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-XE 3.17 – 15.6(1)S] using SIP

May 20, 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-XE 3.17 – 15.6(1)S can be used. The Cisco Unified Border Element 11.5.0 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-XE 3.17 – 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

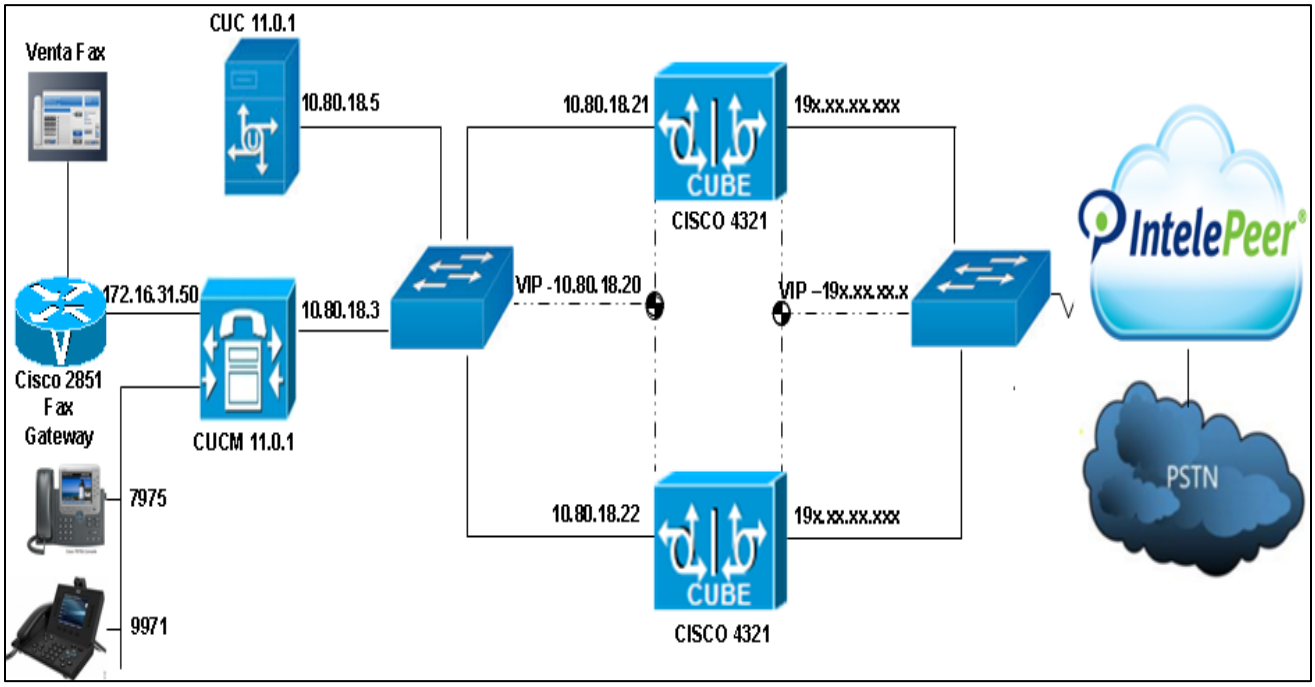

Figure 1: Network Topology

- 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-XE 3.17 – 15.6(1)S for ISR 4321/K9 Cisco Unified Border Element (CUBE 11.5.0)
- 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
- Privacy is not supported on Loopback to PBX. Anonymous calling out to network and back into same PBX.
    - Inbound to PBX Privacy works without issue
    - Outbound from PBX Privacy works without issue
- Intermittent no way speech path observed for an inbound calls. This issue is not indicative of CUBE/CUCM and Taqua standard behavior but is a Layer 3 issue. The intermittent audio issue is due to the nature of the lab used for testing by IntelePeer. IntelePeer has CUBE/CUCM customers running this deployment load on Taqua in production without audio issues

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.

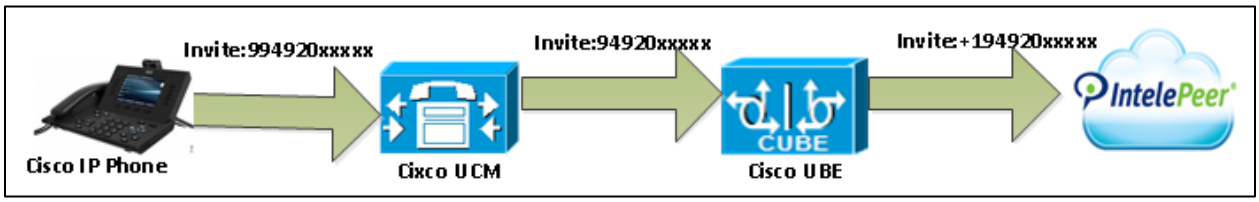

Figure 2: Outbound Voice Call

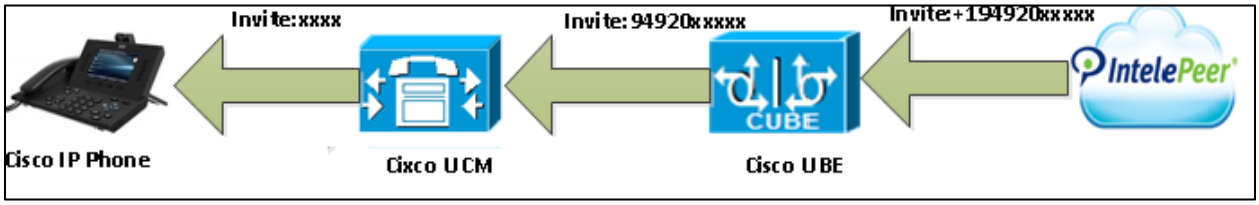

Figure 3: Inbound Voice Call

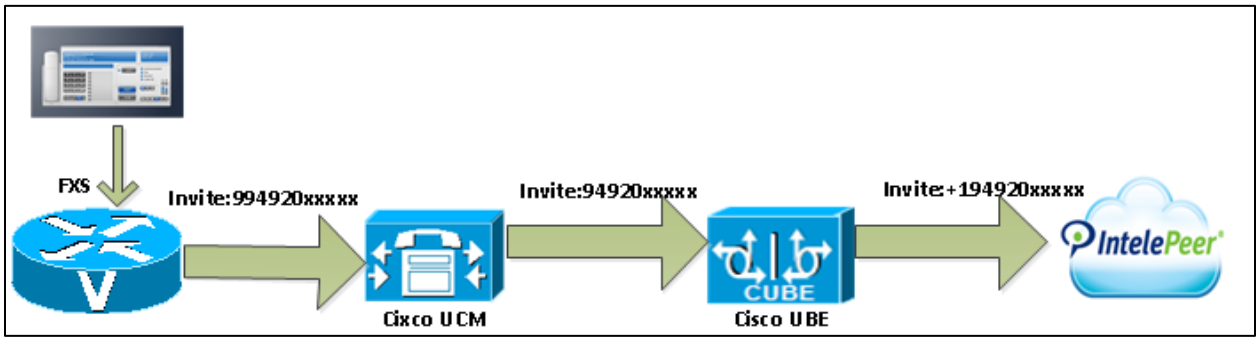

Figure 4: Outbound Fax Call

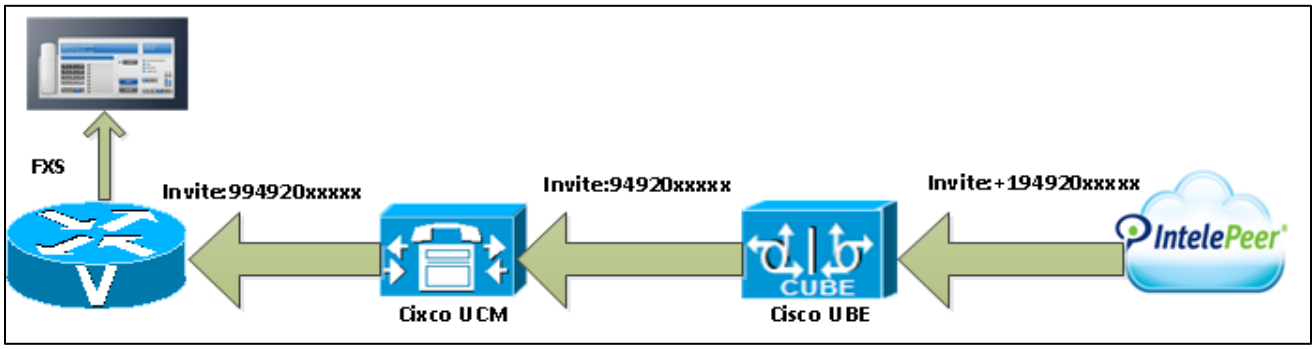

Figure 5: Inbound Fax Call

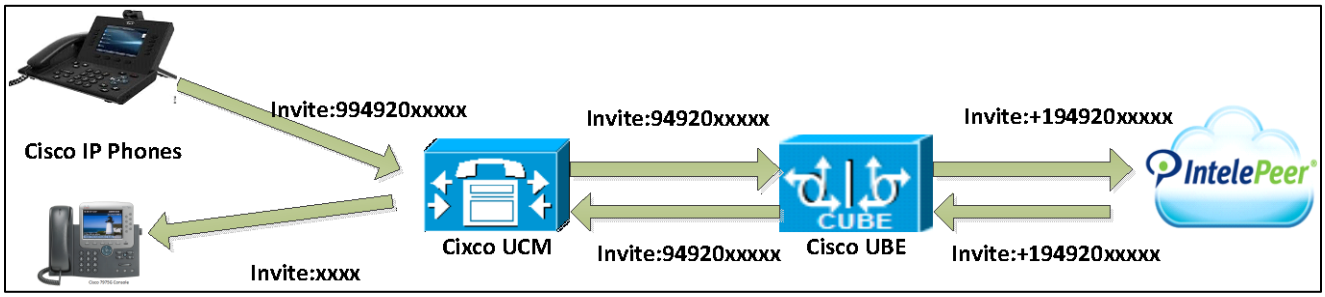

Figure 6: PBX to PBX via IntelePeer Call

Configuration Example

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

Active Cisco UBE

```
IN_CUBE1#sh running
!
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
!
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 FDO19280MSQ
!
spanning-tree extend system-id
```

```
!
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

```
!
IN_CUBE2#sh run
!
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
!
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 FDO19290MQ9
!
spanning-tree extend system-id
!
```

```
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
!
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
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 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 ipv4:68.68.123.174:5060
 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#
```

Cisco UCM Version

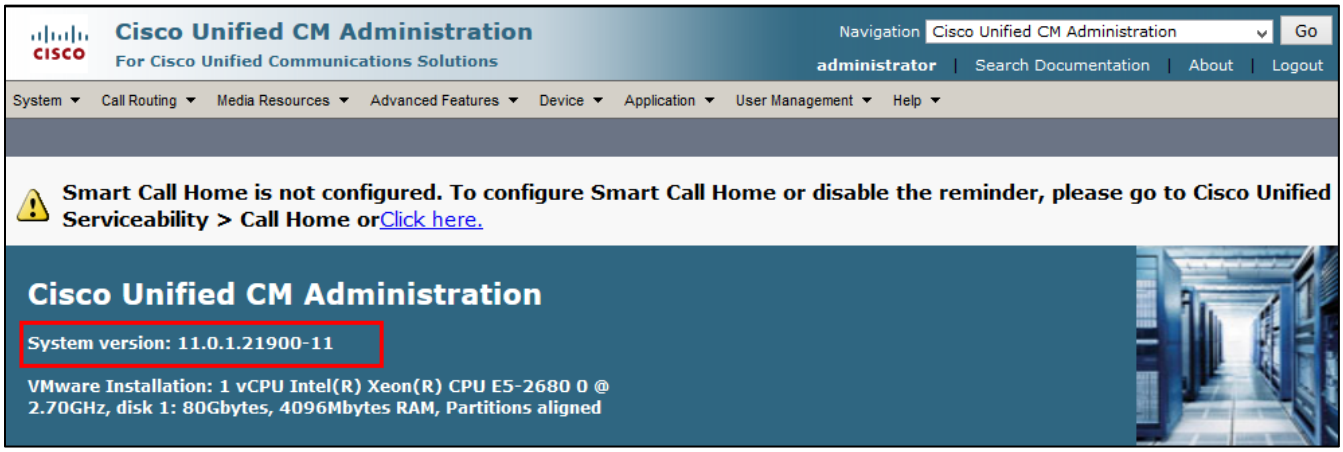


Figure 7: Cisco UCM Version

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

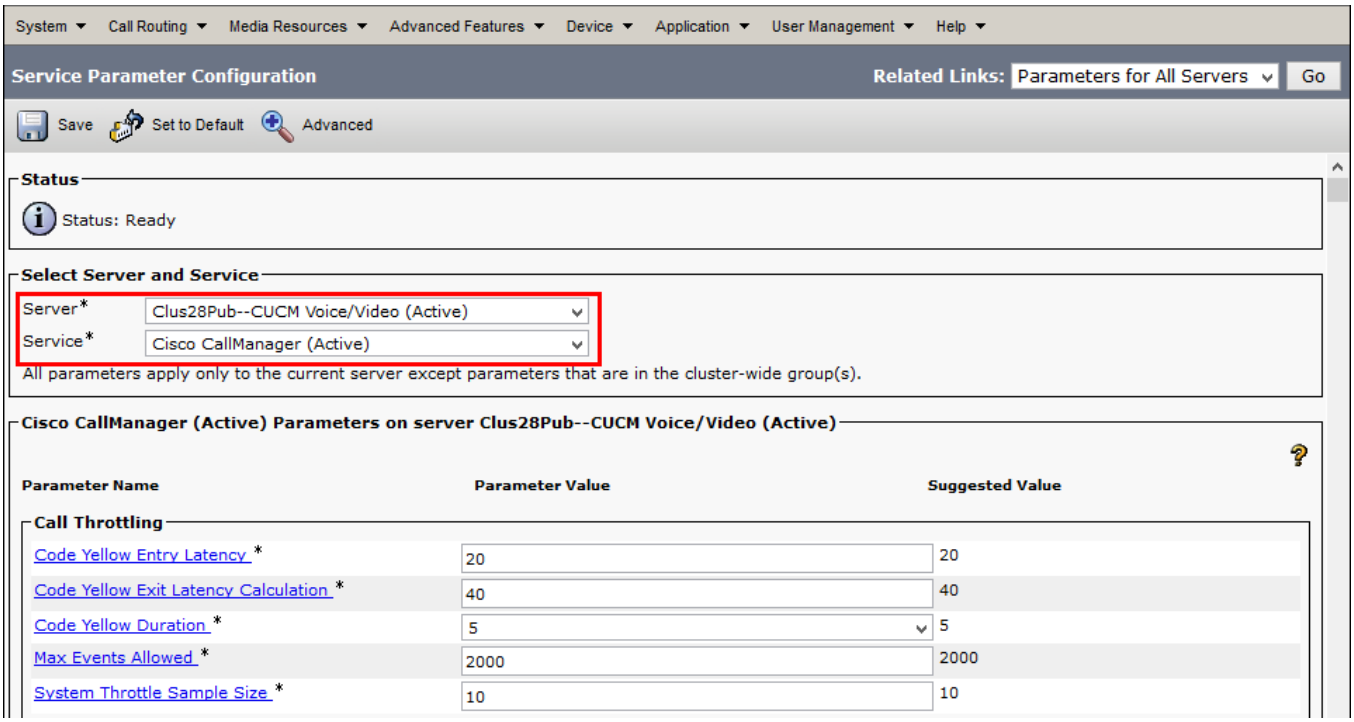


Figure 8: Service Parameters

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

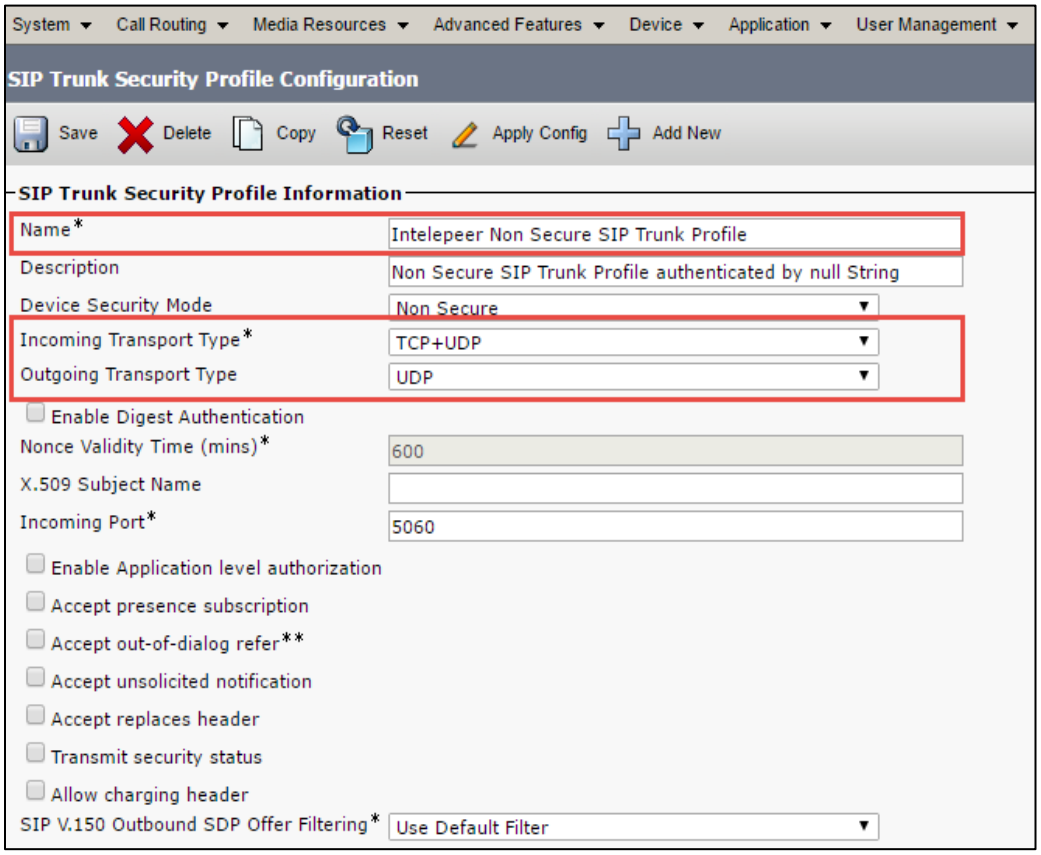


Figure 9: SIP Trunk Security Profile

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

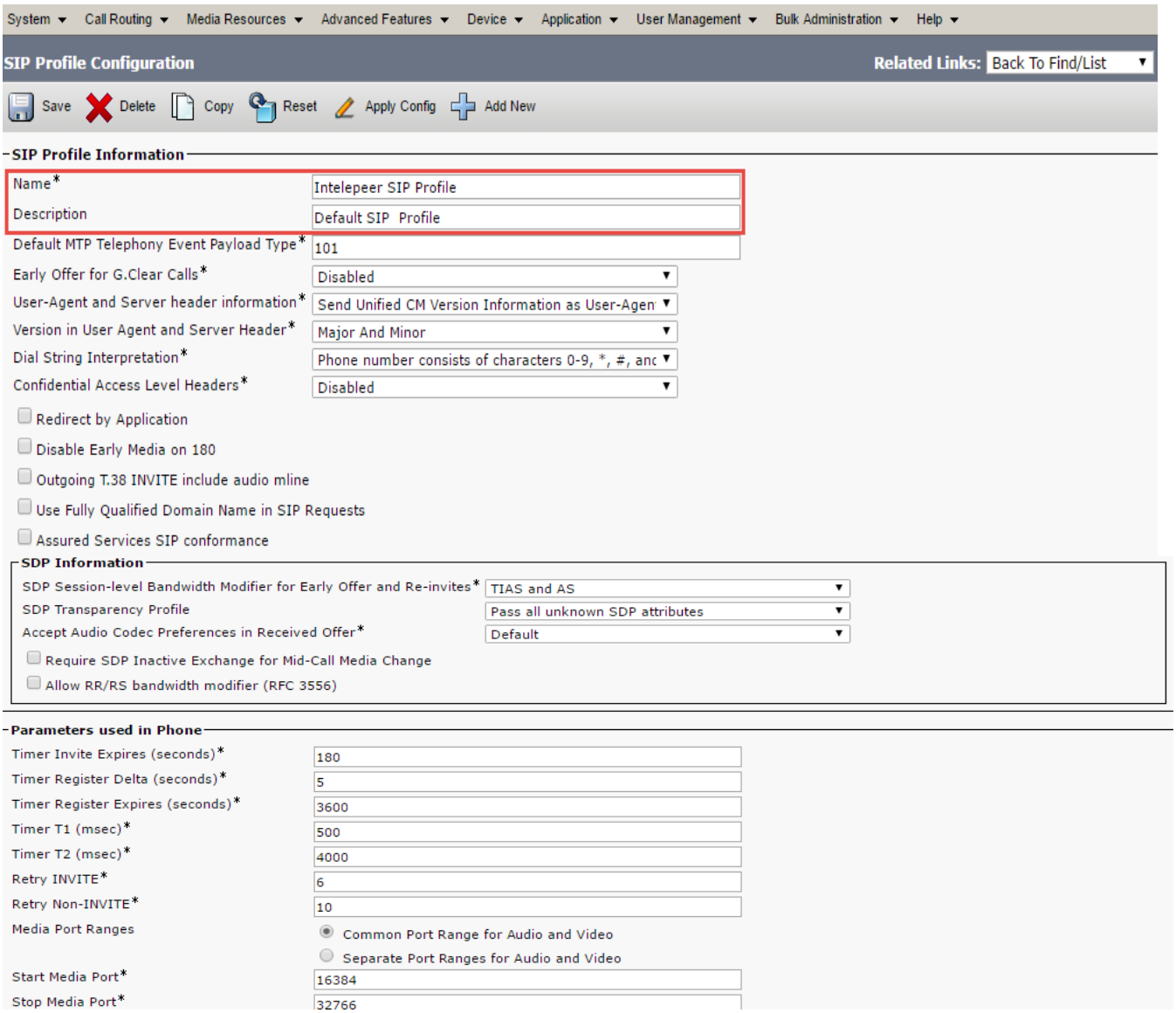


Figure 10: SIP Profile

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 11: SIP Profile (Cont.)

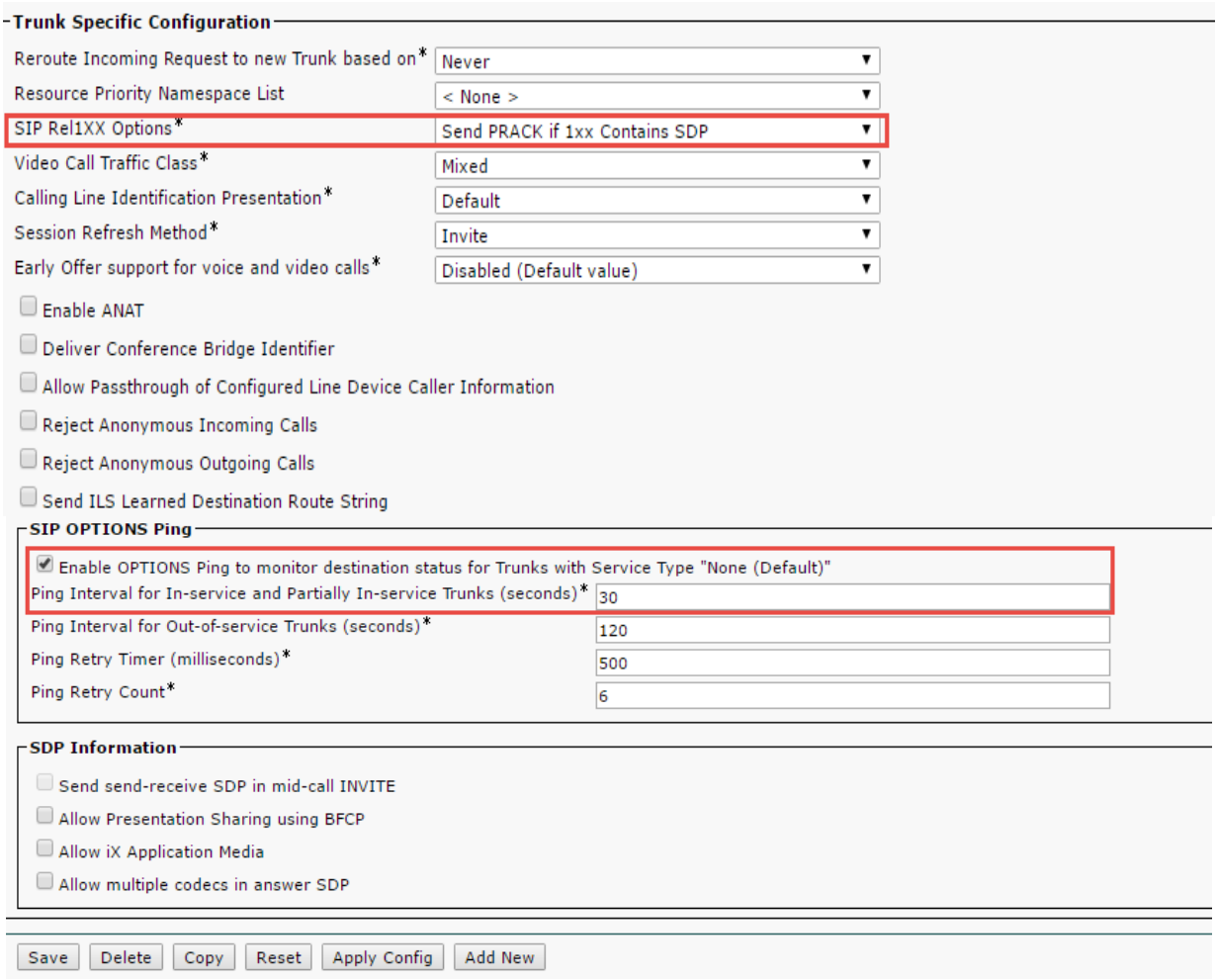


Figure 12: SIP Profile (Cont.)

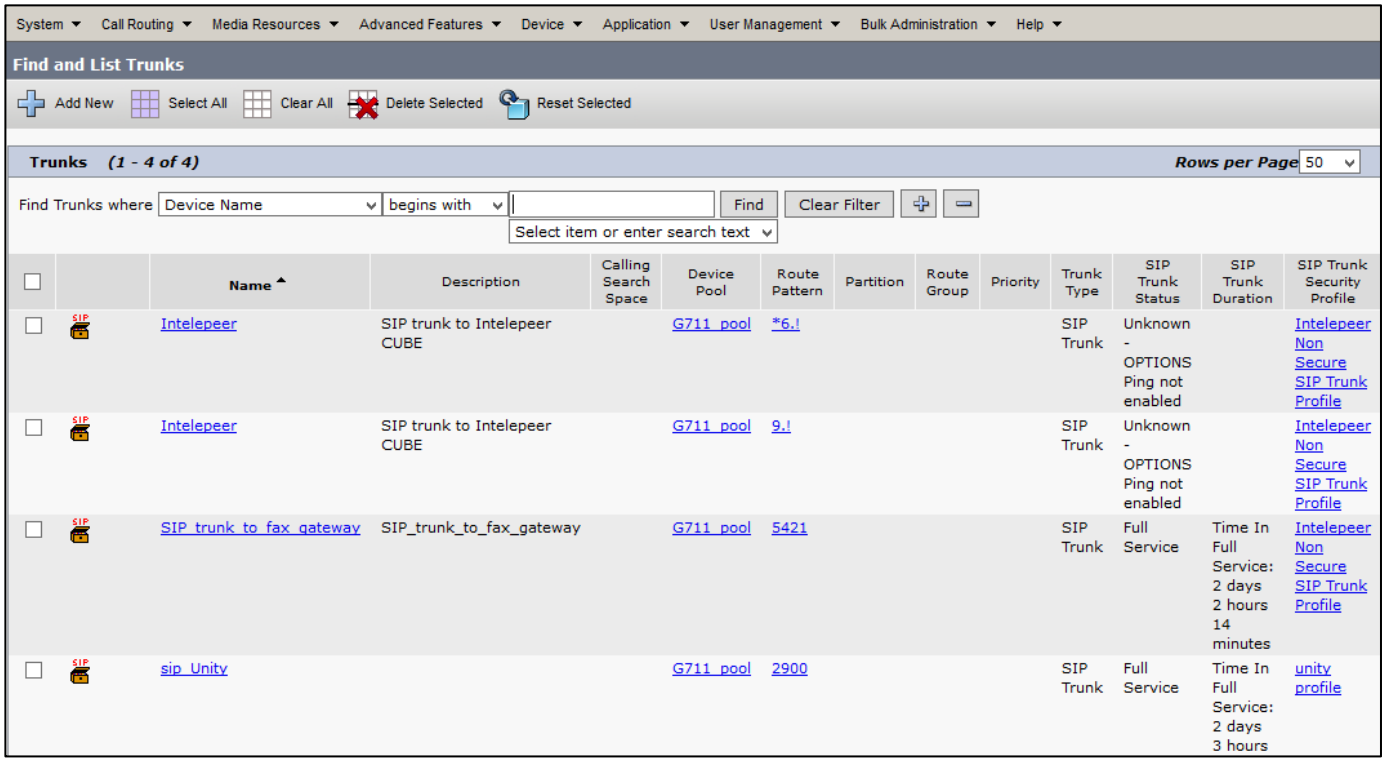
Explanation

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

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Trunk Configuration — Related Links: Back To Find/List

Save | Delete | Reset | Add New

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Intelepeer
Description	SIP trunk to Intelepeer CUBE
Device Pool*	G711_pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_Default
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

- [ ] Path Replacement Support
- [ ] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU
- [ ] Unattended Port
- [ ] SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default

- [x] PSTN Access
- [x] Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile: < None >

MLPP and Confidential Access Level Information

MLPP Domain	< None >
Confidential Access Mode	< None >
Confidential Access Level	< None >

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

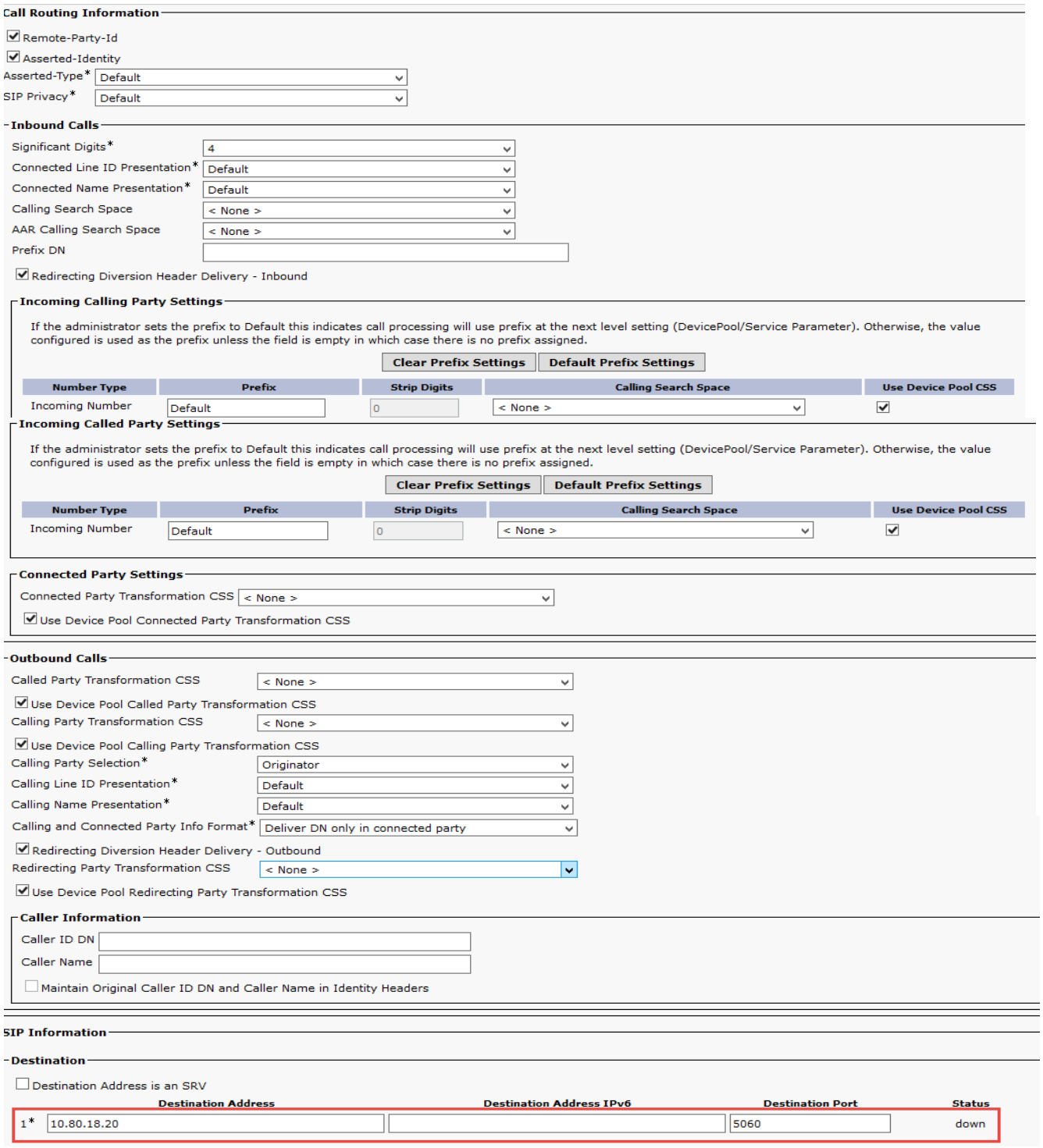


Figure 15: SIP Trunk to Cisco UBE (Cont.)

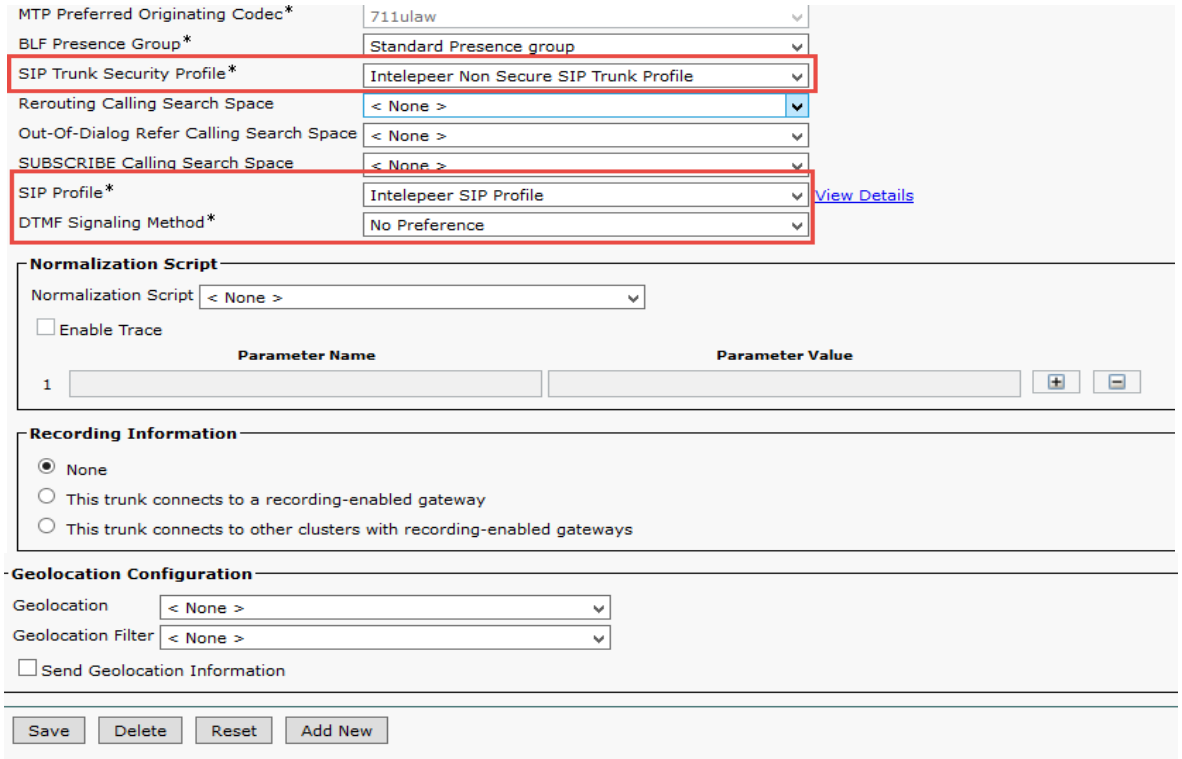


Figure 16: SIP Trunk to Cisco UBE (Cont.)

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

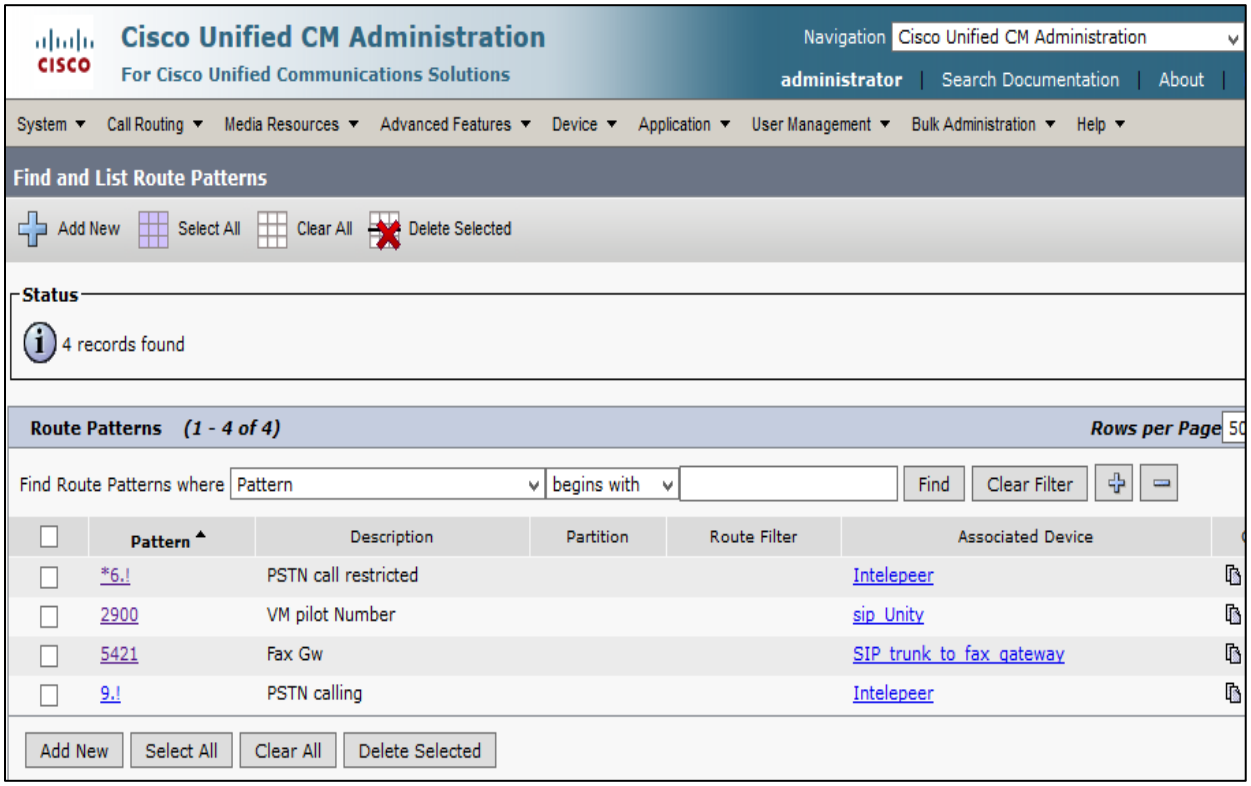


Figure 17: Route Patterns List

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Route Pattern Configuration — Related Links: Back To Find/List

Save | Delete | Copy | Add New

Pattern Definition

Route Pattern*	9.!
Route Partition	< None >
Description	PSTN calling
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	Intelepeer (Edit)
Route Option	Route this pattern / Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >

Allow Device Override | Provide Outside Dial Tone | Allow Overlap Sending | Urgent Priority

Require Forced Authorization Code

Authorization Level* 0

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Called Party Transformations

Discard Digits	PreDot
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol	-- Not Selected --
Carrier Identification Code	

Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Save | Delete | Copy | Add New

Figure 16: Route Pattern for Voice

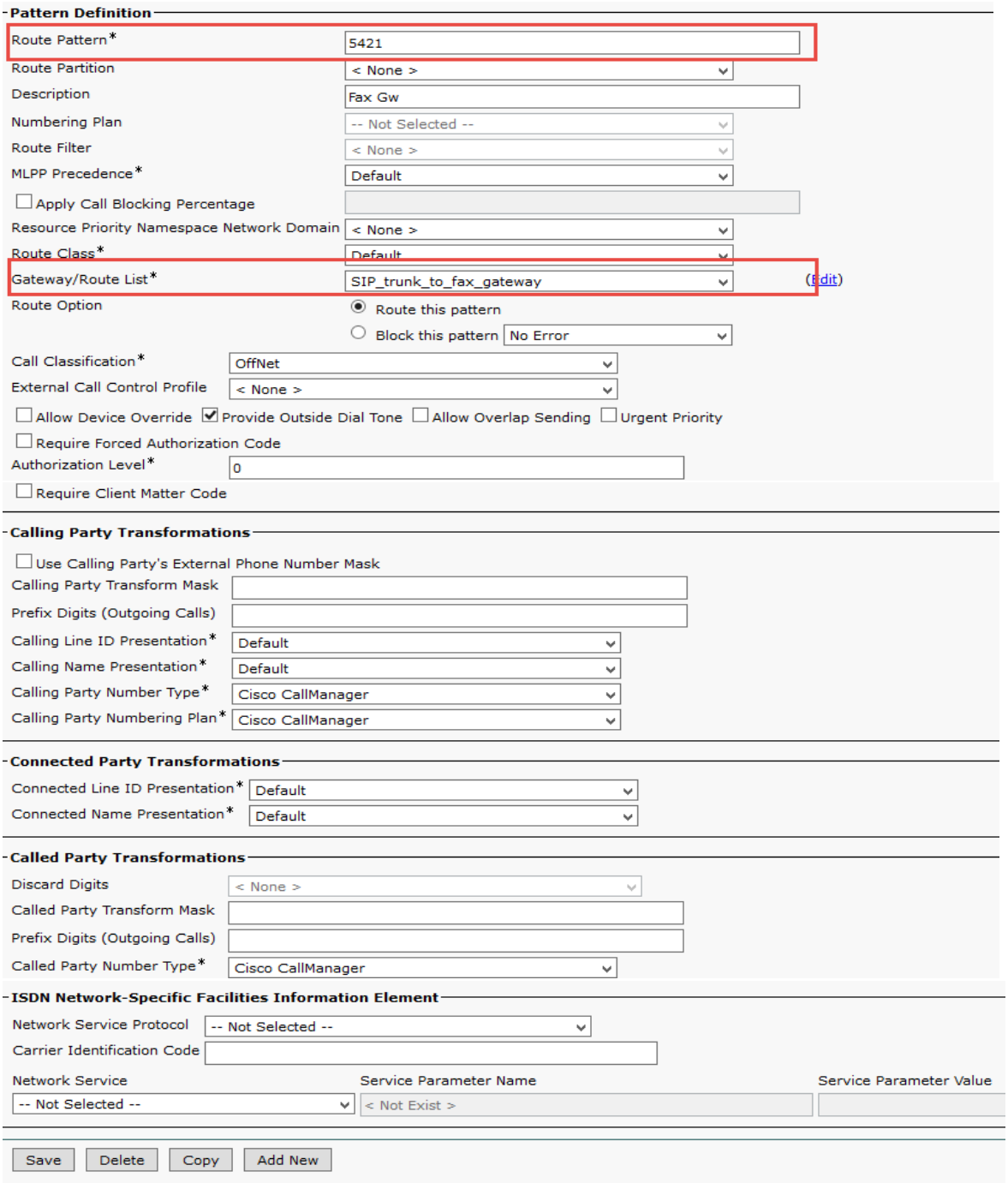


Figure 19: Route Pattern for Voice

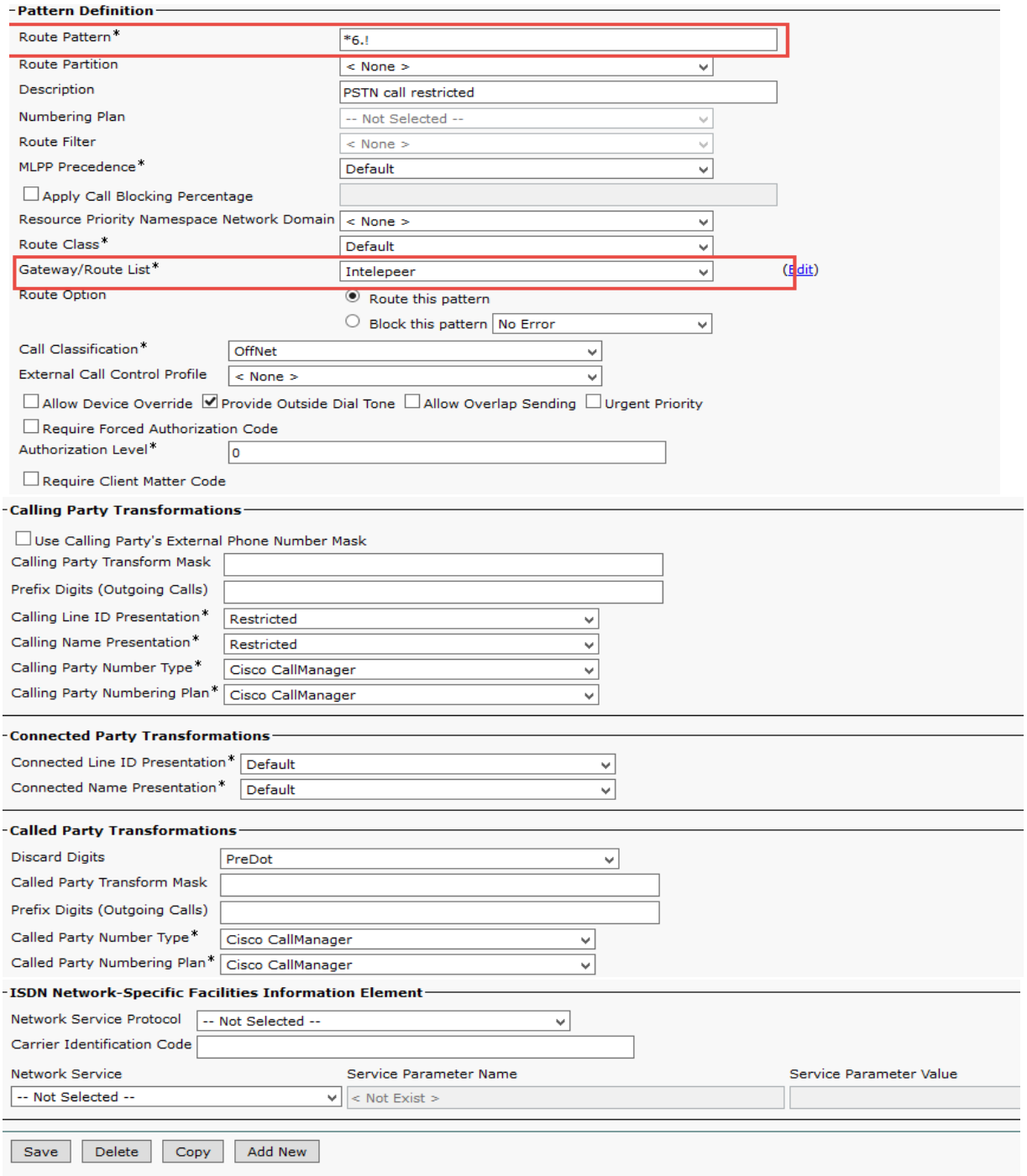


Figure 21: Route Pattern for Voice – Cont.

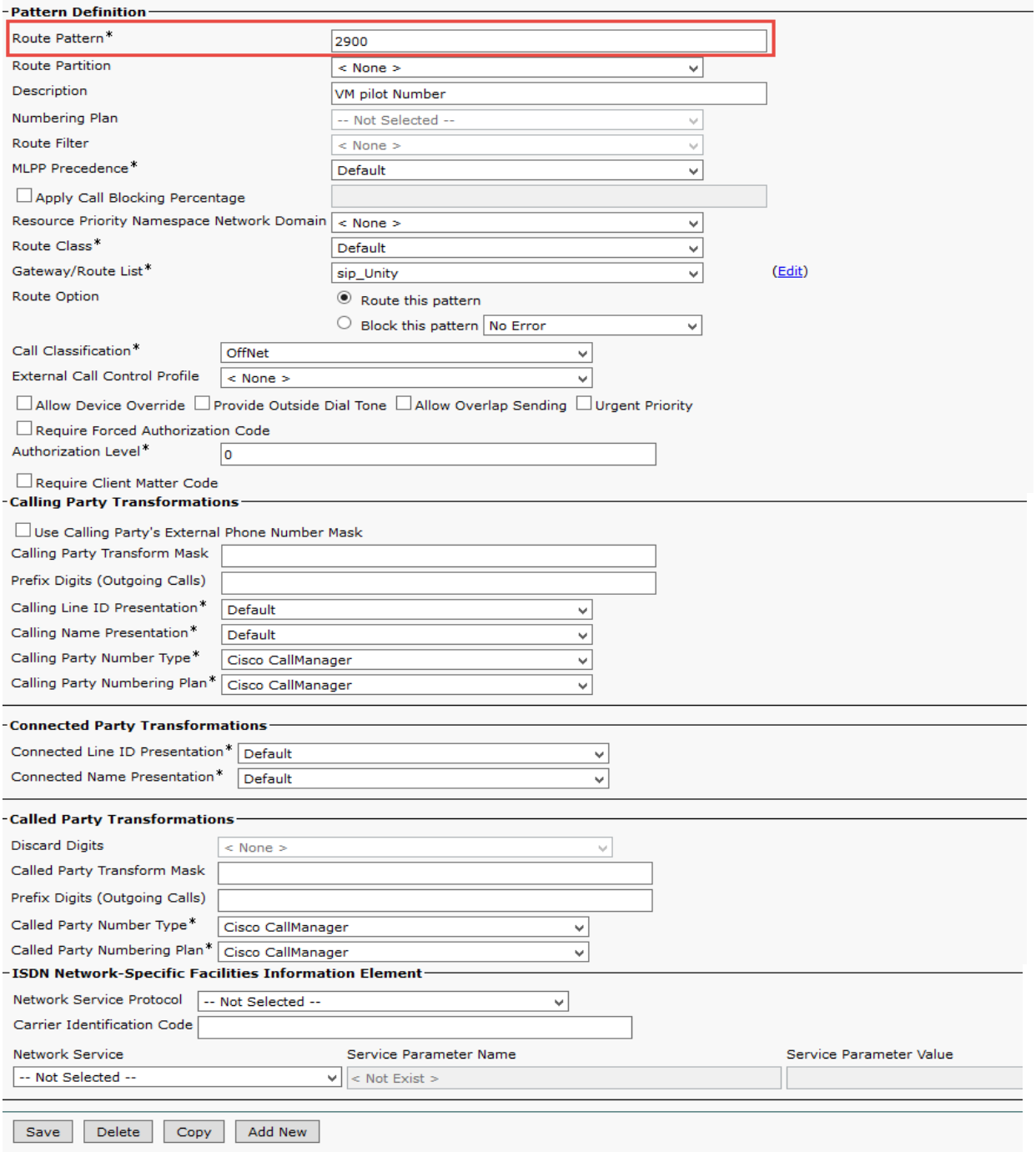


Figure 22: Route Pattern for Voice

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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