Application Note

IntelePeer SIP Trunking:

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

July 26, 2017

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Introduction

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

A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and Spectrum network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS 16.5.1b can be used. The Cisco Unified Border Element 16.5.1b provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to IntelePeer 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) 11.5.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS - 16.5.1b] for connectivity to IntelePeer SIP Trunking service available in the former IntelePeer Business service area[1] . The deployment model covered in this application note is CPE (Cisco UCM 11.5.1) 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

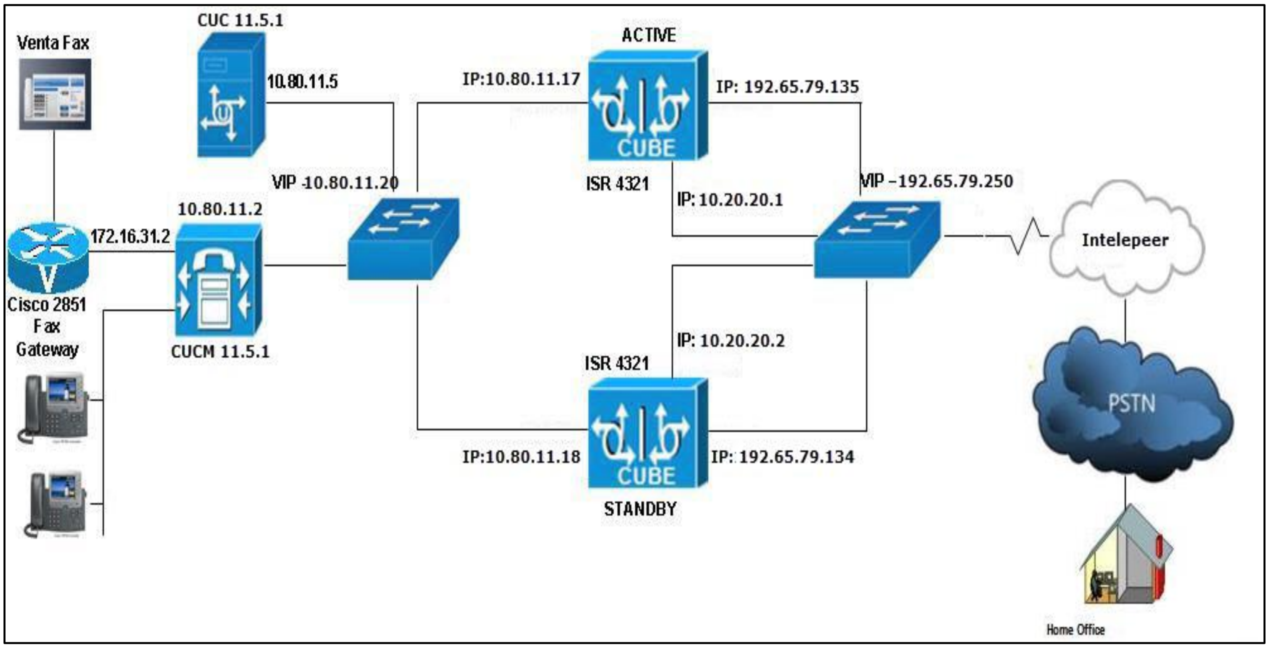

Figure 1: Network Topology

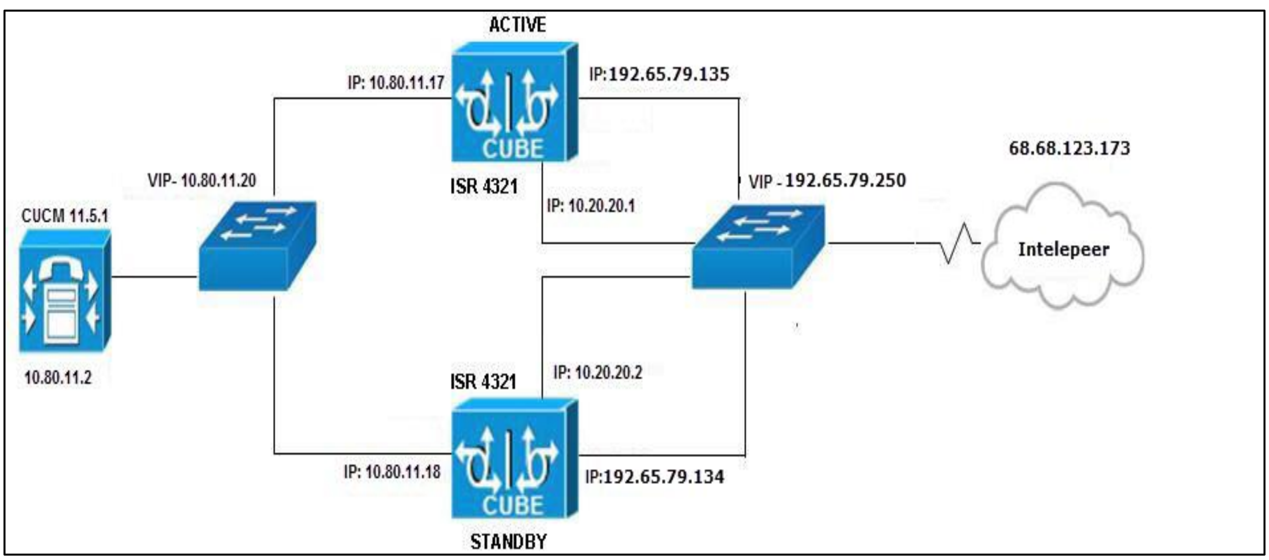

Figure 2: Cisco UBE High Availability

System Components

Hardware Requirements

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

Software Requirements

- Cisco Unified Communications Manager 11.5.1
- Cisco Unity Connection 11.5.1
- IOS-XE 16.05.01b for ISR 4431/K9 Cisco Unified Border Element
- Cisco IOS XE Software, Version 03.17.01.S
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway

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 and T38)

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer
- 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.

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 192.65.79.135 255.255.255.128
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.79.250 exclusive
!
interface GigabitEthernet0/0/2
 ip address 10.80.11.17 255.255.255.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.11.20 exclusive
```

Global Cisco UBE Settings

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

```
voice service voip
 no ip address trusted authenticate
 mode border-element
 allow-connections sip to sip
 redundancy-group 1
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  rel1xx supported "rel100"
  header-passing
  asserted-id pai
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
  pass-thru content sdp
```

Explanation

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

Codecs

G711Ulaw is used as the preferred codec for this testing and changed the preferences according to the test plan description

```
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
```

Dial Peer

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

```
dial-peer voice 10 voip
 description Incoming from CUCM
 huntstop
 session protocol sipv2
 incoming called-number [0-9]T
 voice-class codec 2
 voice-class sip profiles 100
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 20 voip
 description Outgoing to Intelepeer
 huntstop
 destination-pattern [0-9]T
 session protocol sipv2
 session target dns:tekv.vsiptest.Intelepeer .com
 voice-class codec 2
 voice-class sip options-ping 60
 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 sg3-to-g3
 fax rate 14400
 fax nsf 000000
```

```
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 30 voip
 description Incoming from Intelepeer
 huntstop
 session protocol sipv2
 incoming called-number +1980
 voice-class codec 2
 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 sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 40 voip
 description Outgoing to CUCM
 huntstop
 destination-pattern +1980
 session protocol sipv2
 session target ipv4:10.80.11.2
 voice-class codec 2
 voice-class sip profiles 100
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 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 “7” 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 “7”. A “7.@” 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 7 prefix followed by the target 1+10Digit DID no for that extension number, 7 was stripped and the 1 +10 digits number was send to Cisco UBE, Cisco UBE sends the full 1+ 10 digits DID under Dial Peer 20 and send to Spectrum network which will direct back to Cisco UBE and handled same as normal incoming PSTN call. For International calls same pattern 7 followed by 011, country code and calling no is used.

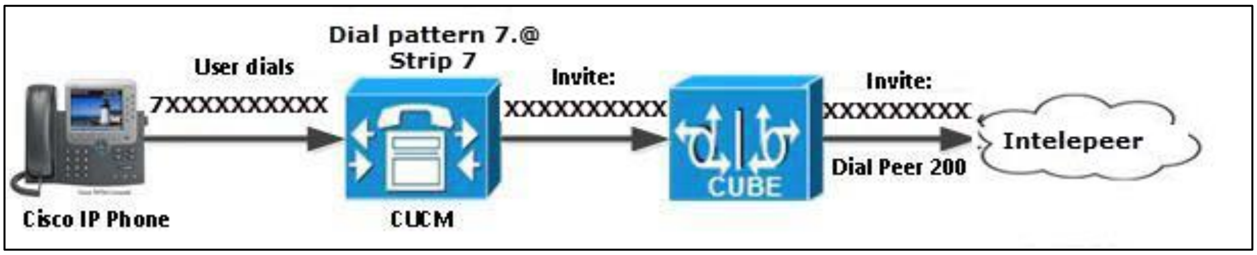

Figure 3: Outbound Voice Call

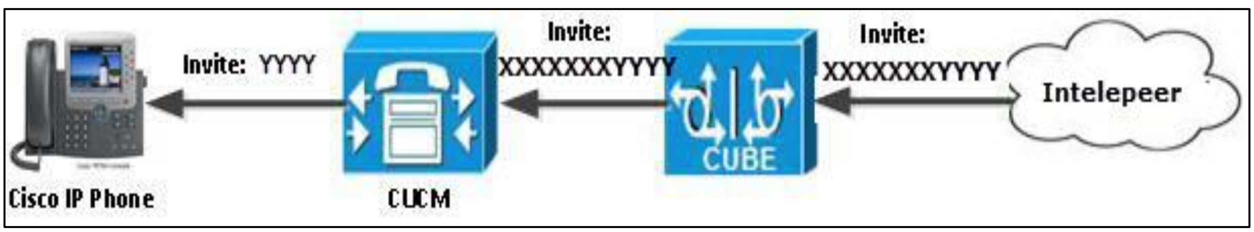

Figure 4: Inbound Voice Call

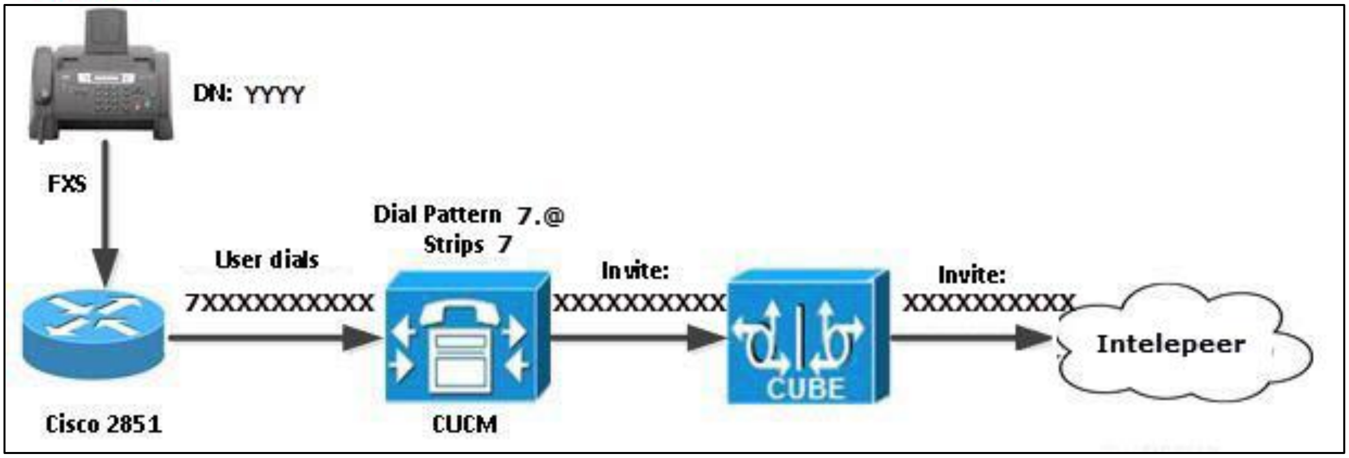

Figure 5: Outbound Fax Call

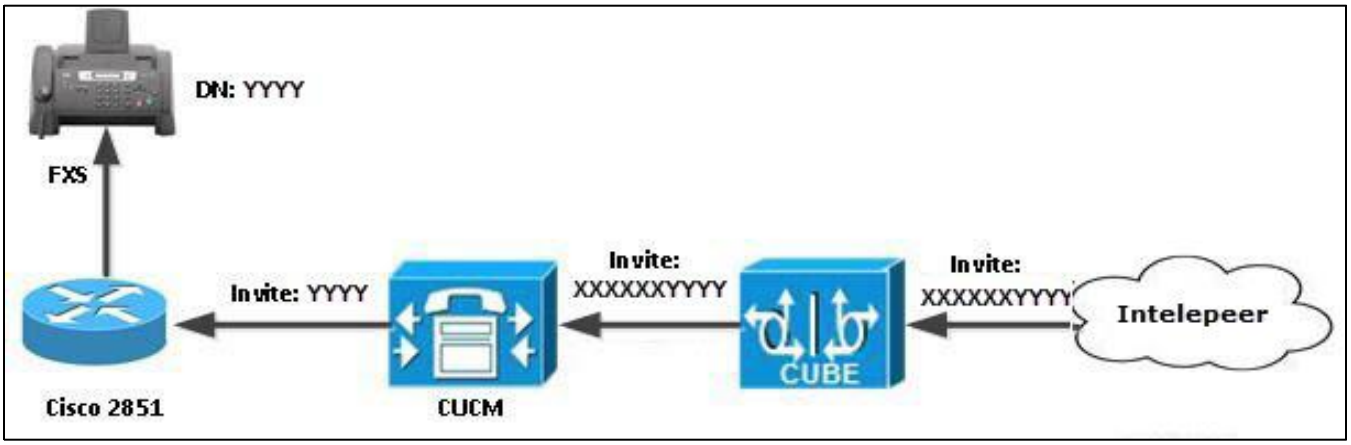

Figure 6: Inbound Fax Call

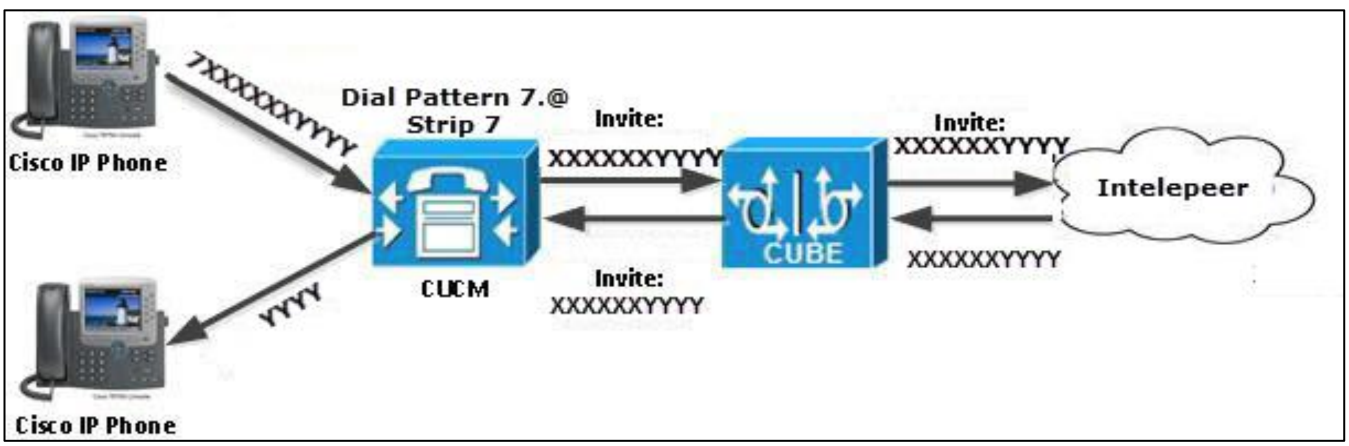

Figure 7: PBX to PBX via Spectrum Call

Configuration Example

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

Active Cisco UBE

```
User Access Verification
Username: cisco
Password:
isr4K2Intelepeer #sh run
isr4K2Intelepeer #sh running-config

version 16.5
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname isr4K2Intelepeer
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging console emergencies
enable secret 5 ******
!
no aaa new-model
!
ip name-server 8.8.8.8
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-988930787
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-988930787
```

```
 revocation-check none
 rsakeypair TP-self-signed-988930787
!
crypto pki certificate chain TP-self-signed-988930787
!
voice service voip
 no ip address trusted authenticate
 mode border-element
 allow-connections sip to sip
 redundancy-group 1
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  rel1xx supported "rel100"
  header-passing
  asserted-id pai
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
  pass-thru content sdp
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class sip-profiles 100
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:980233\1@\2"
!
license udi pid ISR4431/K9 sn FOC18232988
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level uck9
license boot level securityk9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 *****
!
```

```
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   timers delay 30 reload 60
   control GigabitEthernet0/0/3 protocol 1
   data GigabitEthernet0/0/3
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/2 line-protocol
!
track 2 interface GigabitEthernet0/0/0 line-protocol
!
interface GigabitEthernet0/0/0
 ip address 192.65.79.135 255.255.255.128
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.79.250 exclusive
!
interface GigabitEthernet0/0/1
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/2
 ip address 10.80.11.17 255.255.255.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.11.20 exclusive
!
interface GigabitEthernet0/0/3
 ip address 10.20.20.1 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
threat-visibility
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
```

```
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.255.0 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 10 voip
 description Incoming from CUCM
 huntstop
 session protocol sipv2
 incoming called-number [0-9]T
 voice-class codec 2
 voice-class sip profiles 100
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 20 voip
 description Outgoing to IntelePeer
 huntstop
 destination-pattern [0-9]T
 session protocol sipv2
 session target dns:tekv.vsiptest.Intelepeer .com
 voice-class codec 2
 voice-class sip options-ping 60
 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 sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 30 voip
 description Incoming from IntelePeer
 huntstop
 session protocol sipv2
 incoming called-number +1980
 voice-class codec 2
 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 sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
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 session protocol sipv2
 session target ipv4:10.80.11.2
 voice-class codec 2
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 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
sip-ua
!
line con 0
 exec-timeout 0 0
 transport input none
 stopbits 1
```

```
line aux 0
 stopbits 1
line vty 0 4
 logging synchronous
 login local
!
no network-clock synchronization automatic
ntp server pool.ntp.org
ntp server time-pnp.cisco.com
```

Standby Cisco UBE

```
isr4k1Intelepeer #sh running-config

version 16.5
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname isr4k1Intelepeer
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging console emergencies
enable secret 5
!
no aaa new-model
!
ip name-server 8.8.8.8
!
subscriber templating
!
multilink bundle-name authenticated
!
```

```
crypto pki trustpoint TP-self-signed-1179880555
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-1179880555
 revocation-check none
 rsakeypair TP-self-signed-1179880555
!
crypto pki certificate chain TP-self-signed-1179880555
!
voice service voip
 no ip address trusted authenticate
 mode border-element
 allow-connections sip to sip
 redundancy-group 1
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  rel1xx supported "rel100"
  header-passing
  asserted-id pai
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
  pass-thru content sdp
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class sip-profiles 100
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:980233\1@\2"
!
voice-card 0/1
 no watchdog
!
license udi pid ISR4431/K9 sn FOC18261KJL
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level uck9
license boot level securityk9
```

```
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 *****
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   timers delay 30 reload 60
   control GigabitEthernet0/0/3 protocol 1
   data GigabitEthernet0/0/3
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/2 line-protocol
!
track 2 interface GigabitEthernet0/0/0 line-protocol
!
interface GigabitEthernet0/0/0
 ip address 192.65.79.134 255.255.255.128
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.79.250 exclusive
!
interface GigabitEthernet0/0/1
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/2
 ip address 10.80.11.18 255.255.255.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.11.20 exclusive
!
interface GigabitEthernet0/0/3
 ip address 10.20.20.2 255.255.255.0
 negotiation auto
!
interface Service-Engine0/1/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
```

```
 no ip address
 negotiation auto
!
threat-visibility
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.2.100 255.255.255.255 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
control-plane
!
voice-port 0/1/0
!
voice-port 0/1/1
 !
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 10 voip
 description Incoming from CUCM
 huntstop
 session protocol sipv2
 incoming called-number [0-9]T
 voice-class codec 2
 voice-class sip profiles 100
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax-relay sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
```

```
dial-peer voice 20 voip
 description Outgoing to Intelepeer
 translation-profile outgoing 10
 huntstop
 destination-pattern [0-9]T
 session protocol sipv2
 session target dns:tekv.vsiptest.Intelepeer .com
 voice-class codec 2
 voice-class sip options-ping 60
 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-relay sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 30 voip
 description Incoming from Intelepeer
 huntstop
 session protocol sipv2
 incoming called-number +1980
 voice-class codec 2
 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-relay sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 40 voip
 description Outgoing to CUCM
 huntstop
 destination-pattern +1980
 session protocol sipv2
 session target ipv4:10.80.11.2
 voice-class codec 2
 voice-class sip profiles 100
```

```
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax-relay sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
sip-ua
!
line con 0
 exec-timeout 0 0
 logging synchronous
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 logging synchronous
 login local
!
no network-clock synchronization automatic
ntp server pool.ntp.org
ntp server time-pnp.cisco.com
```

Configuring Cisco Unified Communications Manager

Cisco UCM Version

Figure 8: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

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

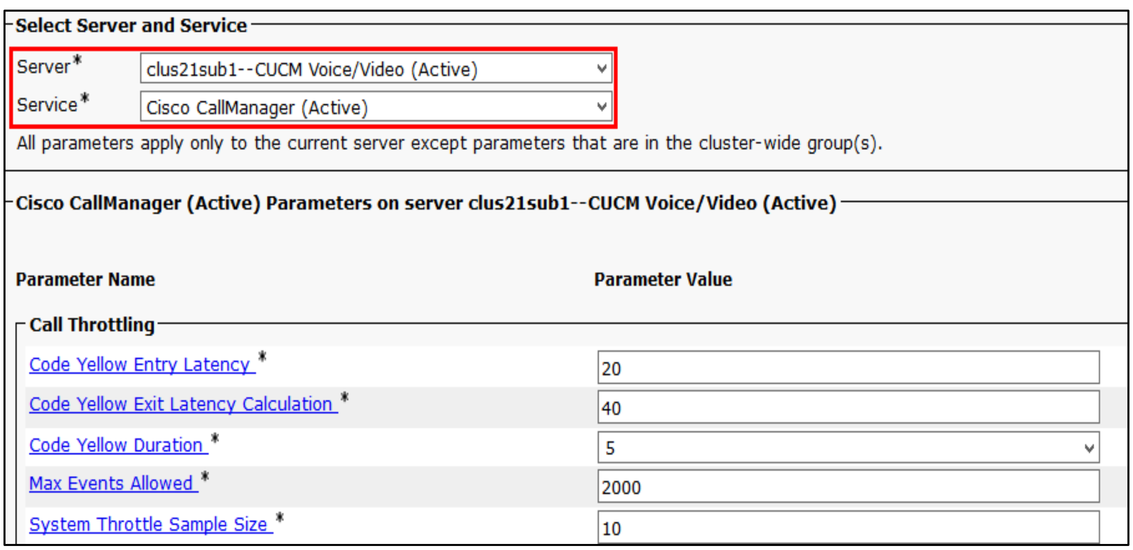


Figure 9: Service Parameters

Offnet Calls via IntelePeer SIP Trunk

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

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

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

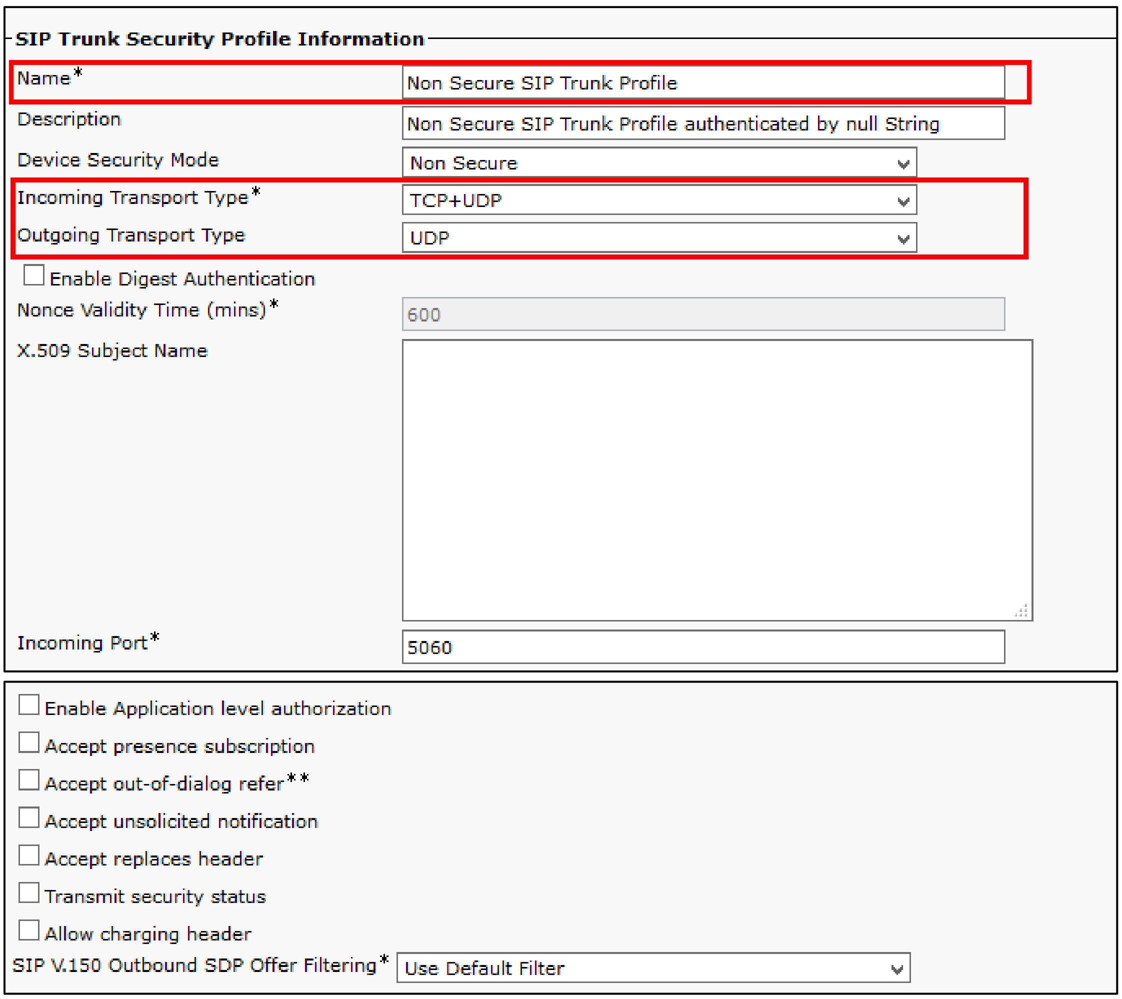


Figure 10: SIP Trunk Security Profile

Explanation

Parameter	**Value**	**Description**
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to 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: Device → Device Settings → SIP Profile

1. **Name***: Intelepeer SIP Profile
2. **Description**: Intelepeer SIP Profile

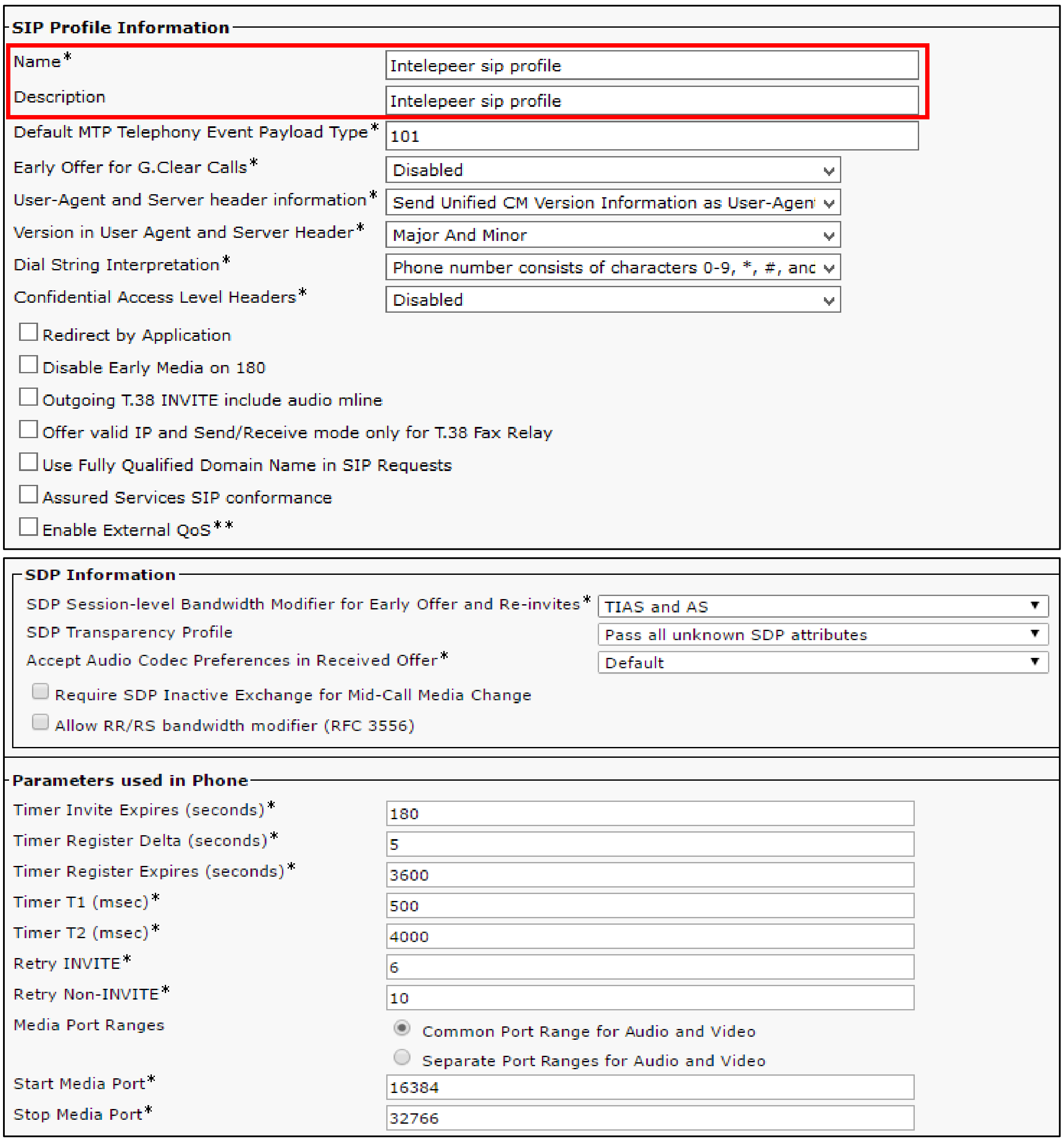


Figure 11: SIP Profile

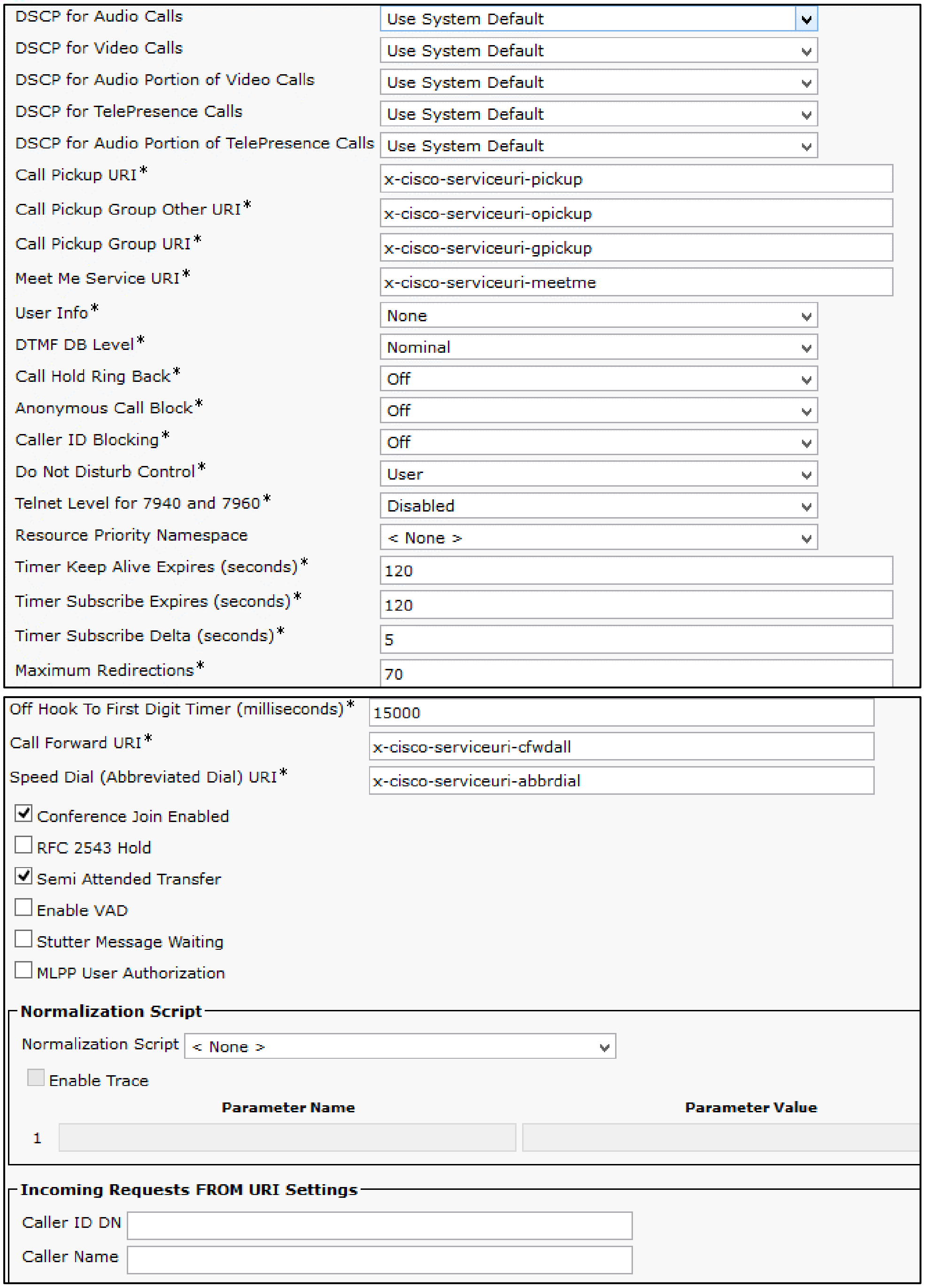


Figure 12: SIP Profile (Cont.)

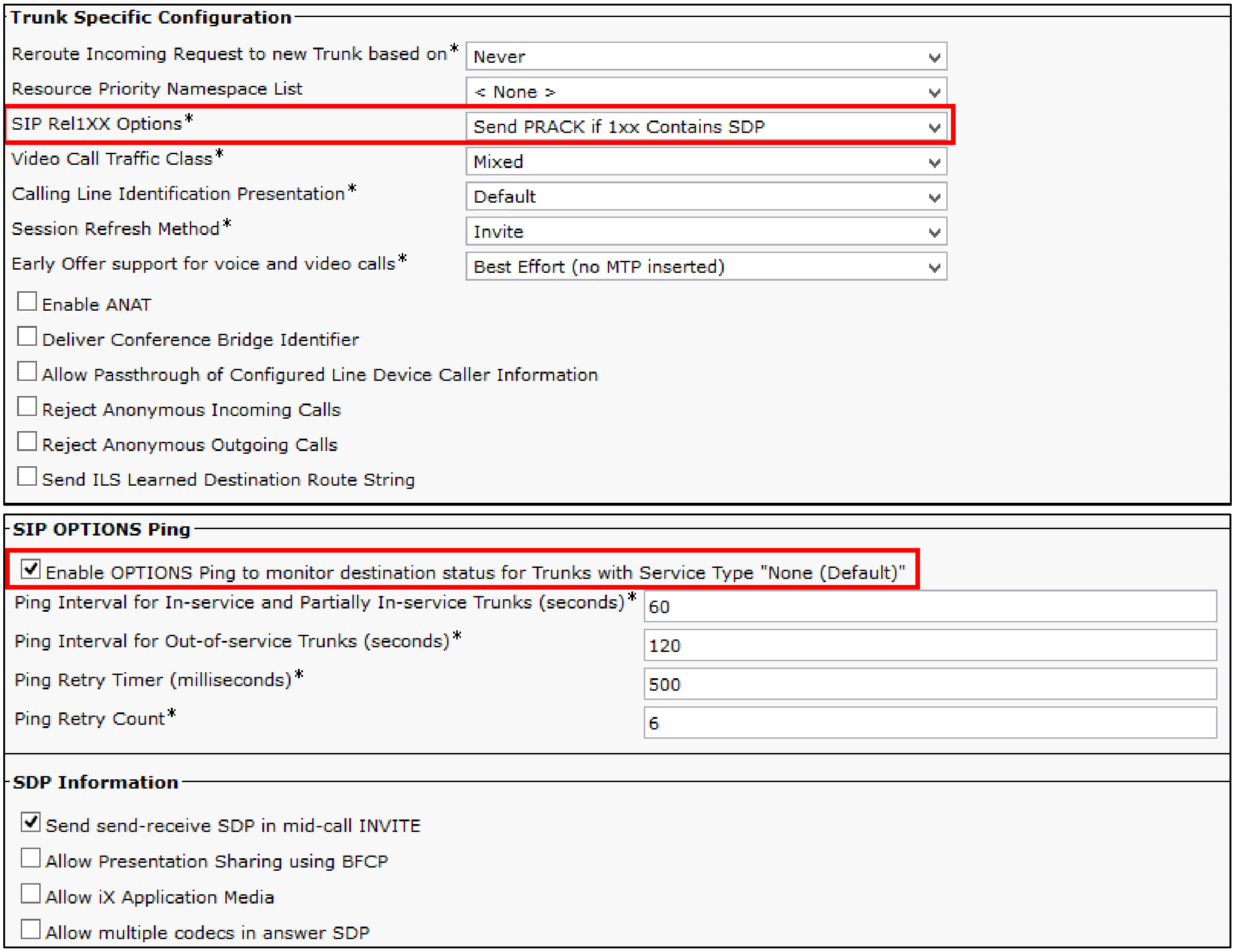


Figure 13: SIP Profile (Cont.)

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

	Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
SIP	cucm-faxgateway	cucm-faxgateway		G711 Pool	8021				SIP Trunk	Full Service	Time In Full Service: 0 day 9 hours 9 minutes	Non Secure SIP Trunk Profile
SIP	Intelepeer_Trunk	Intelepeer_Trunk		G729_Pool	7.@				SIP Trunk	Unknown - OPTIONS Ping not enabled		Non Secure SIP Trunk Profile

Figure 14: SIP Trunks List

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Intelepeer_Trunk
Description	Intelepeer_Trunk
Device Pool*	G711 Pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_MTP
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

- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [ ] 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 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

Figure 15: SIP Trunk to Cisco UBE

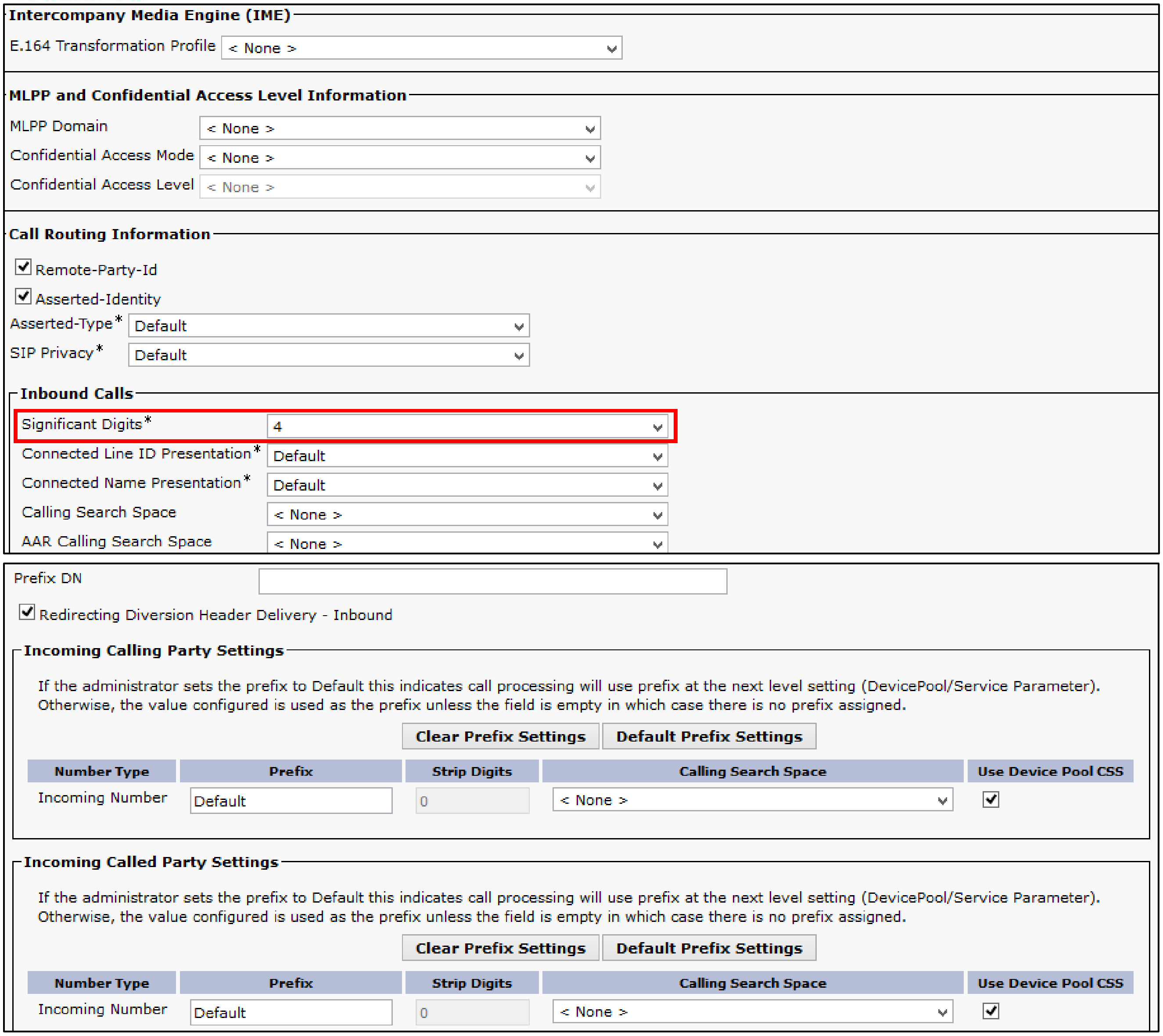
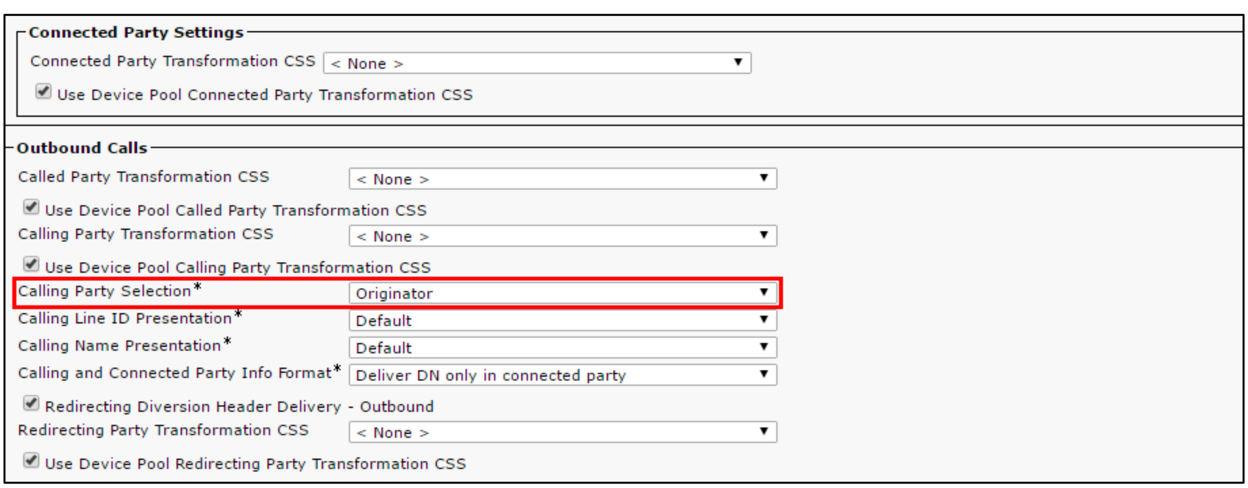


Figure 16: SIP Trunk to Cisco UBE (Cont.)



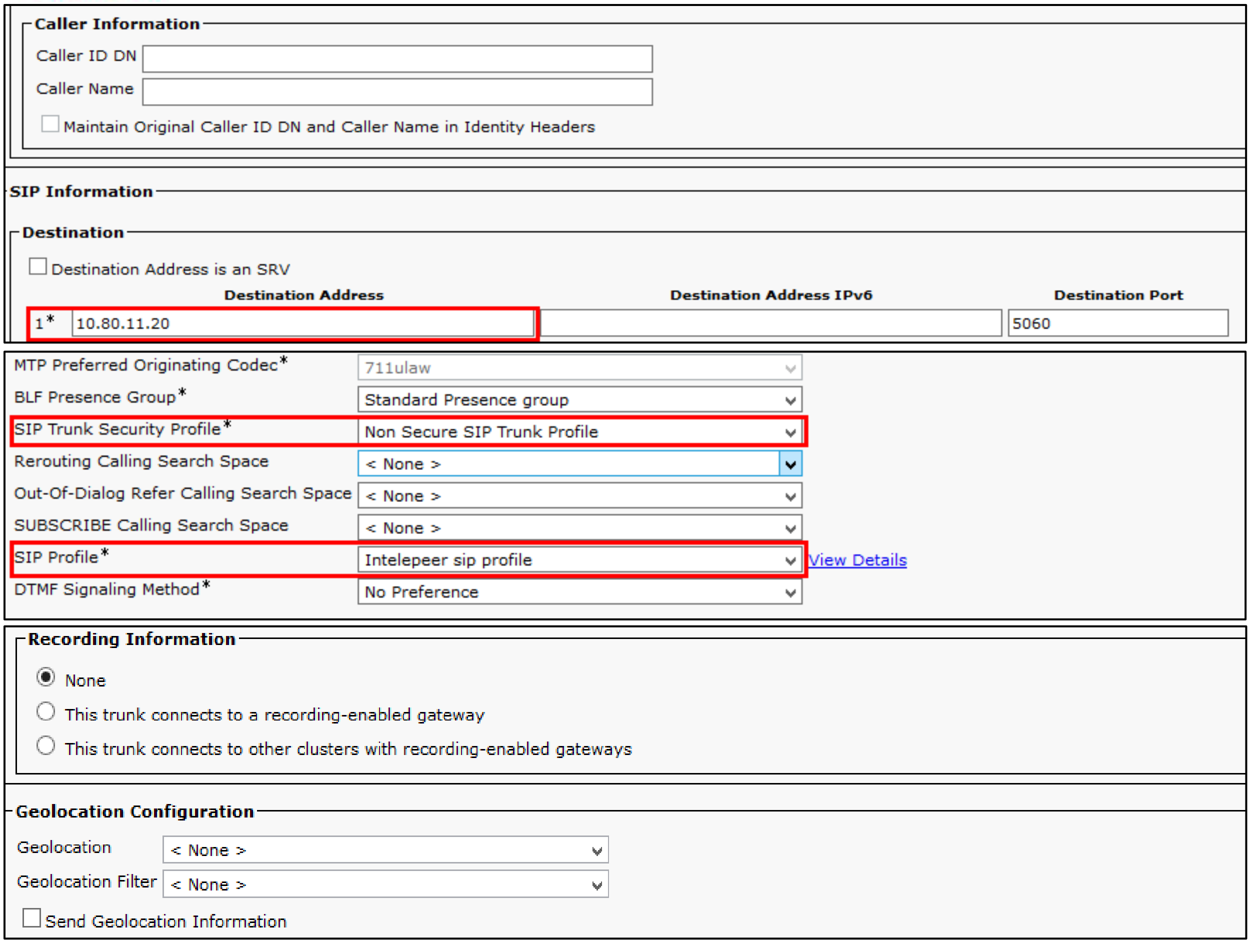


Figure 17: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	Intelepeer _Trunk	Name for the trunk
Device Pool	G711pool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_MTP	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.11.20	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Intelepeer SIP Profile	SIP Profile configured earlier

Dial Plan

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial “9”.1+10 digits number to access PSTN via Cisco UBE
  - “9” is removed before sending to Cisco UBE
- For FAX call, Access Code “9”+ 1+10 digits number is used at Cisco Fax gateway
  - “9” is removed at Cisco UCM
  - The rest of the number is sent to Cisco UBE to Intelepeer network
- Incoming fax call to 2063 will be sent to Cisco Fax gateway

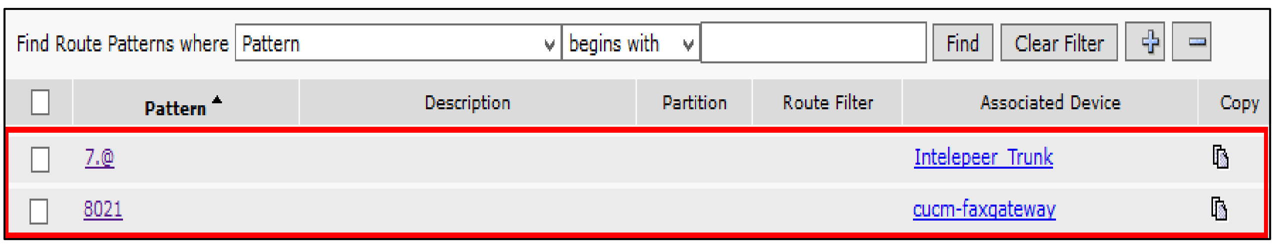
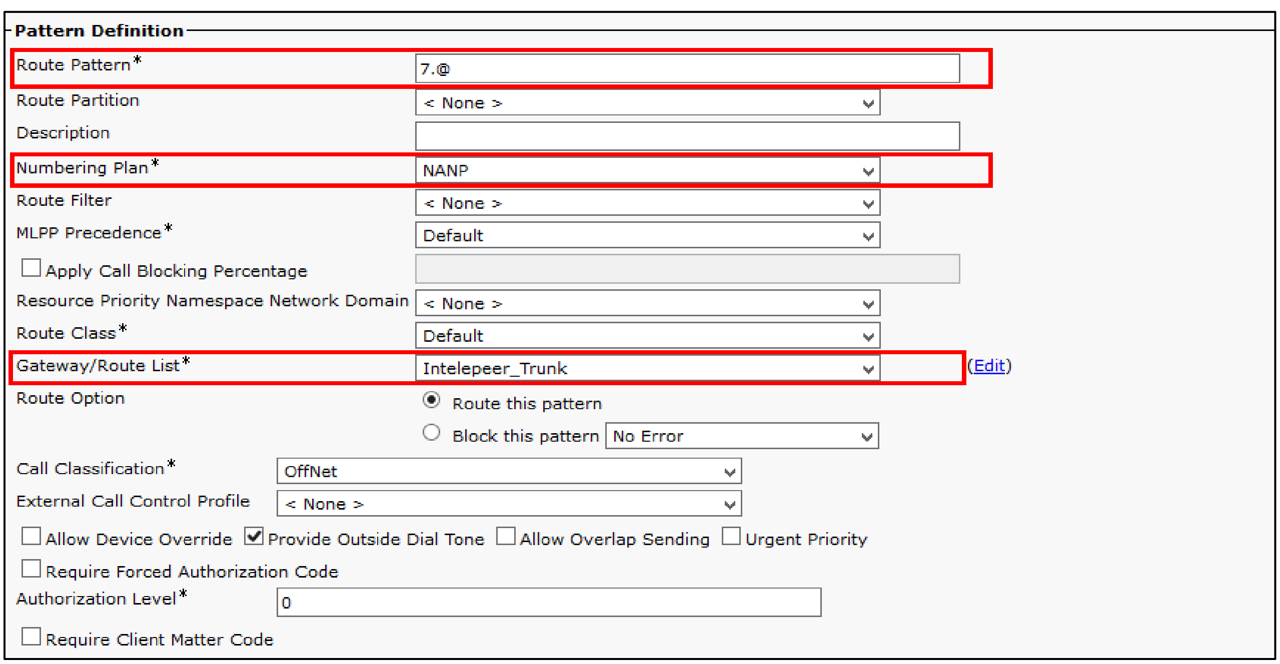
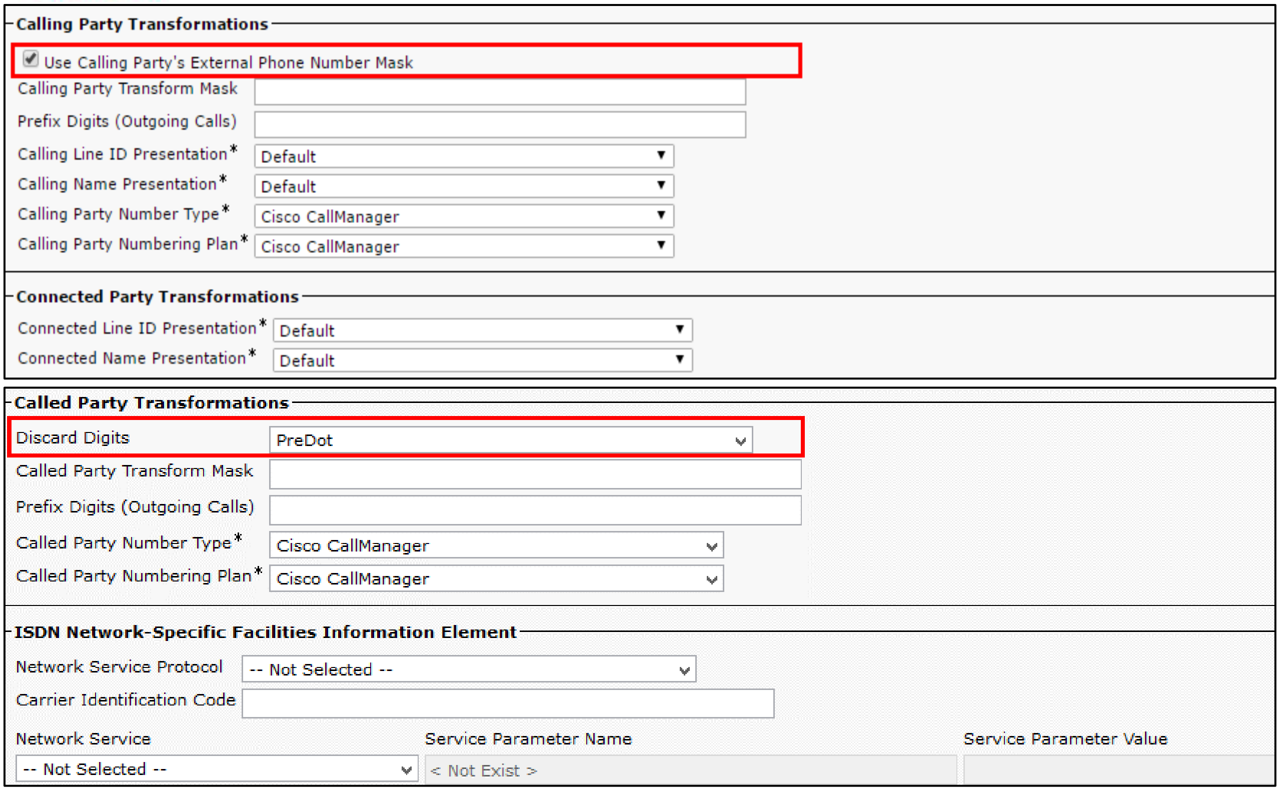


Figure 18: Route Patterns List




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 >	

Figure 19: Route Pattern for Voice

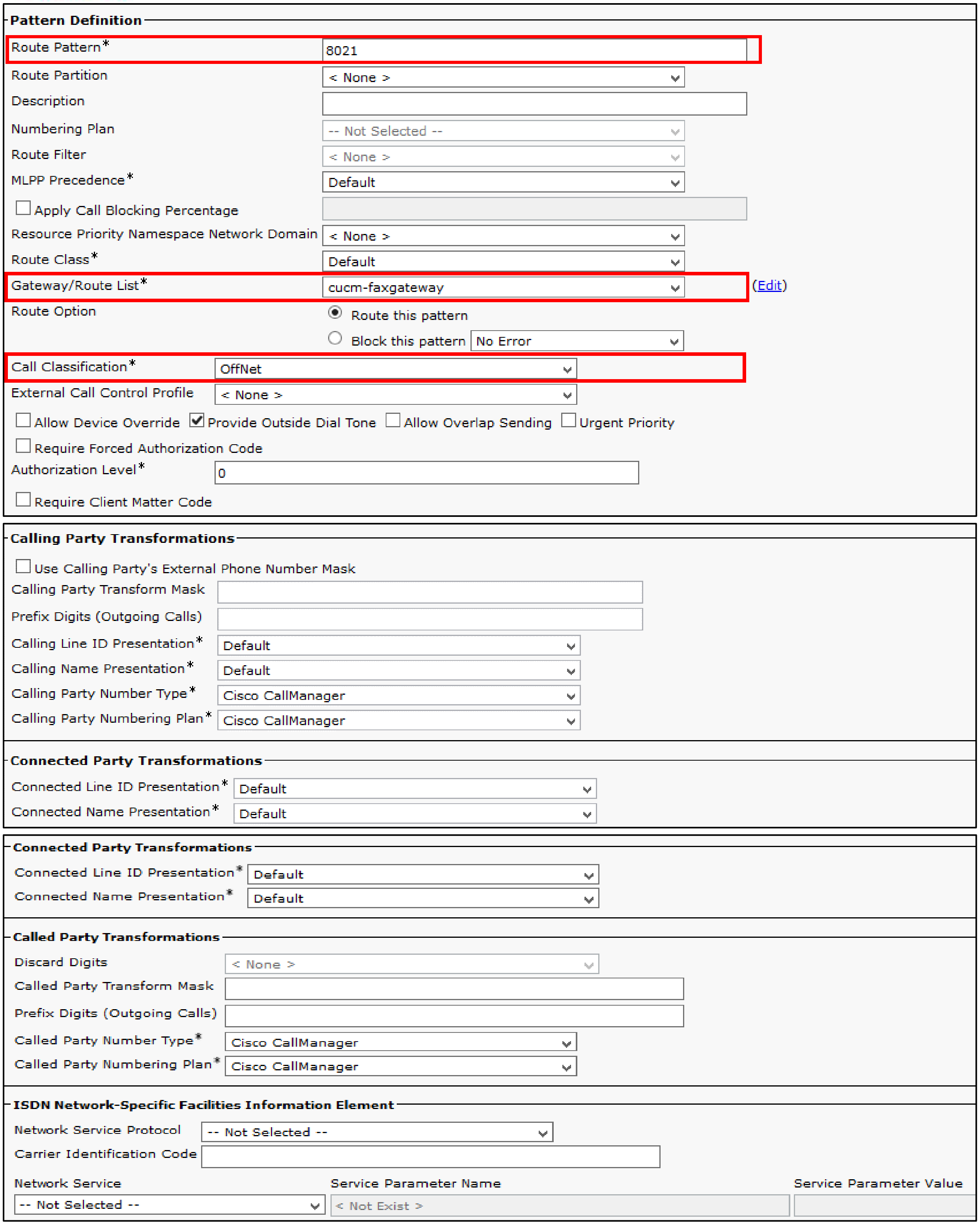


Figure 20: Route Pattern for Fax

Explanation

Setting	Value	Description
Route Pattern	7.@ for Voice & International Calls and 8021 for Fax Call	Specify appropriate Route Pattern
Gateway/Route List	Intelepeer for Route Pattern 7.@ and 8021 for SIP Trunk To Fax Gateway.	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 7.@	North American Numbering Plan
Call Classification	OffNet for Route Pattern 7.@ and 8021	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 7.@	Specifies how to modify digit before they are sent to Spectrum network

Acronyms

Acronym	Definition
CPE	Customer Premise Equipment
Cisco UBE	Cisco Unified Border Element
Cisco UCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
ESBC	Enterprise Session Border Controller
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol