dial-peer voice 200 voip
 description Outbound-from IP PBX to PSTN - WAN facing
 translation-profile outgoing e16four
 huntstop
 destination-pattern .T
 session protocol sipv2
```

```
 session target sip-server
 session transport tcp
 voice-class codec 2
 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 14400
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 210 voip
 description outgoing call to intelepeer - LAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 incoming called-number .T
 voice-class codec 2
 voice-class sip asserted-id pai
 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 14400
```

```
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 500 voip
 description cube-dp incomming call from PSTN
 huntstop
 session protocol sipv2
 session transport tcp
 incoming called-number +19492…...
 voice-class codec 2
 voice-class sip asserted-id pai
 voice-class sip profiles 100
 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 14400
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 100 voip
 description Inbound-from PSTN to IP PBX - LAN facing
 translation-profile outgoing noe16four
 huntstop
 destination-pattern +19492…...
 session protocol sipv2
 session target ipv4:10.80.18.3:5060
```

```
 voice-class codec 2
 voice-class sip asserted-id pai
 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 14400
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
!
sip-ua
 keepalive target ipv4:68.68.XXX.xx:5060
 timers keepalive active 180
 sip-server ipv4:68.68.XX.xx:5060
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password xxxx
 login local
!
!
End
```

Standby Cisco UBE

```
User Access Verification
cisco
Password:
IN_CUBE2#sh run
Building configuration...
Current configuration : 5630 bytes
!
! Last configuration change
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname IN_CUBE2
!
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
```

```
!
enable secret 5 $1$o64N$3mgK4kNZKFwv3VUAouCDm1
enable password cisco
!
no aaa new-model
!
!
!
!
!
!
!
!
!
no ip domain lookup
ip domain name tekvizion.com
 --More--         !
!
!
!
!
!
!
!
!
!
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 codec 2
 codec preference 1 g729r8
 codec preference 2 g729br8
 codec preference 3 g711ulaw
!
!
!
!
!
voice class sip-profiles 100
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:+1949204\1@\2"
!
!
voice translation-rule 1
 rule 1 /\(^..........$\)/ /+1\1/
!
voice translation-rule 2
 rule 1 /\(^..........$\)/ /+1\1/
 rule 2 /\(^............$\)/ /+1\1/
 rule 3 /\(^442037……\)/ /+\1/
!
voice translation-rule 3
 rule 1 /\+1\(..........\)/ /\1/
!
voice translation-rule 4
 rule 1 /\+1\(..........\)/ /\1/!
```

```
!
voice translation-profile e16four
translate calling 1
translate called 2
!
voice translation-profile noe16four
translate calling 3
translate called 4
!
!
!
license udi pid ISR4321/K9 sn FDO19220MQ9
!
spanning-tree extend system-id
!
username cisco privilege 15 password 0
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bhaIntelePeer
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/1/0 protocol 1
   data GigabitEthernet0/1/0
   track 1 shutdown
   track 2 shutdown!
!
```

```
!
!
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
```

```
!
interface GigabitEthernet0/0/0
 ip address 10.80.18.22 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.20 exclusive
!
interface GigabitEthernet0/0/1
 ip address 192.65.XX.XXX 255.255.255.128
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.XX.XXX 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
 shutdown
 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
ip route 0.0.0.0 0.0.0.0 192.65.XX.XXX
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
!
!
!
!
!
!
control-plane
!
 !
 !
 !
 --More--         !
!
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
 translation-profile outgoing e16four
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target sip-server
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 session target sip-server
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```

```
 voice-class codec 2
 voice-class sip asserted-id pai
 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 14400
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 500 voip
 description cube-dp incomming call from PSTN
 huntstop
 session protocol sipv2
 session transport tcp
 incoming called-number +19492…...
 voice-class codec 2
 voice-class sip asserted-id pai
 voice-class sip profiles 100
 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 14400
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
```

```
!
dial-peer voice 100 voip
 description Inbound-from PSTN to IP PBX - LAN facing
 translation-profile outgoing noe16four
 huntstop
 destination-pattern +19492…...
 session protocol sipv2
 session target ipv4:10.80.18.3:5060
 voice-class codec 2
 voice-class sip asserted-id pai
 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 14400
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
sip-ua
 keepalive target ipv4:68.68.123.174:5060
 timers keepalive active 180
 sip-server ipv4:68.68.123.174:5060
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
```

```
line vty 0 4
 password xxxx
 login local
!
!
end

IN_CUBE2#
```

Configuring Cisco Unified Communications Manager

Cisco UCM Version

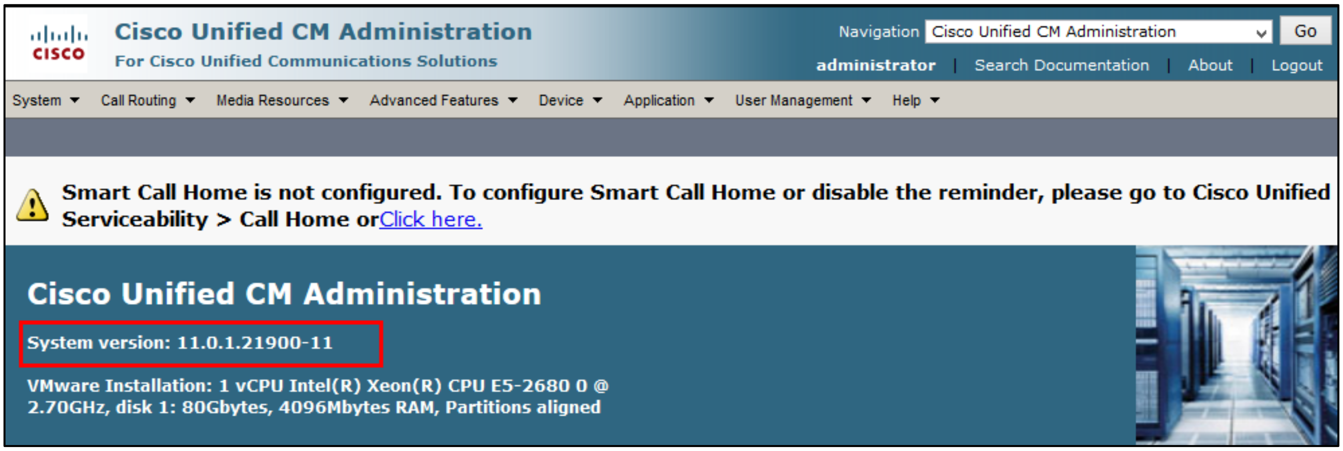


Figure 7: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation Path: System > Service Parameters

Select Server* = Clus28Sub1--CUCM Voice/Video (Active)

Select Service*= Cisco CallManager (Active)

All other fields are set to default values

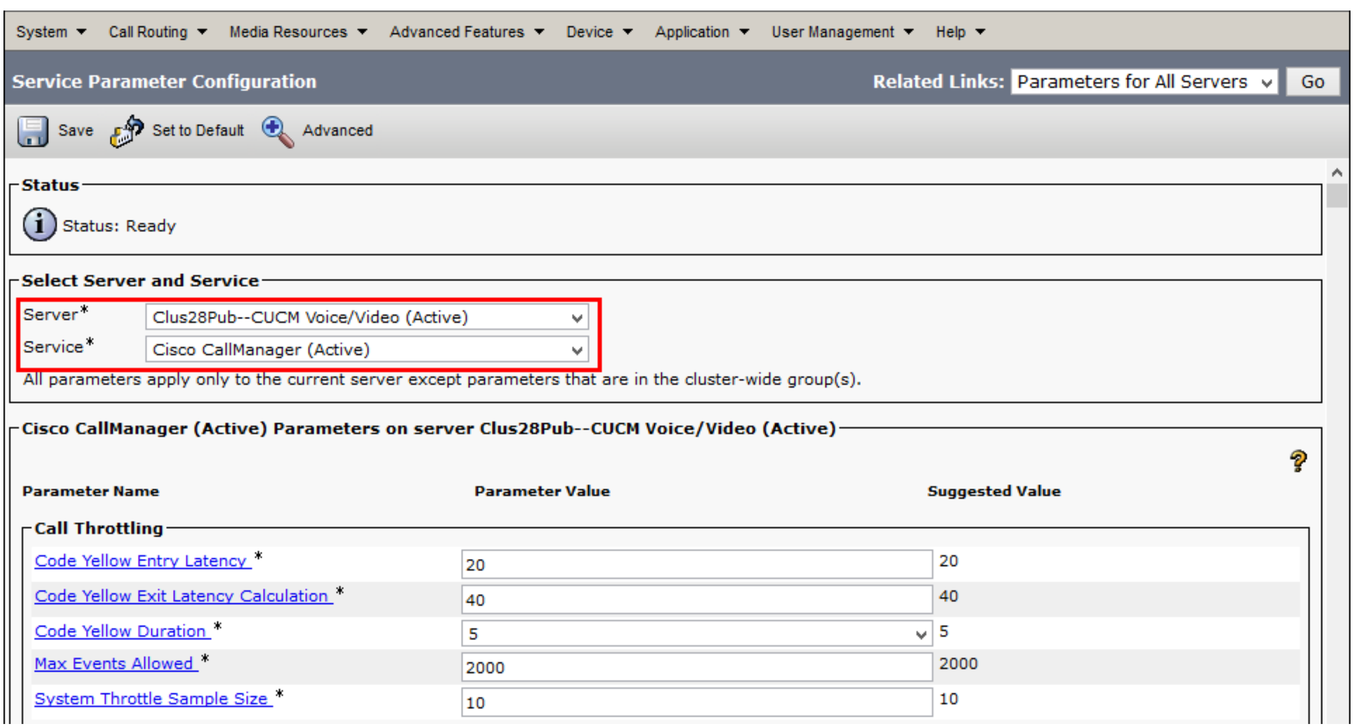


Figure 8: Service Parameters

Offnet Calls via IntelePeer SIP Trunk

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

SIP Trunk Security Profile

Navigation Path: System > Security > SIP Trunk Security Profile

Name*= Intelepeer Non Secure SIP Trunk Profile

Description = non Secure SIP Trunk Profile authenticated by null String

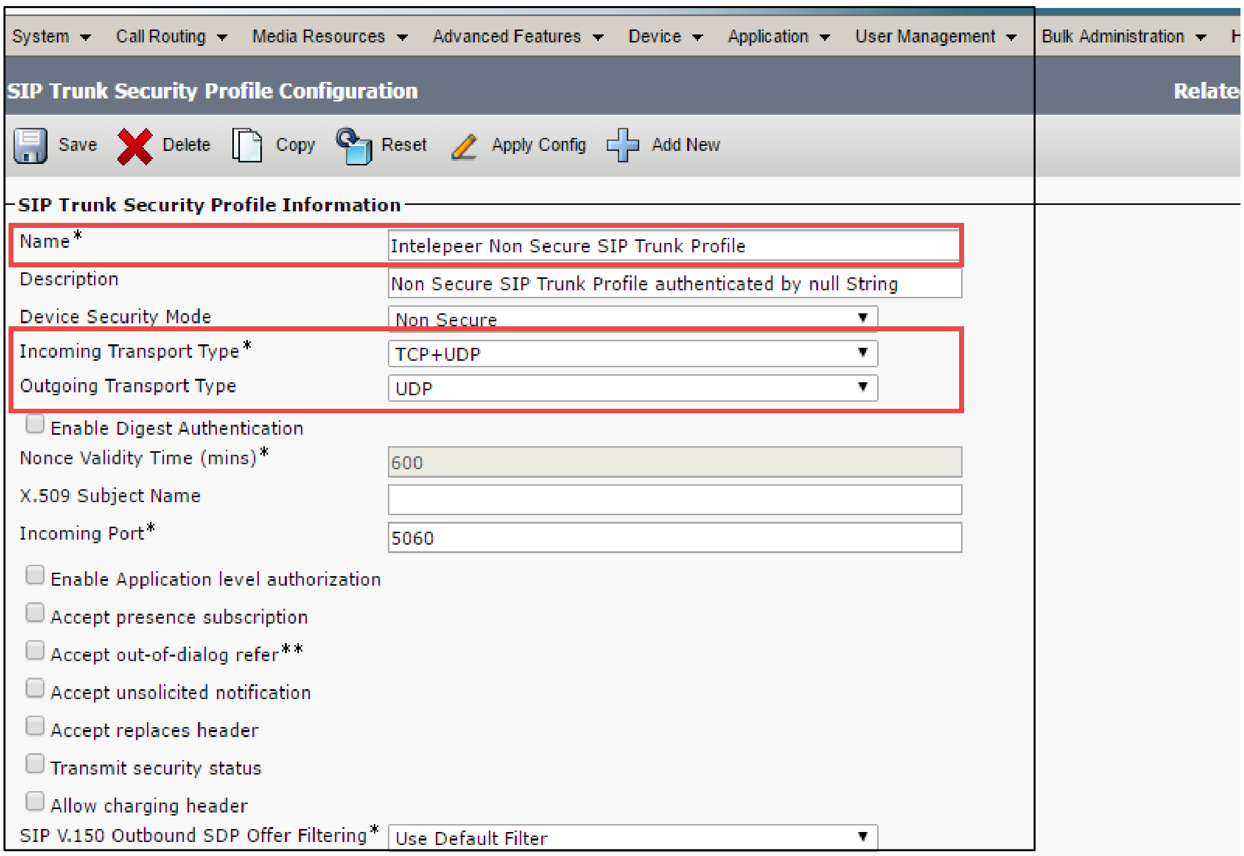


Figure 9: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to IntelePeer 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 Path: Device > Device Settings > SIP Profile

Name*= Intelepeer SIP Profile

Description = Default SIP Profile

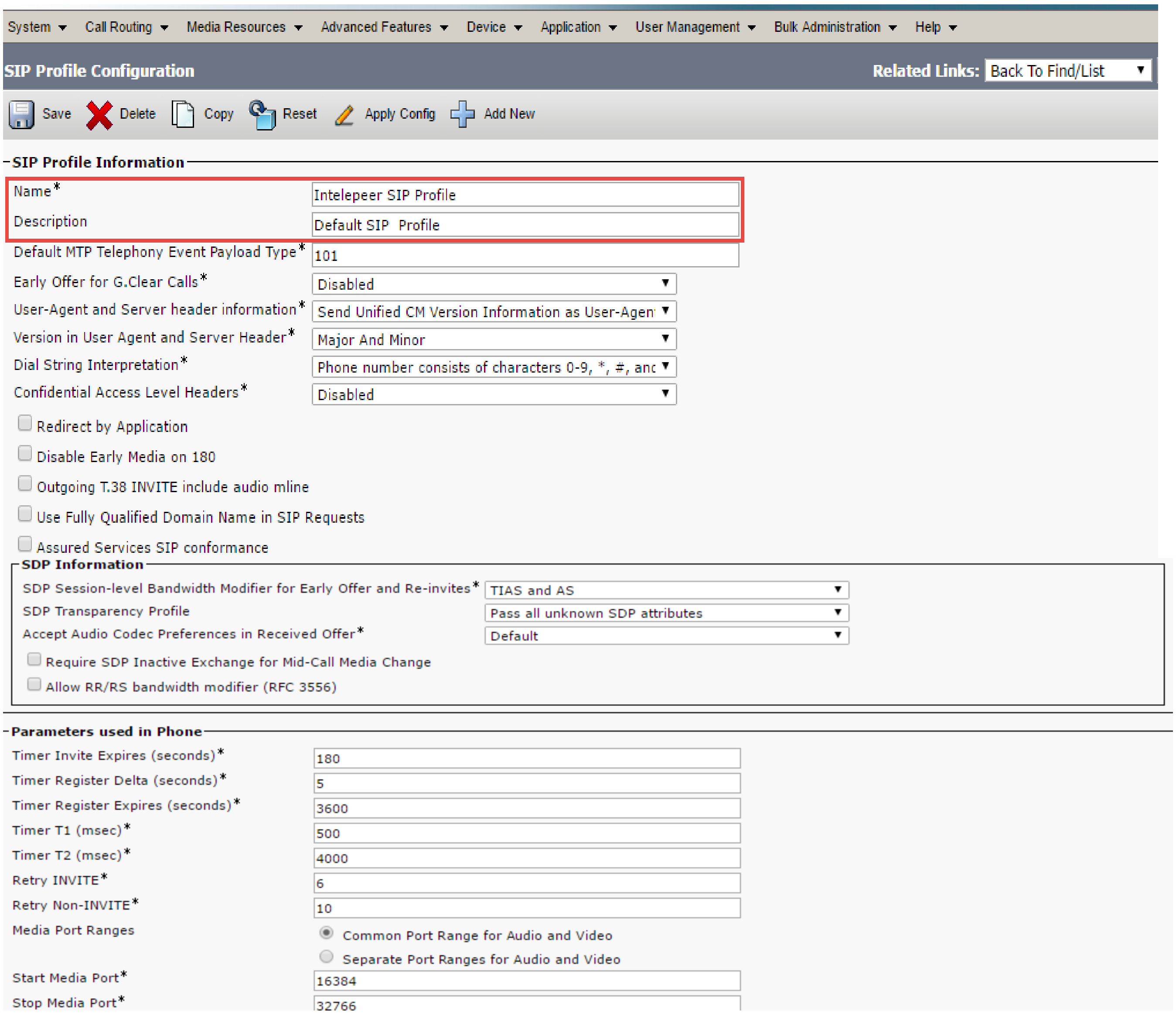


Figure 10: SIP Profile

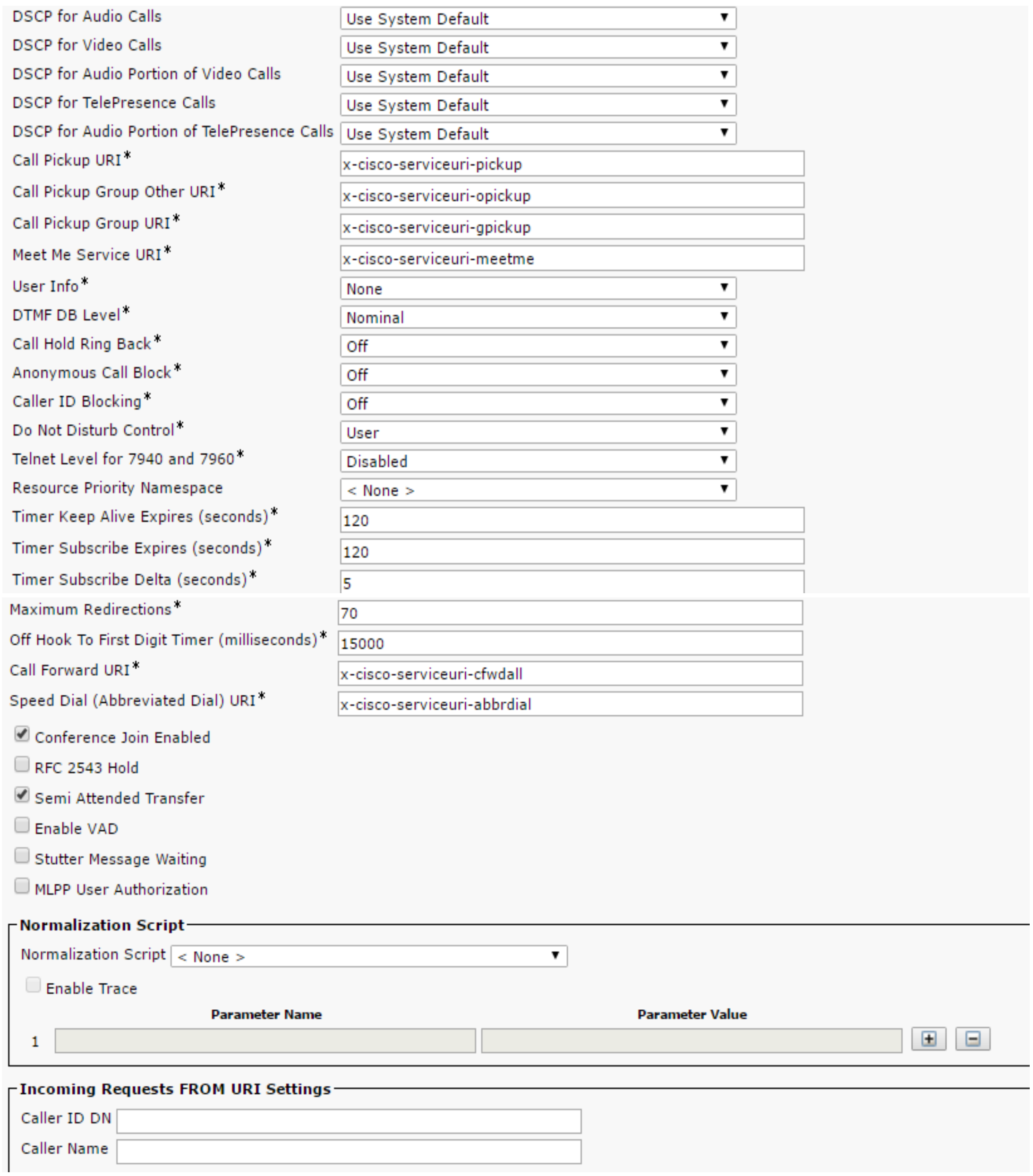


Figure 11: SIP Profile (Cont.)

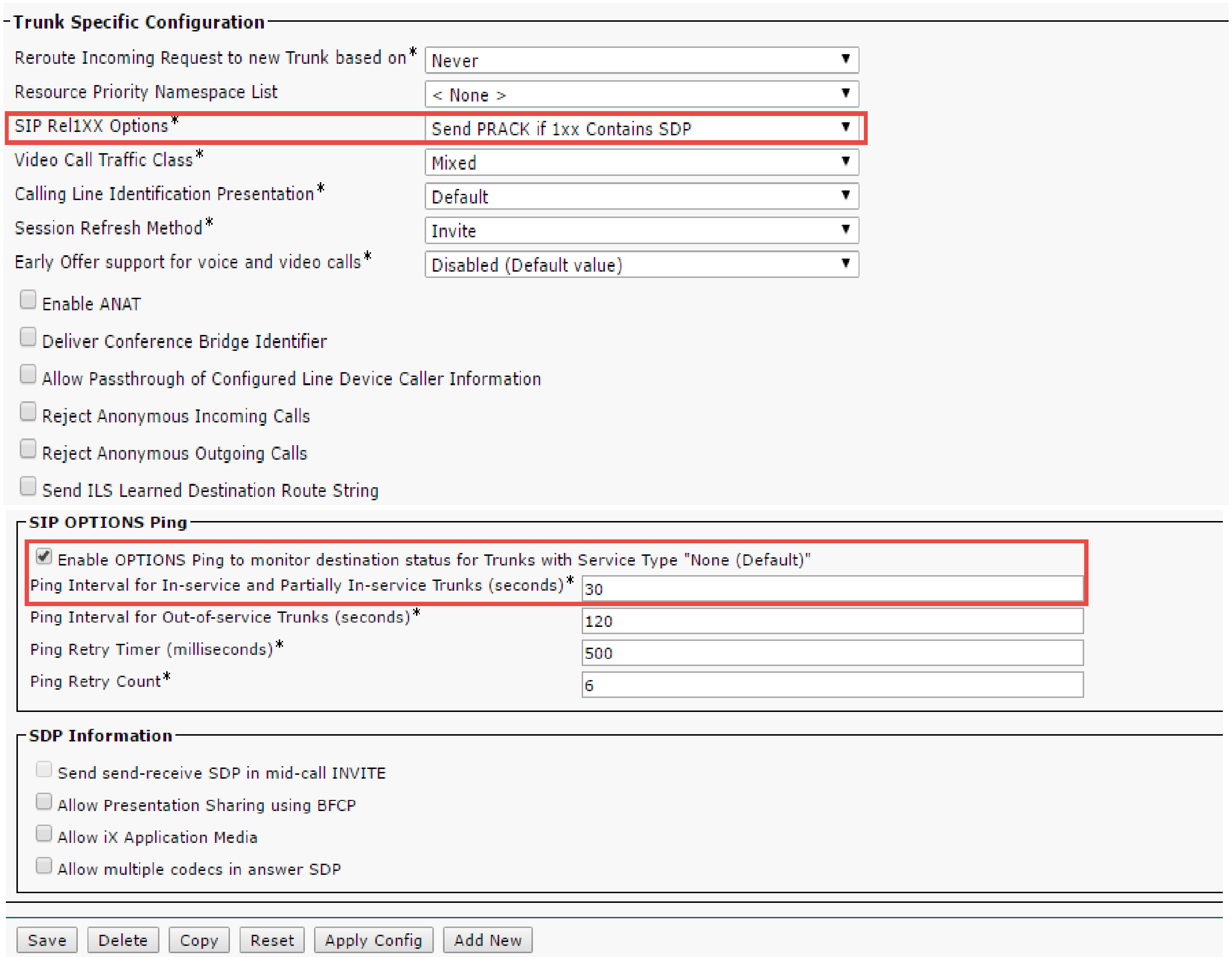


Figure 12: SIP Profile (Cont.)

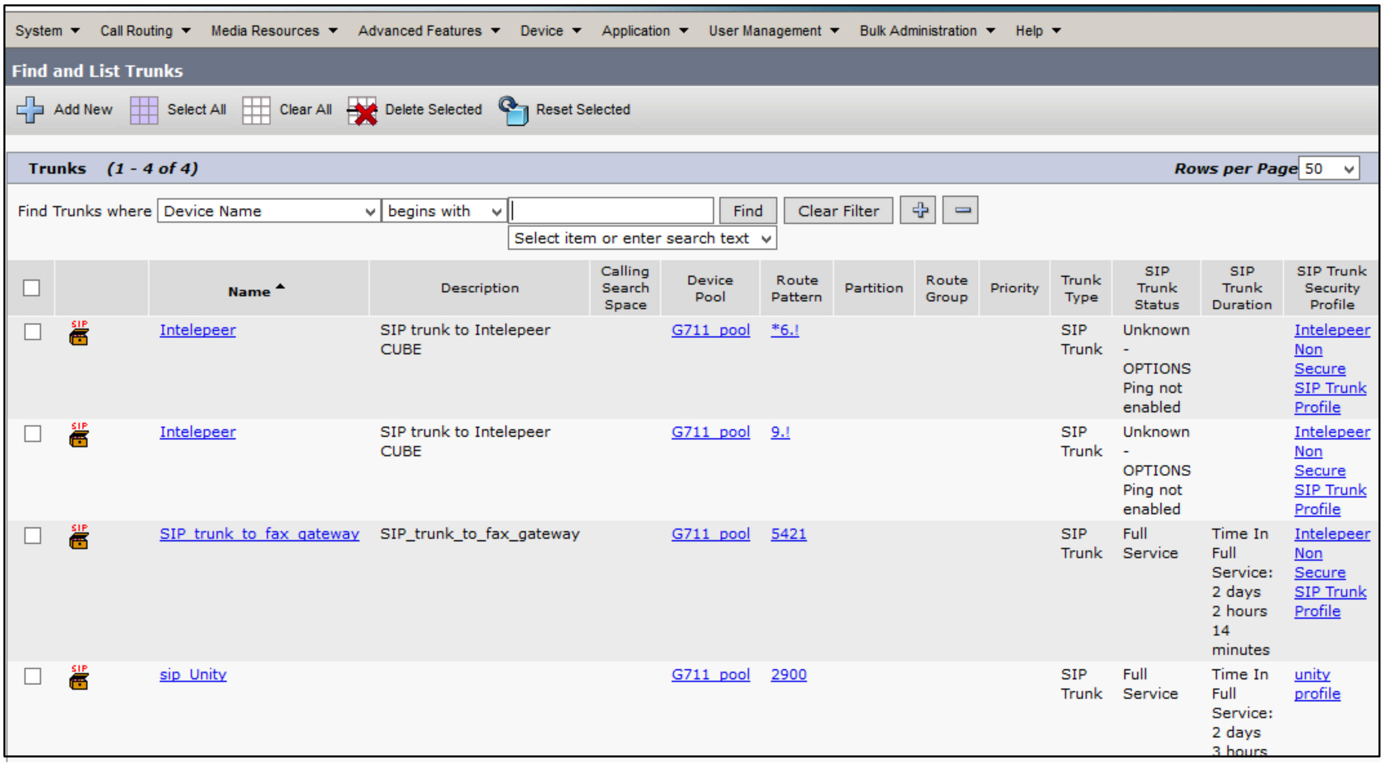
Explanation

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)	30	OPTIONS message parameters- interval time- This is used in this example
Ping Interval for Out-of-service Trunks (seconds)	120	OPTIONS message parameters- interval time

SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation Path: 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
		Intelepeer	SIP trunk to Intelepeer CUBE		G711_pool	*6.!				SIP Trunk	Unknown - OPTIONS Ping not enabled		Intelepeer Non Secure SIP Trunk Profile
		Intelepeer	SIP trunk to Intelepeer CUBE		G711_pool	9.!				SIP Trunk	Unknown - OPTIONS Ping not enabled		Intelepeer Non Secure SIP Trunk Profile
		SIP_trunk_to_fax_gateway	SIP_trunk_to_fax_gateway		G711_pool	5421				SIP Trunk	Full Service	Time In Full Service: 2 days 2 hours 14 minutes	Intelepeer Non Secure SIP Trunk Profile
		sip_Unity			G711_pool	2900				SIP Trunk	Full Service	Time In Full Service: 2 days 3 hours	unity profile

Figure 13: SIP Trunks List

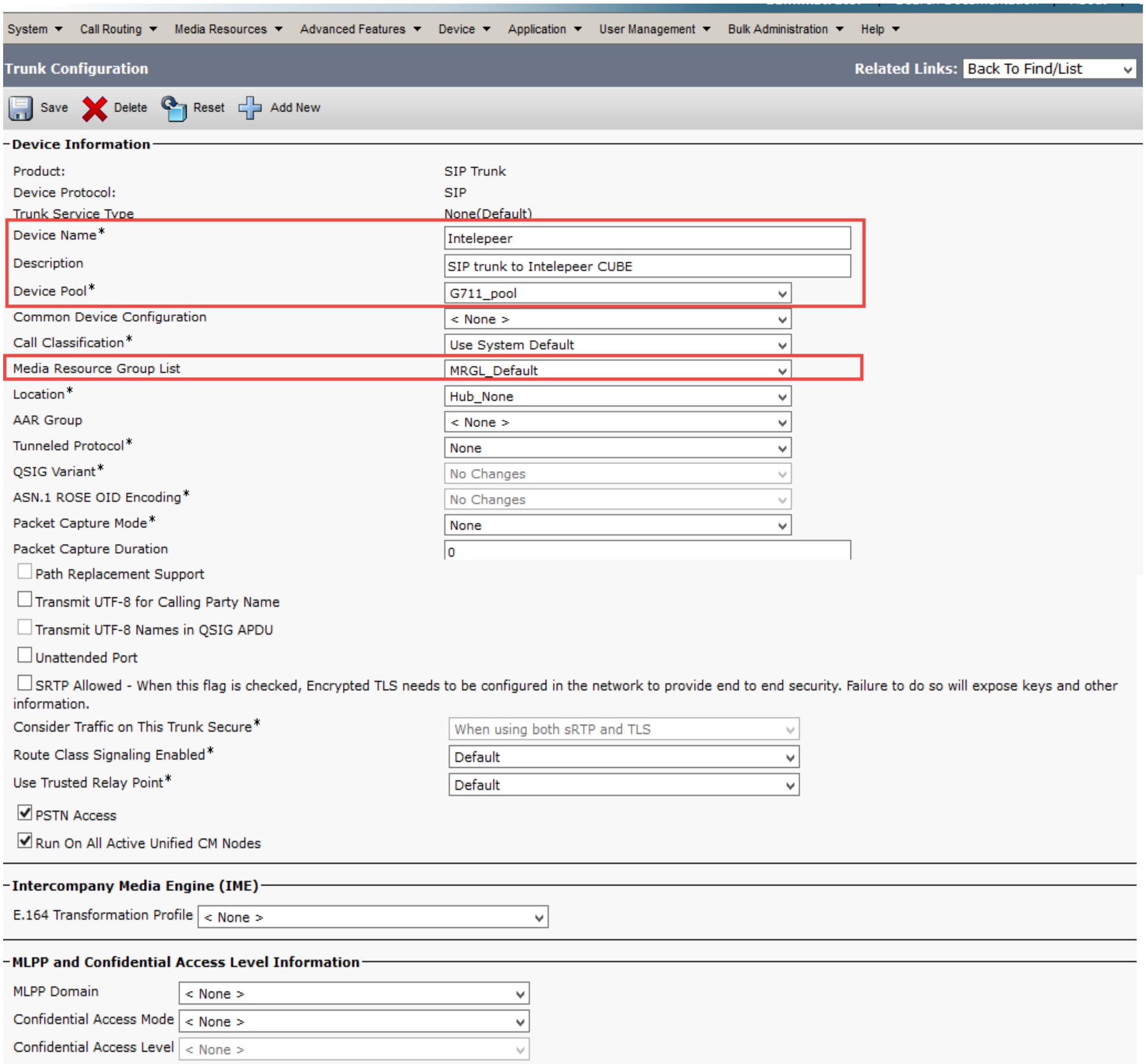
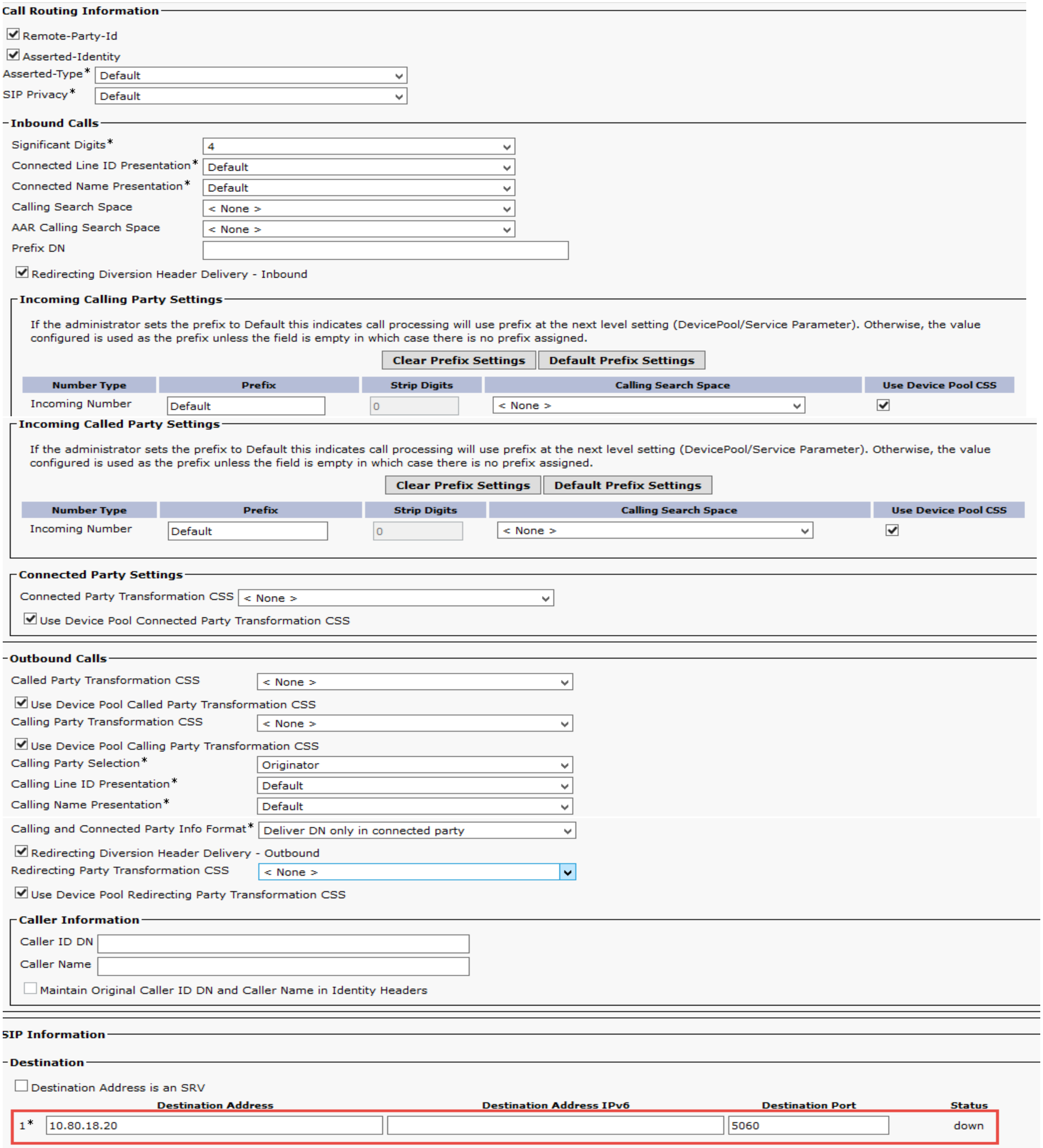


Figure 14: SIP Trunk to Cisco UBE (Cont.)


Call Routing Information

☑ Remote-Party-Id

☑ Asserted-Identity

Asserted-Type* Default

SIP Privacy* 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

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.80.18.20		5060	down

Figure 15: SIP Trunk to Cisco UBE (Cont.)

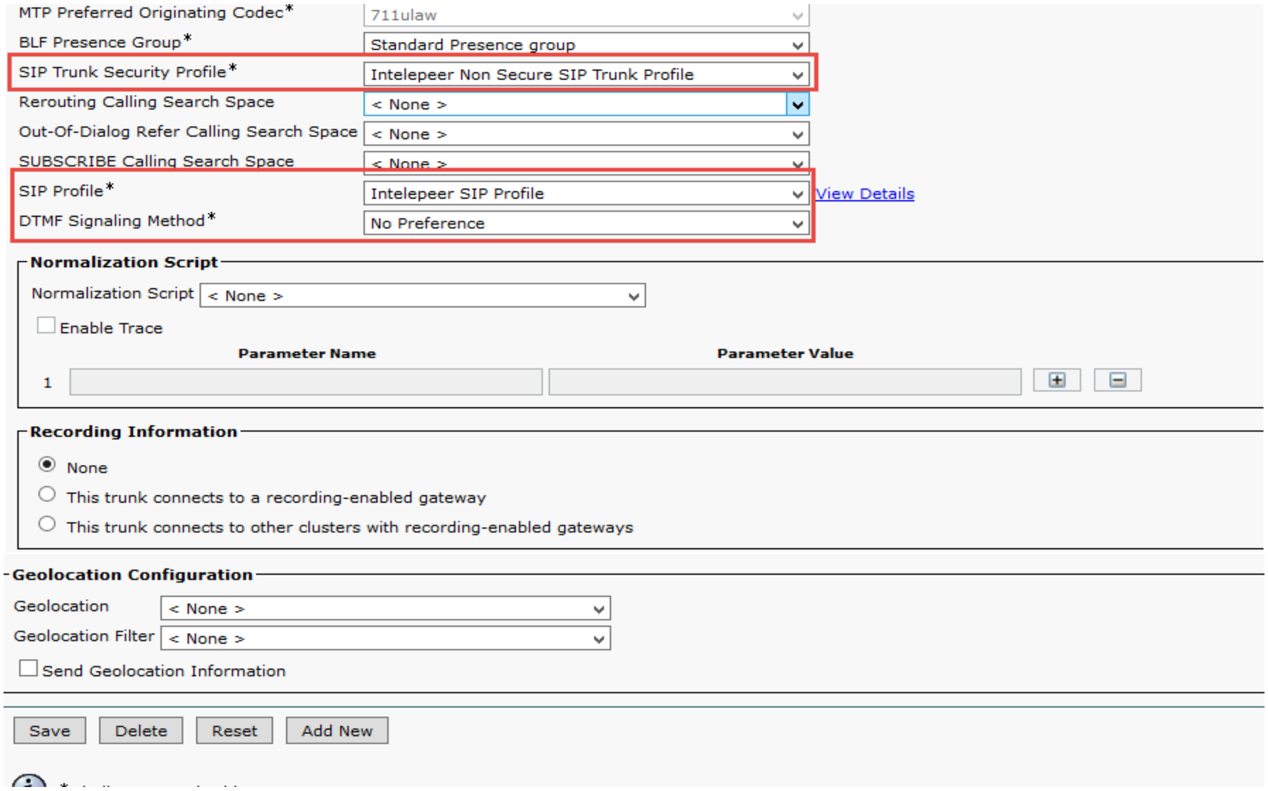


Figure 16: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	Intelepeer	Name for the trunk
Device Pool	G711_pool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_Default	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.18.20	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	Intelepeer Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Intelepeer Standard SIP Profile	SIP Profile configured earlier

Dial Plan

Route Pattern Configuration

Navigation Path: Call Routing > Route/Hunt > Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial “9”.10 digits number to access PSTN via Cisco UBE
    - “9” is removed before sending to Cisco UBE
- For FAX call, Access Code “9”+ 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 IntelePeer network
- Incoming fax call to xxxx will be sent to the Cisco Fax gateway

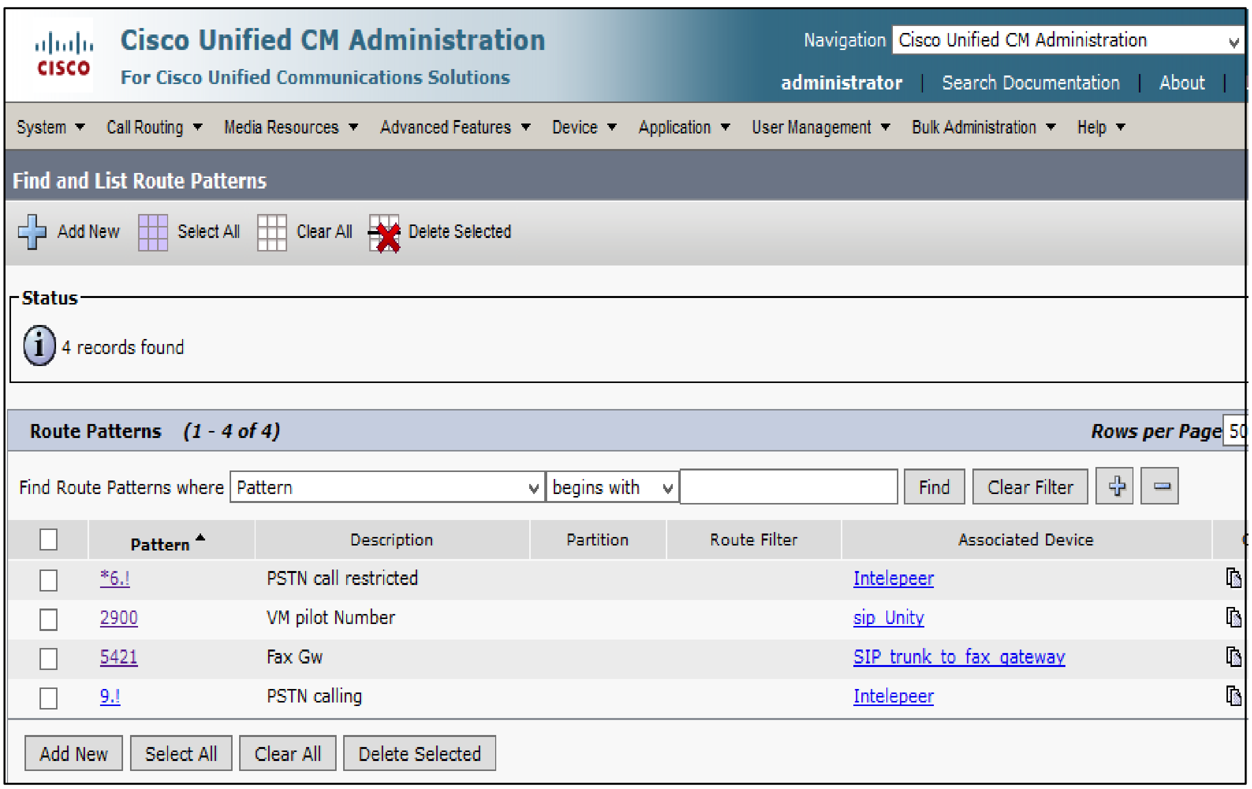


Figure 17: Route Patterns List

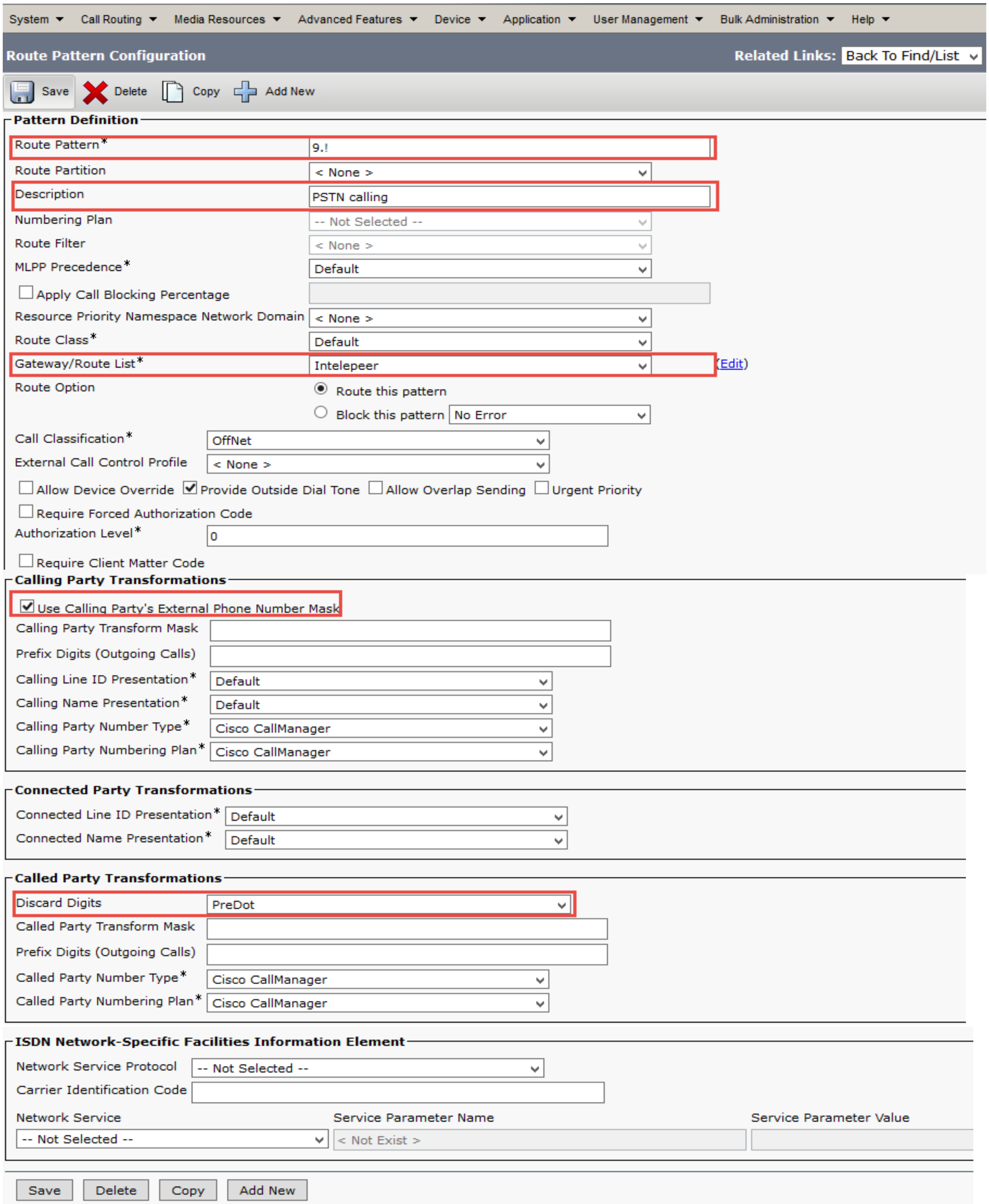


Figure 16: Route Pattern for Voice

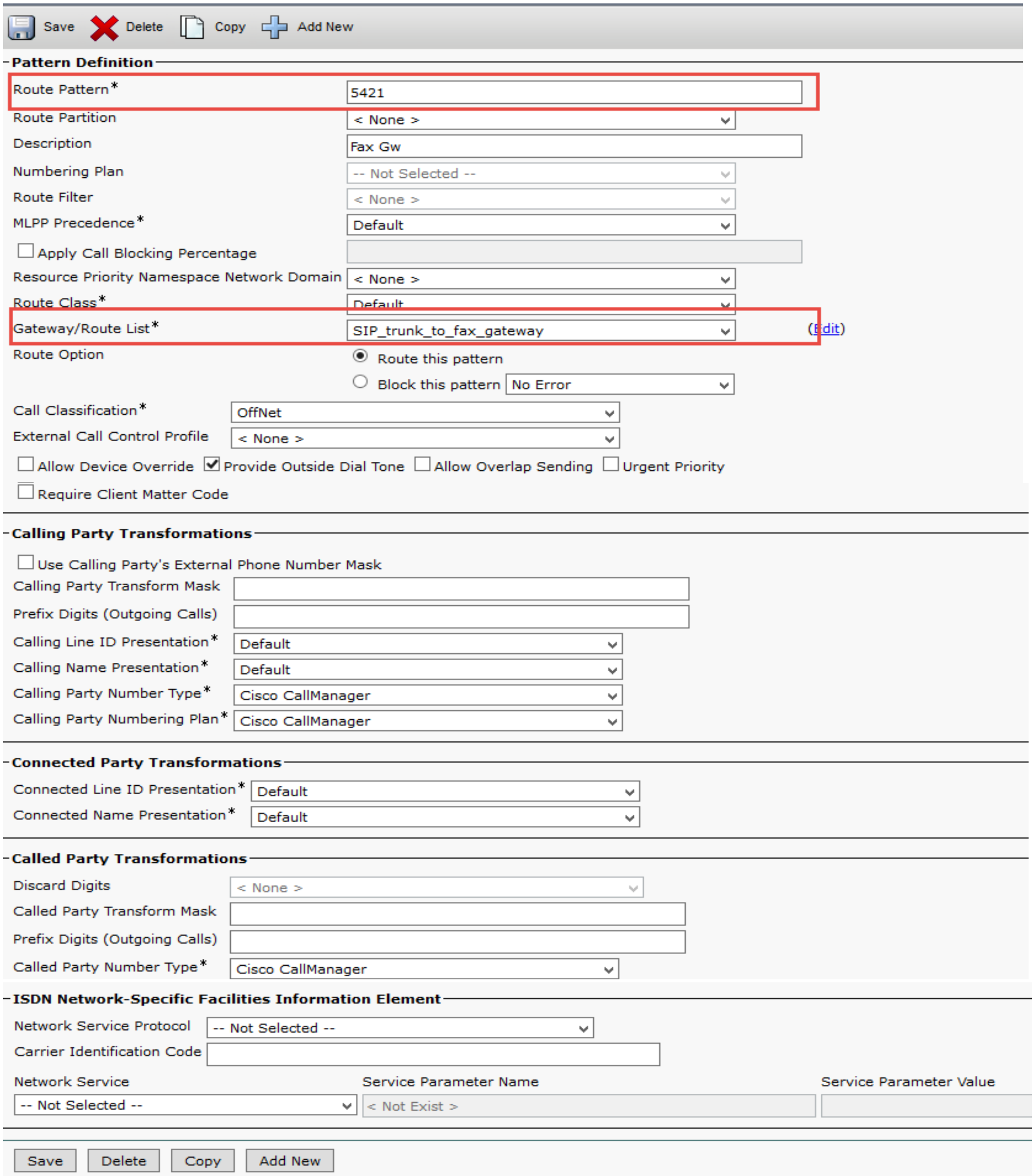


Figure 19: Route Pattern for Voice

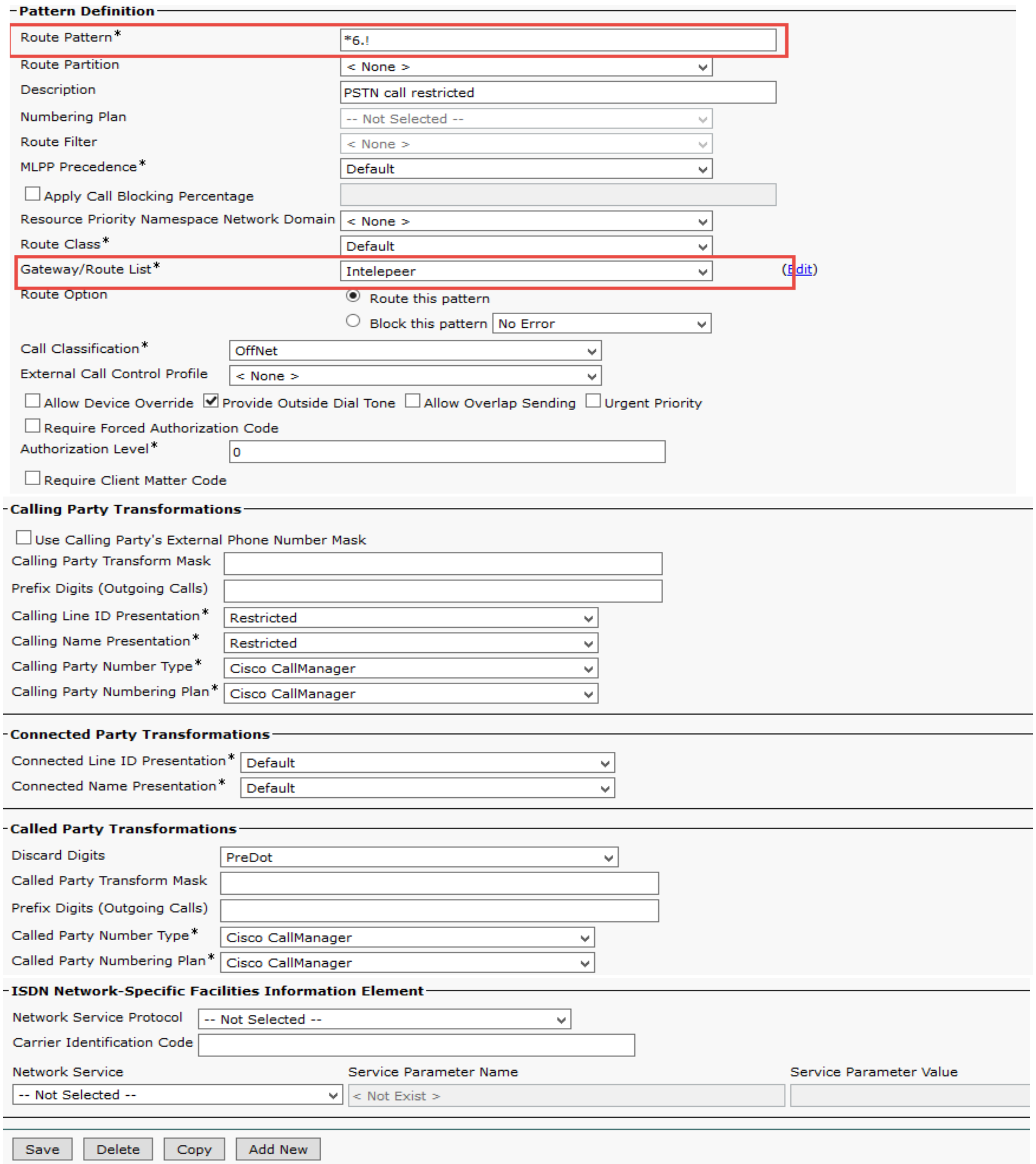


Figure 21: Route Pattern for Voice – Cont.

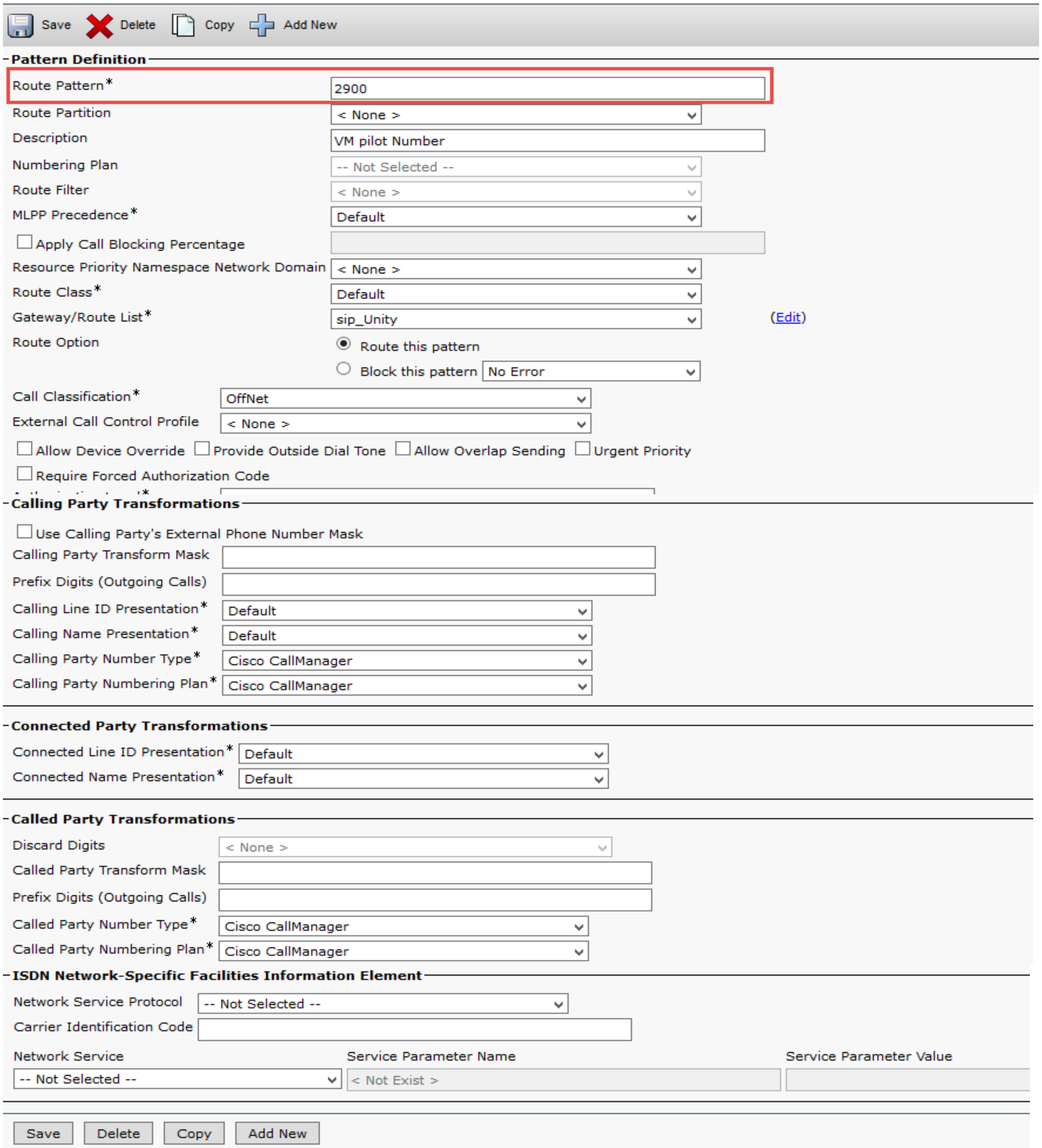


Figure 22: Route Pattern for Voice

Explanation

Setting	Value	Description
Route Pattern	9.! For Voice call, 5421 for fax call,*6.! For restricted calls,2900 for Voice mail pilot number	Specify appropriate Route Pattern
Gateway/Route List	Routed the calls to the appropriate trunk and gateways	SIP Trunk name configured earlier
Discard Digits	PreDot for Route Pattern 9.!,*6.! .	Specifies how to modify digit before they are sending to IntelePeer network

Acronyms

Acronym	Definitions
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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Appendix A: Test Results

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