TELUS IP Trunking: Connecting Cisco Unified Communications Manager 11 via the Cisco Unified Border Element 11.5 (Enterprise Edition) using SIP

May 2016

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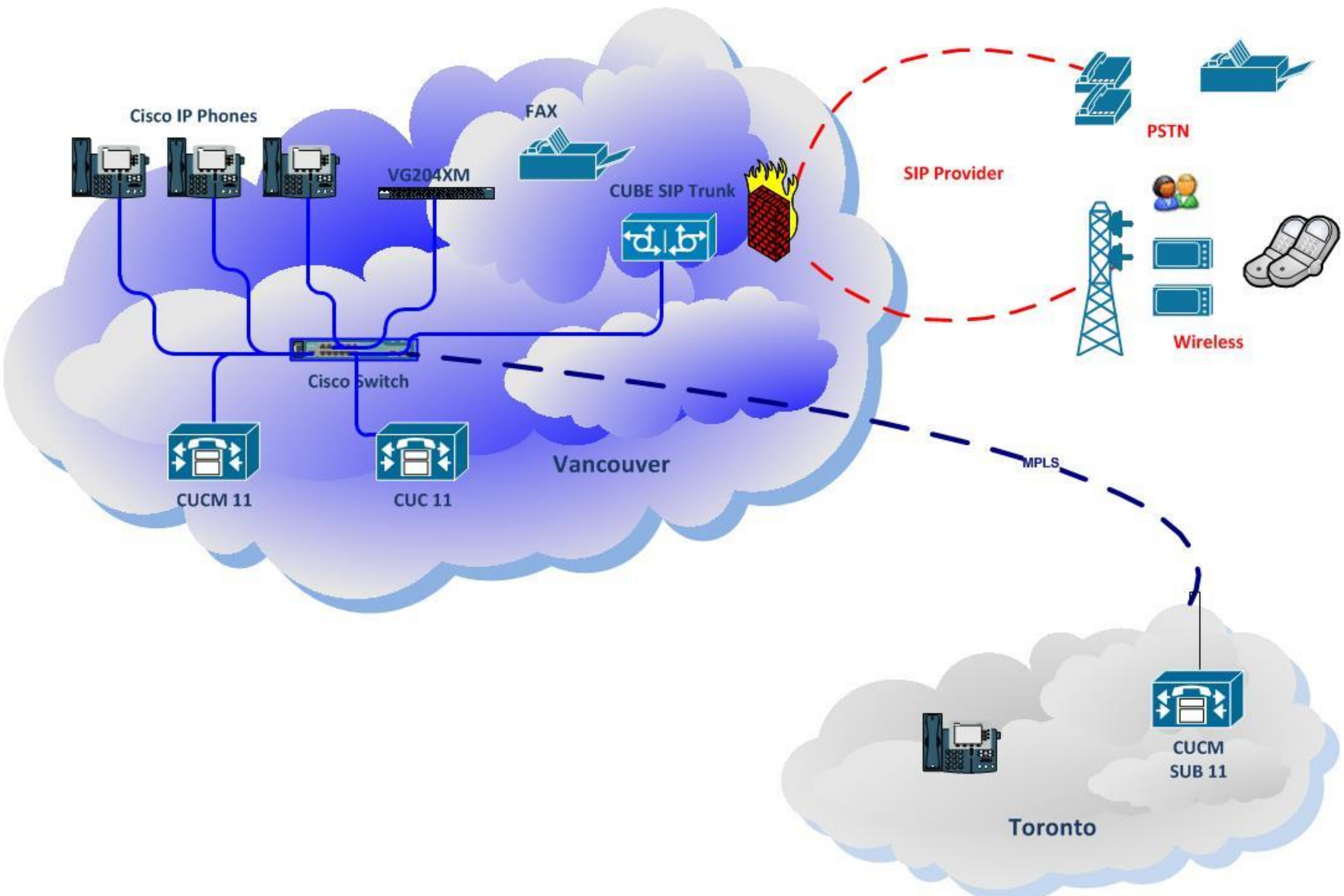
Note: Testing was conducted in Telus labs

Introduction

Service Providers today, such as TELUS, 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 a centralized IP to TDM gateway to provide on-net and off-net services. TELUS IP Trunking is a SP offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either Analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element provides demarcation, security, interworking and session management services.

- This application note describes how to configure a Cisco Unified Communications Manager (CUCM) 11.0.1.20000-2 with a Cisco Unified Border Element (CUBE) 11.5 [IOS 15.6(1)T0a] for connectivity to TELUS IP Trunking SIP trunk service. The deployment model covered in this application note is CPE to PSTN. This document does not address 911 emergency outbound calls. For 911 feature service details contact TELUS directly.
- Testing was performed in accordance to TELUS test plan and all features were verified. Key features verified are: Listed under features in this document.
- The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between TELUS SIP network and Cisco Unified Communications. The configuration described in this document details the important commands to have enabled for interoperability to be successful and care must be taken, by the network administrator deploying CUBE, to ensure these commands are set per each dial-peer requiring to interoperate to TELUS SIP network.
- This application note does not cover the use of Calling Search Spaces (CSS) or Partitions on Cisco Unified Communications Manager. To understand and learn how to apply CSS and Partitions refer to the cisco.com website.

Network Topology



System Components

Hardware Components

- Cisco 2911/K9 (Cisco 2900 family router)
- CUCM cluster with (2) Cisco MCS 7800 Series server (Cisco Unified Communications Manager)
- 4 Cisco Unified IP Phones (7965)
- 1 Cisco IP Communicator
- Cisco 3560C powered Ethernet switch
- Unity connection for voice mail as well as Auto Attendant
- VG204XM MGCP gateway for fax
- Fax machine G3

Software Requirements

The following software is required:

- Cisco Unified Communications Manager Release 11. This solution was tested with 11.0.1.20000-2
- Cisco Unity Connection Release 11. This solution was tested with 11.0.1.20000-2
- Cisco Unified Border Element Release 11.5 with IOS version 15.6.(1)T0a release. This configuration was tested with c2900-universalk9-mz.SPA.156-1.T0a.bin
- Cisco VG204XM with IOS version 15.3.2T. This configuration was tested with vg20xxm-ipvoice-mz.153-2.T.bin

Features

Features Supported

- Basic Call using G.729 and G711 (inbound and outbound).
- Calling Party Number Presentation (CLIP).
- Calling Party Number Restriction (CLIR).
- Calling Name.
- Intra-site Call Transfer (Attended and Unattended).
- Intra-site Conference.
- Call Hold and Resume.
- Call Forward All, Busy and No Answer.
- Toll-free numbers.
- Long calls durations.
- DTMF (RFC2833).
- Fax using G.711 pass-through.
- Cisco Unity Connection Auto-attendant.
- Cisco Unity Connection Voice mail.
- Calling number privacy.

Features Not Supported

- Emergency 911 calls were not tested.
- Failover was not tested.

Configuration

Configuring Cisco Unified Border Element (CUBE)

Show Version

```
Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.6(1)T0a, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2015 by Cisco Systems, Inc.
Compiled Thu 03-Dec-15 15:16 by prod_rel_team
ROM: System Bootstrap, Version 15.0(1r)M16, RELEASE SOFTWARE (fc1)
XXXXXXX uptime is X weeks, X days, X hours, X minutes
System returned to ROM by power-on
System restarted at 03:07:42 PDT Fri Apr 15 2016
System image file is "flash:c2900-universalk9-mz.SPA.156-1.T0a.bin"
Last reload type: Normal Reload
Last reload reason: power-on
This product contains cryptographic features and is subject to United
States and local country laws governing import, export, transfer and
use. Delivery of Cisco cryptographic products does not imply
third-party authority to import, export, distribute or use encryption.
Importers, exporters, distributors and users are responsible for
compliance with U.S. and local country laws. By using this product you
agree to comply with applicable laws and regulations. If you are unable
to comply with U.S. and local laws, return this product immediately.
A summary of U.S. laws governing Cisco cryptographic products may be found at:
http://www.cisco.com/wwl/export/crypto/tool/stqrg.html
If you require further assistance please contact us by sending email to
export@cisco.com.
Cisco CISCO2911/K9 (revision 1.0) with 475136K/49152K bytes of memory.
Processor board ID FGL174711EA
4 Gigabit Ethernet interfaces
2 terminal lines
1 Channelized (E1 or T1)/PRI port
1 Virtual Private Network (VPN) Module
1 Internal Services Module (ISM) with Services Ready Engine (SRE)
Cisco Unity Express 8.6.6 in slot/sub-slot 0/0
DRAM configuration is 64 bits wide with parity enabled.
255K bytes of non-volatile configuration memory.
250880K bytes of ATA System CompactFlash 0 (Read/Write)
License Info:
License UDI:
-------------------------------------------------
Device#   PID                 SN
-------------------------------------------------
*1        CISCO2911/K9        FGL174711EA
Suite License Information for Module:'c2900'
--------------------------------------------------------------------------------
Suite               Suite Current         Type           Suite Next reboot
--------------------------------------------------------------------------------
FoundationSuiteK9     None                  None           None
securityk9
datak9
AdvUCSuiteK9          None                  None           None
uck9
cme-srst
cube
Technology Package License Information for Module:'c2900'
-----------------------------------------------------------------------
Technology    Technology-package                  Technology-package
              Current             Type           Next reboot
-----------------------------------------------------------------------
ipbase        ipbasek9            Permanent      ipbasek9
security      securityk9          Permanent      securityk9
uc            uck9                Permanent      uck9
data          datak9              Permanent      datak9
Configuration register is 0x2102
```

Show Running-Configuration

```
Current configuration: 6140 bytes
! Last configuration change at 22:25:11 PDT Sun May 1 2016 by admin
! NVRAM config last updated at 10:28:18 PDT Tue Apr 26 2016 by admin
!
version 15.6
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
service sequence-numbers
!
hostname XYZ
!
boot-start-marker
boot system flash c2900-universalk9-mz.SPA.156-1.T0a.bin
boot-end-marker
!
aqm-register-fnf
!
! card type command needed for slot/vwic-slot 0/0
logging buffered 10000000
no logging rate-limit
no logging console
!
no aaa new-model
ethernet lmi ce
clock timezone EST -8 0
clock summer-time PDT recurring
!
ip domain name xyz
ip cef
no ipv6 cef
!
multilink bundle-name authenticated
!
voice-card 0
 dspfarm
 dsp services dspfarm
!
voice service voip[1]
 no ip address trusted authenticate
 mode border-element license capacity 25
 allow-connections sip to sip
 redirect ip2ip
 fax protocol pass-through g711ulaw
 h323
 sip
  referto-passing
  asserted-id pai[2]
  asymmetric payload dtmf[3]
  early-offer forced[4]
  midcall-signaling passthru[5]
  privacy-policy passthru[6]
  g729 annexb-all
!
```

---

[1] This introduces the mode border-element command to distinguish between Cisco Unified Communications Manager Express and Cisco UBE configuration.

[2] Enables the P-Asserted-Identity (PAI) privacy header in incoming and outgoing SIP requests or response messages.

```
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class sip-profiles 150
 request INVITE sip-header From modify "Anonymous" "IPT-Cert-Phone"
 request INVITE sip-header From modify "<sip:anonymous@anonymous.invalid>" "<sip:+1234567890@X.X.X.X>"
!
voice iec syslog
!
license udi pid CISCO2911/K9 sn FGL174711EA
hw-module ism 0
!
hw-module pvdm 0/0
!
username YYYYY privilege 15 secret XXXXXXXXXXXXX
!
redundancy
!
interface Embedded-Service-Engine0/0
 no ip address
 shutdown
!
interface GigabitEthernet0/0
 description OUTSIDE CUBE Interface
 ip address X.X.X.X 255.255.255.252
 duplex auto
 speed auto
!
interface ISM0/0
 no ip address
 shutdown
 !Application: CUE Running on ISM
!
interface GigabitEthernet0/1
 description INSIDE CUBE Interface
 ip address X.X.X.X 255.255.255.0
 duplex auto
 speed auto
!
interface GigabitEthernet0/2
 no ip address
duplex auto
 speed auto
!
interface ISM0/1
 description Internal switch interface connected to Internal Service Module
 no ip address
!
interface Vlan1
 no ip address
!
ip forward-protocol nd
no ip http server
no ip http secure-server
!
ip route X.X.X.X 255.255.255.192 Y.Y.Y.Y
```

---

[3] Specifies that the asymmetric payload support is dual-tone multi-frequency (DTMF) only.

[4] To force a Cisco Unified Border Element (Cisco UBE) to send a SIP invite with Early-Offer (EO) on the Out-Leg (OL), use the early-offer command in SIP or dial peer configuration mode.

[5] Passes SIP messages that involve media-change from one IP leg to another IP leg.

[6] Passes the privacy values from the received message to the next call leg.

```
!
logging trap debugging
logging host X.X.X.X
!
control-plane
!
sccp local GigabitEthernet0/1
sccp ccm 172.24.0.120 identifier 1 version 7.0
sccp
!
sccp ccm group 1
 bind interface GigabitEthernet0/1
 associate ccm 1 priority 1
 associate profile 2 register MTP24e9b3735829
 associate profile 1 register CFB24e9b3735829
!
dspfarm profile 2 transcode
 codec g729br8
 codec g729r8
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 maximum sessions 3
 associate application SCCP
!
dspfarm profile 1 conference
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 maximum sessions 3
 associate application SCCP
!
dial-peer voice 9900 voip
 description "IPT-Cert-CUCM-DID-AB"
 destination-pattern 587756993[5-9]
 signaling forward unconditional
 session protocol sipv2
 session target ipv4:A.A.A.A[7]
 voice-class codec 1
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax protocol pass-through g711ulaw
!
dial-peer voice 2000 voip
 description "TELUS LAB SIP Trunk - NA calls"
 preference 1
 destination-pattern 1..........
 signaling forward unconditional[8]
 session protocol sipv2
 session target ipv4:B.B.B.B[9]
 voice-class codec 1
 voice-class sip g729 annexb-all
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
```

[7] Cisco Unified Communication System

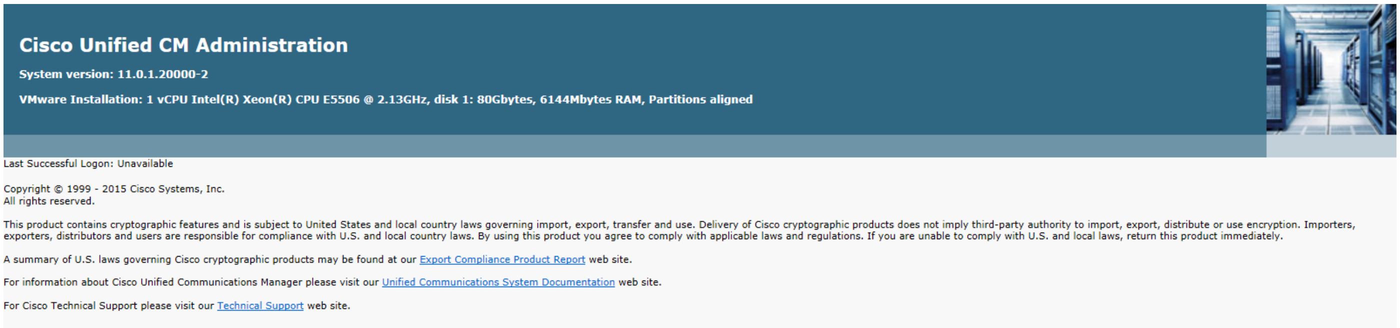
[8] Tunnels Generic Transparency Descriptor (GTD), payload along with QSIG or Q.931 message bodies.

[9] Telus Main SBC IP address

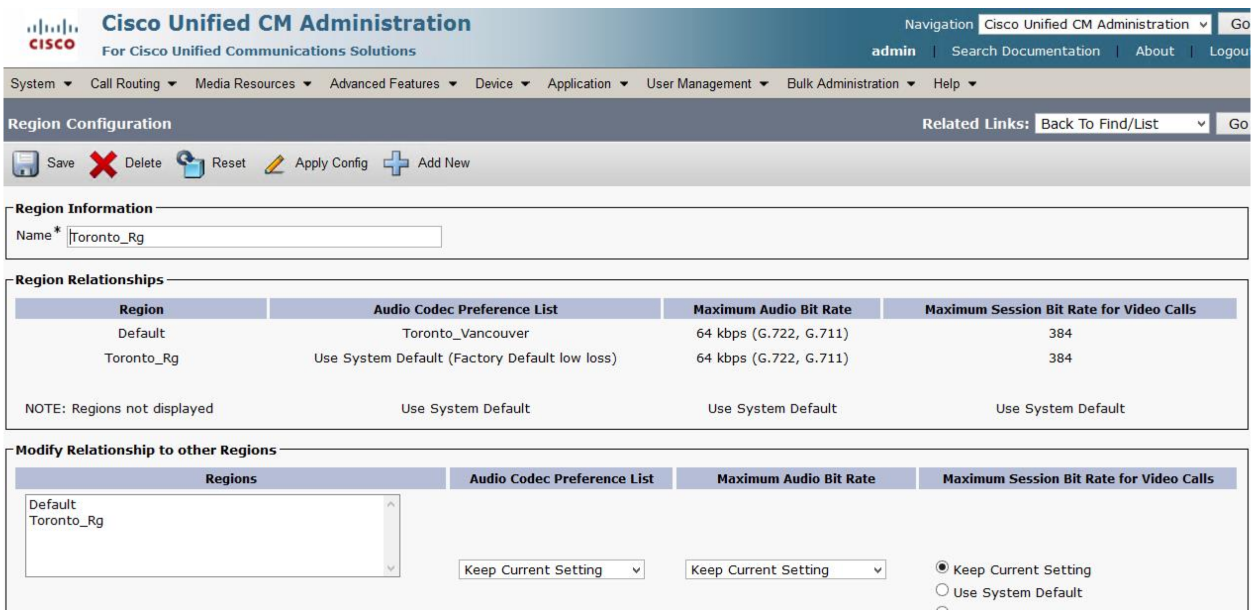
```
!
!
dial-peer voice 2001 voip
 description "TELUS LAB SIP Trunk - International"
 preference 1
 destination-pattern 011T
 signaling forward unconditional
session protocol sipv2
 session target ipv4: B.B.B.B
 voice-class codec 1
 voice-class sip g729 annexb-all
 voice-class sip early-offer forced
 voice-class sip profiles 150
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
!
!
dial-peer voice 2002 voip
 description "TELUS LAB SIP Trunk - Operator"
 preference 1
 destination-pattern 0T
 signaling forward unconditional
 session protocol sipv2
 session target ipv4: B.B.B.B
 voice-class codec 1
 voice-class sip g729 annexb-all
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax protocol pass-through g711ulaw
 no vad
!
!
!
!
gatekeeper
 shutdown
!
line con 0
line aux 0
line 2
 no activation-character
 no exec
 transport preferred none
 transport output lat pad telnet rlogin lapb-ta mop udptn v120 ssh
 stopbits 1
line 131
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output lat pad telnet rlogin lapb-ta mop udptn v120 ssh
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 login local
 transport input ssh
!
!
scheduler allocate 20000 1000
ntp source GigabitEthernet0/1
ntp master
!
end
```

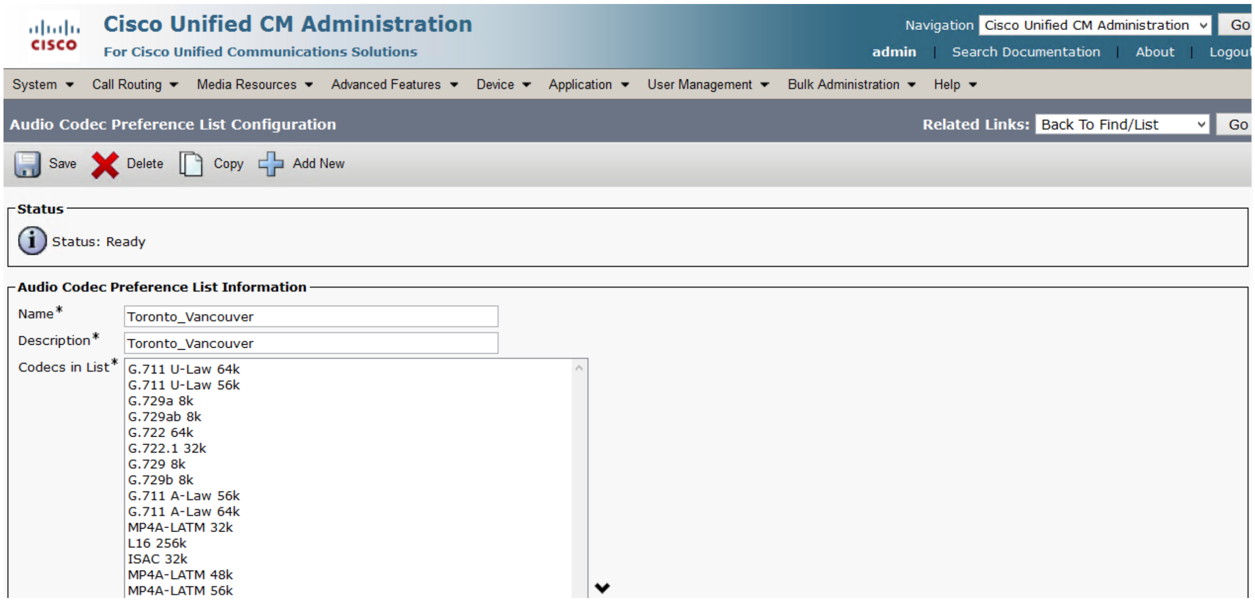
Configuring the Cisco Unified Communications Manager

System Version

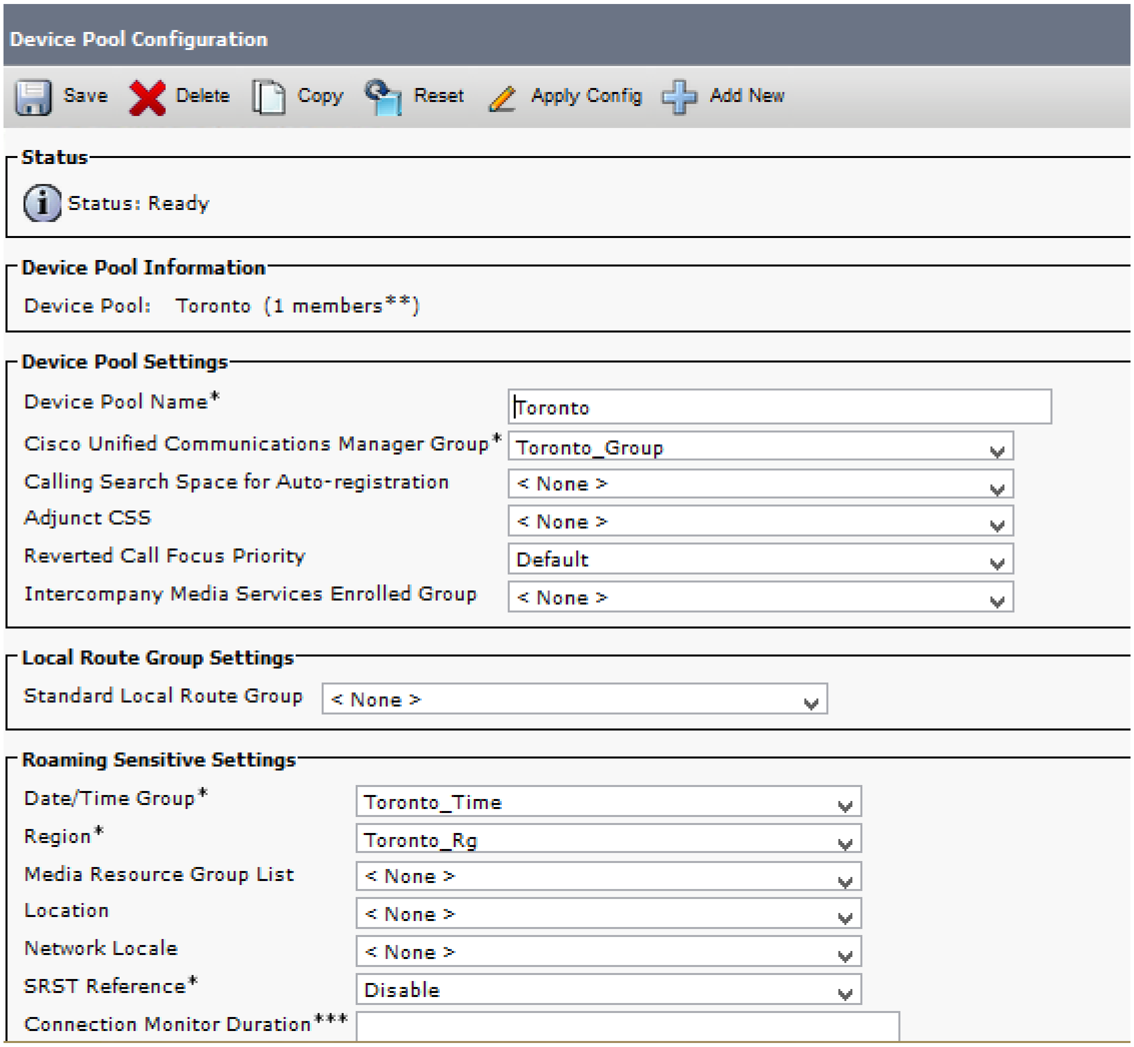


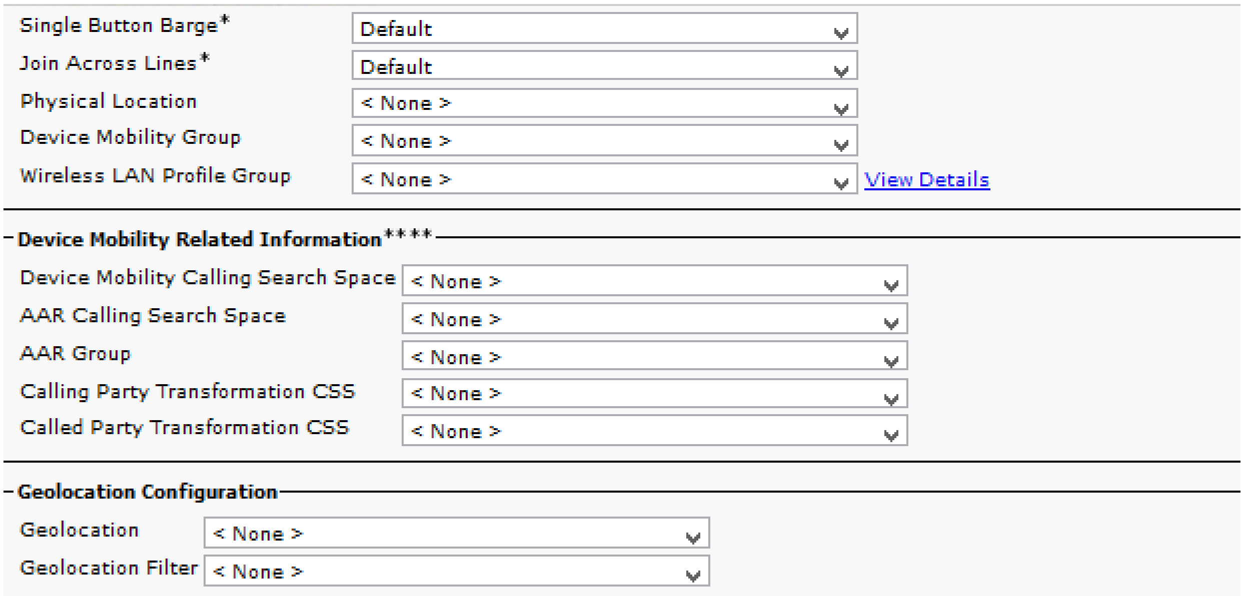
Region (Codec settings)




Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration Go
admin | Search Documentation | About | Logout
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Audio Codec Preference List Configuration
Related Links: Back To Find/List Go
Save Delete Copy Add New
Status
Status: Ready
Audio Codec Preference List Information
Name*
Toronto_Vancouver
Description*
Toronto_Vancouver
Codecs in List*
G.711 U-Law 64k
G.711 U-Law 56k
G.729a 8k
G.729ab 8k
G.722 64k
G.722.1 32k
G.729 8k
G.729b 8k
G.711 A-Law 56k
G.711 A-Law 64k
MP4A-LATM 32k
L16 256k
ISAC 32k
MP4A-LATM 48k
MP4A-LATM 56k

Device Pool


Device Pool Configuration
Save
Delete
Copy
Reset
Apply Config
Add New
Status
Status: Ready
Device Pool Information
Device Pool: Toronto (1 members**)
Device Pool Settings
Device Pool Name*
Toronto
Cisco Unified Communications Manager Group*
Toronto_Group
Calling Search Space for Auto-registration
< None >
Adjunct CSS
< None >
Reverted Call Focus Priority
Default
Intercompany Media Services Enrolled Group
< None >
Local Route Group Settings
Standard Local Route Group
< None >
Roaming Sensitive Settings
Date/Time Group*
Toronto_Time
Region*
Toronto_Rg
Media Resource Group List
< None >
Location
< None >
Network Locale
< None >
SRST Reference*
Disable
Connection Monitor Duration***



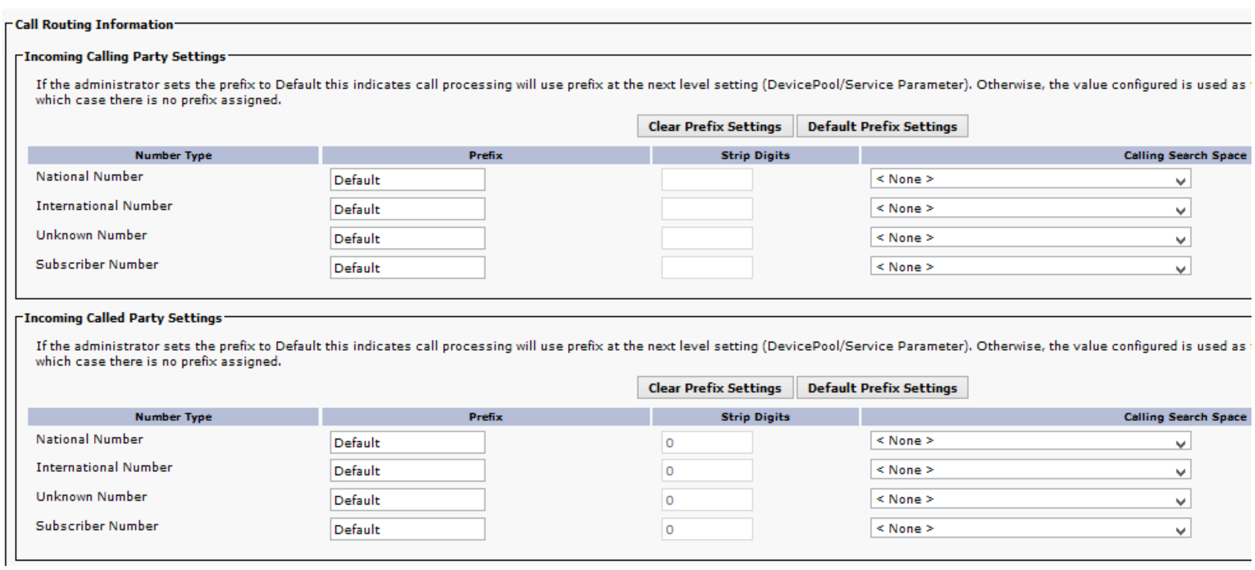
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >
Wireless LAN Profile Group	< None > View Details

Device Mobility Related Information**

Device Mobility Calling Search Space	< None >
AAR Calling Search Space	< None >
AAR Group	< None >
Calling Party Transformation CSS	< None >
Called Party Transformation CSS	< None >

Geolocation Configuration

Geolocation	< None >
Geolocation Filter	< None >



Call Routing Information

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
which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

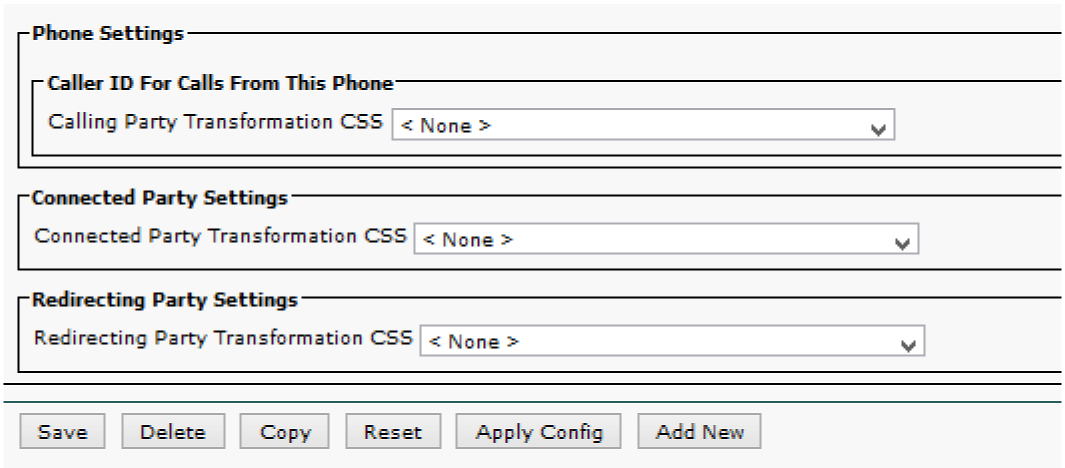
Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default		< None >
International Number	Default		< None >
Unknown Number	Default		< None >
Subscriber Number	Default		< None >

Incoming Called Party Settings

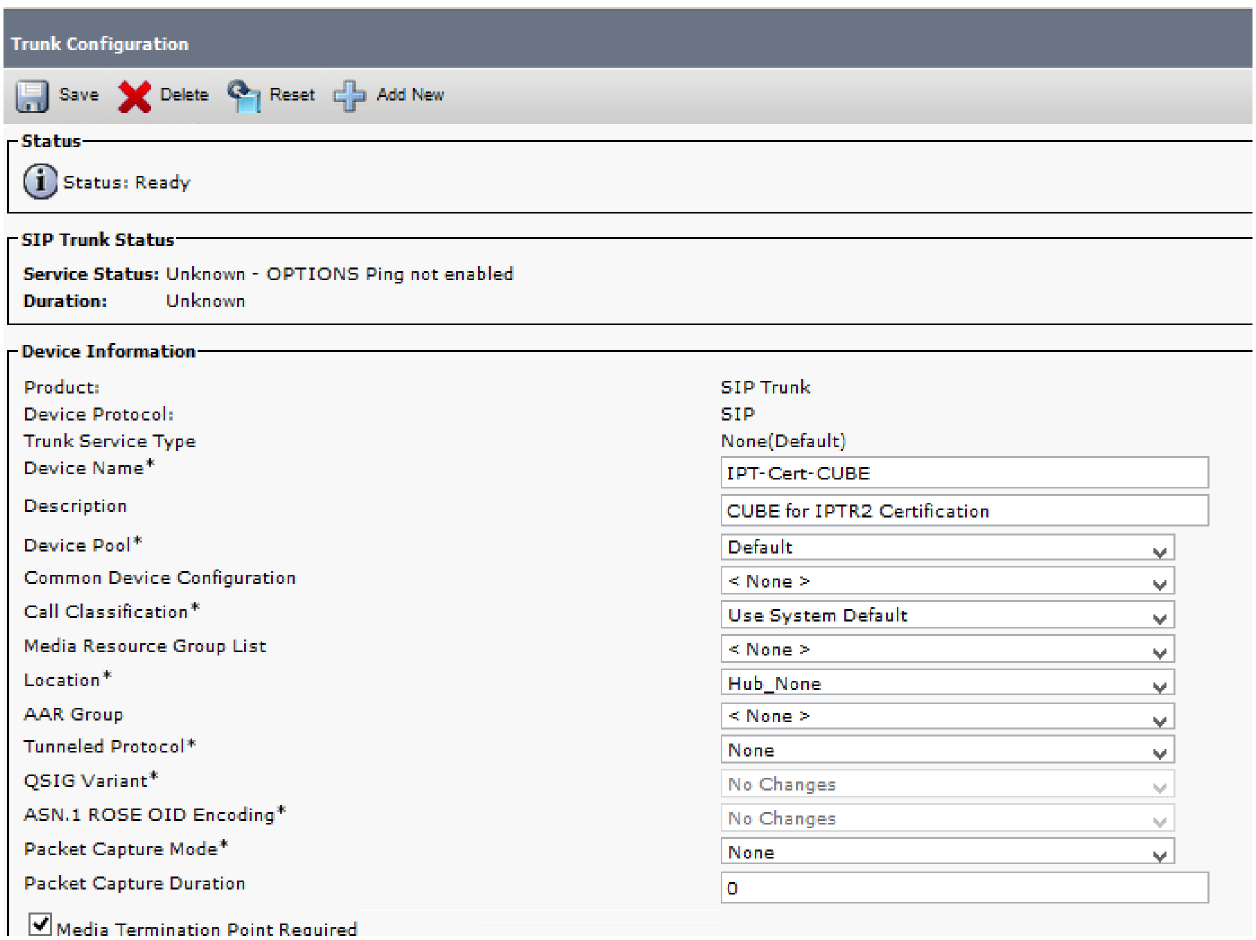
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as
which case there is no prefix assigned.

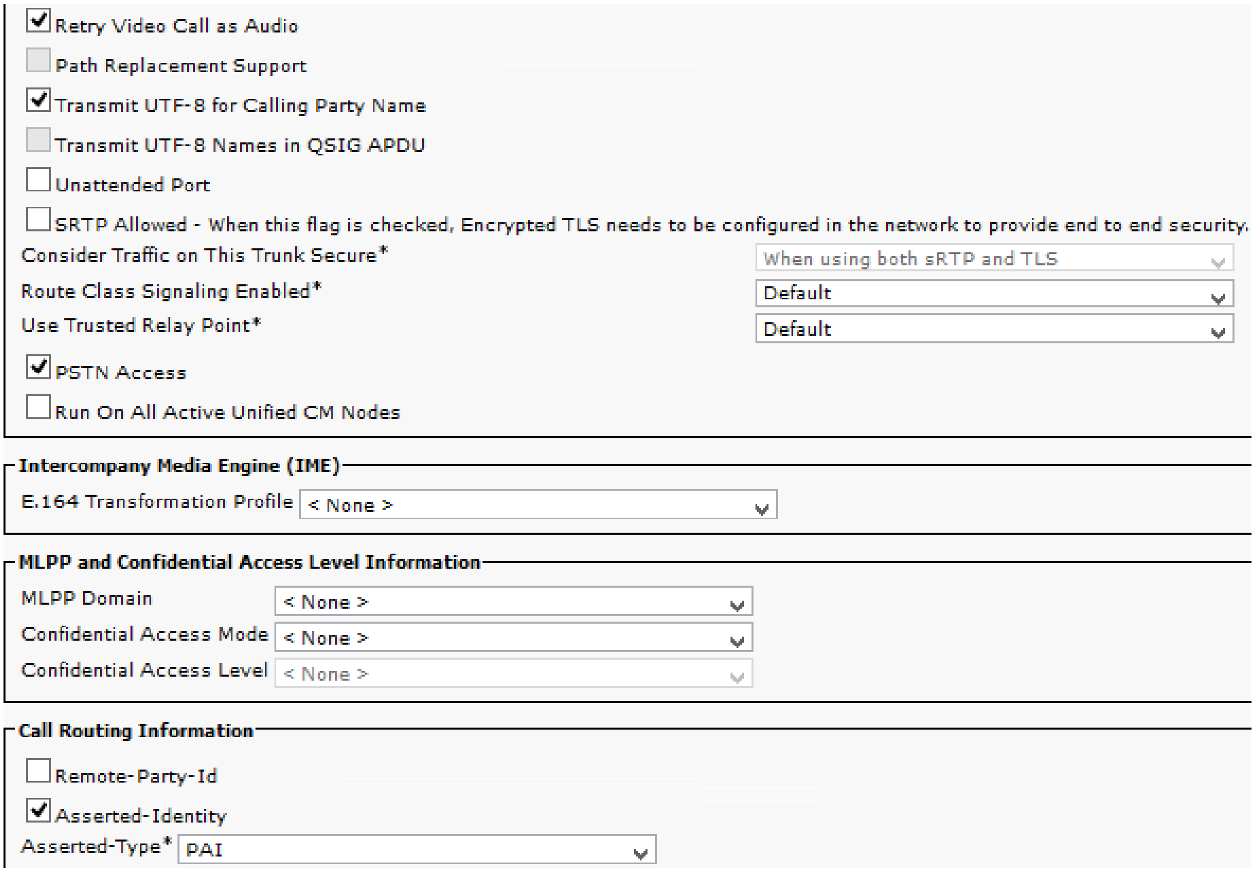
Clear Prefix Settings | Default Prefix Settings

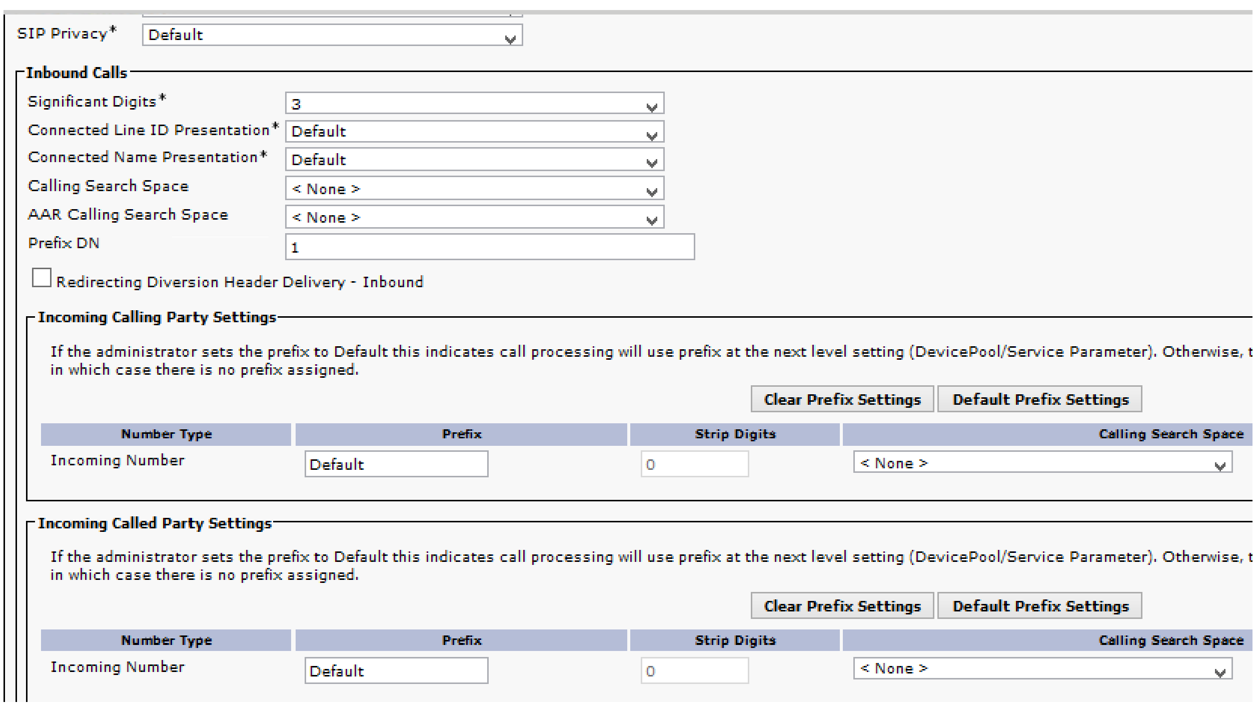
Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >



SIP Trunk Cisco Unified Communications Manager




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.
Consider Traffic on This Trunk Secure*
When using both sRTP and TLS
Route Class Signaling Enabled*
Default
Use Trusted Relay Point*
Default
PSTN Access
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 >
Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type*
PAI


SIP Privacy*
Default
Inbound Calls
Significant Digits*
3
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Calling Search Space
< None >
AAR Calling Search Space
< None >
Prefix DN
1
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, t
in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
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, t
in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
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
1*	[redacted]		5060

MTP Preferred Originating Codec*: 711ulaw

BLF Presence Group*: Standard Presence group

SIP Trunk Security Profile*: Non Secure SIP Trunk Profile

Rerouting Calling Search Space: < None >

Out-Of-Dialog Refer Calling Search Space: < None >

SUBSCRIBE Calling Search Space: < None >

SIP Profile*: Standard SIP Profile [View Details](#)

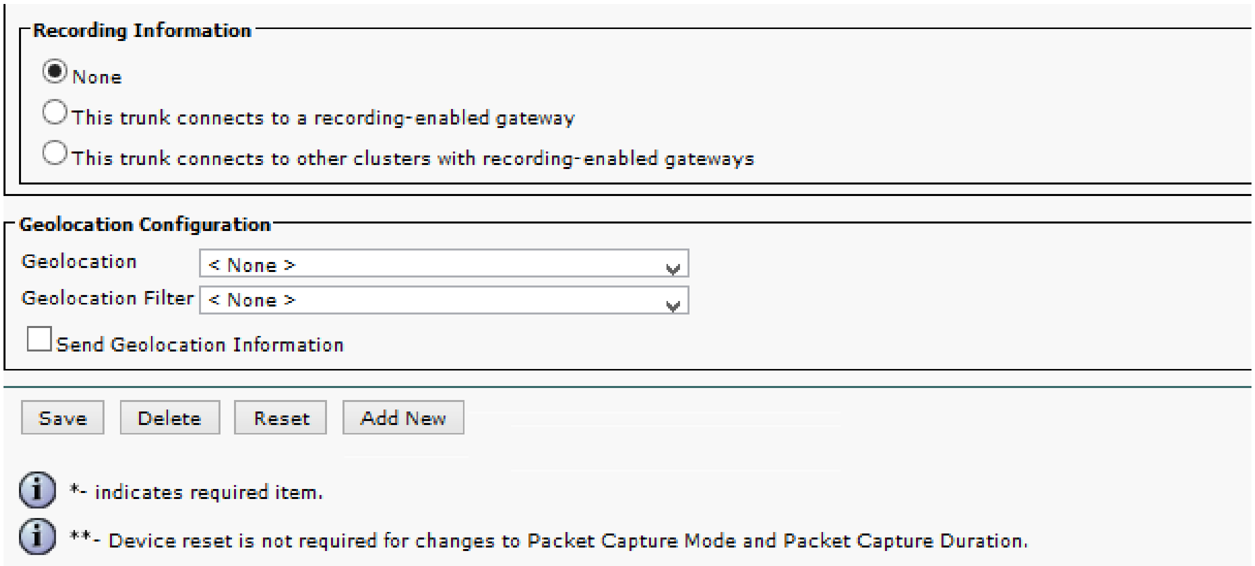
DTMF Signaling Method*: OOB and RFC 2833

Normalization Script

Normalization Script: < None >

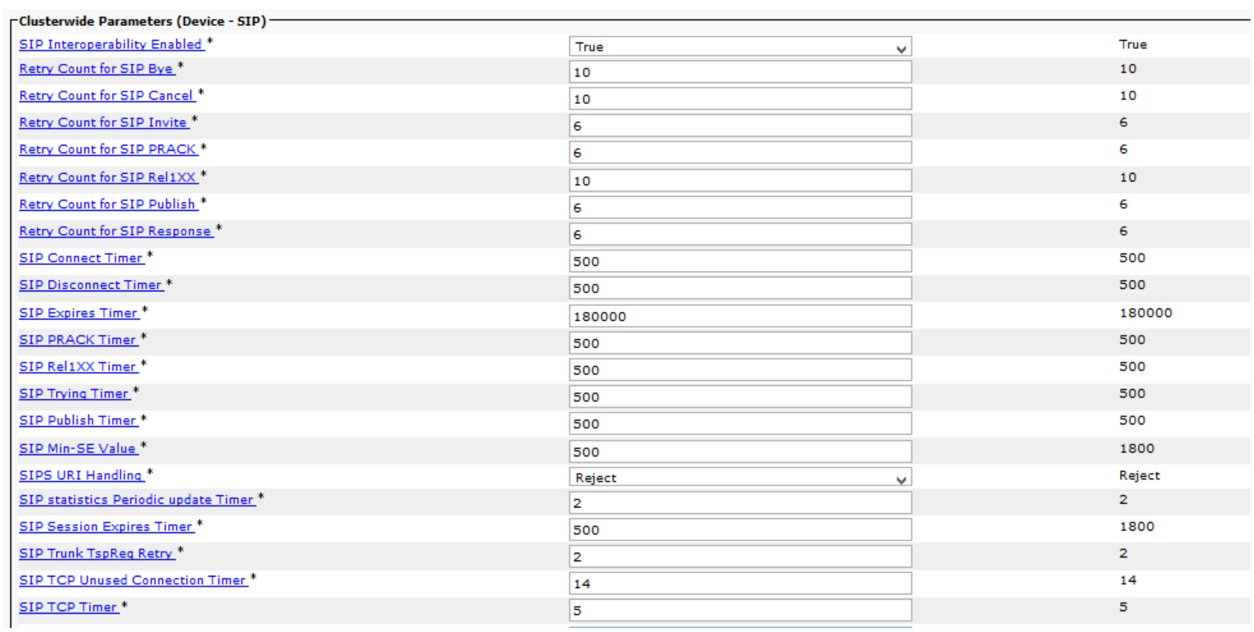
☐ Enable Trace

	Parameter Name	Parameter Value
1		



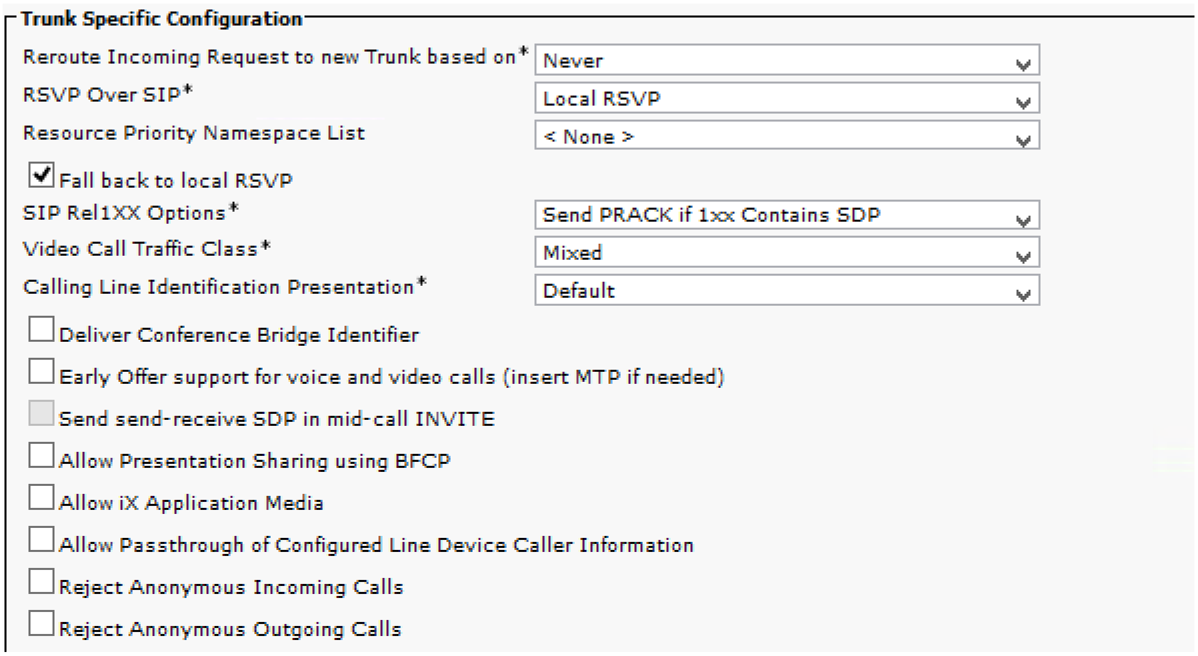
SIP PRACK for early-media negotiation

Telus Mobility requirement for compatibility with Cisco CUCM is to modify the service parameters related to two timers and enable the PRACK on SIP trunk profile in order to acknowledge SDP messages



Clusterwide Parameters (Device - SIP)

SIP Interoperability Enabled *	True	True
Retry Count for SIP Bye *	10	10
Retry Count for SIP Cancel *	10	10
Retry Count for SIP Invite *	6	6
Retry Count for SIP PRACK *	6	6
Retry Count for SIP Rel1XX *	10	10
Retry Count for SIP Publish *	6	6
Retry Count for SIP Response *	6	6
SIP Connect Timer *	500	500
SIP Disconnect Timer *	500	500
SIP Expires Timer *	180000	180000
SIP PRACK Timer *	500	500
SIP Rel1XX Timer *	500	500
SIP Trying Timer *	500	500
SIP Publish Timer *	500	500
SIP Min-SE Value *	500	1800
SIPS URI Handling *	Reject	Reject
SIP statistics Periodic update Timer *	2	2
SIP Session Expires Timer *	500	1800
SIP Trunk TspReq Retry *	2	2
SIP TCP Unused Connection Timer *	14	14
SIP TCP Timer *	5	5


Trunk Specific Configuration

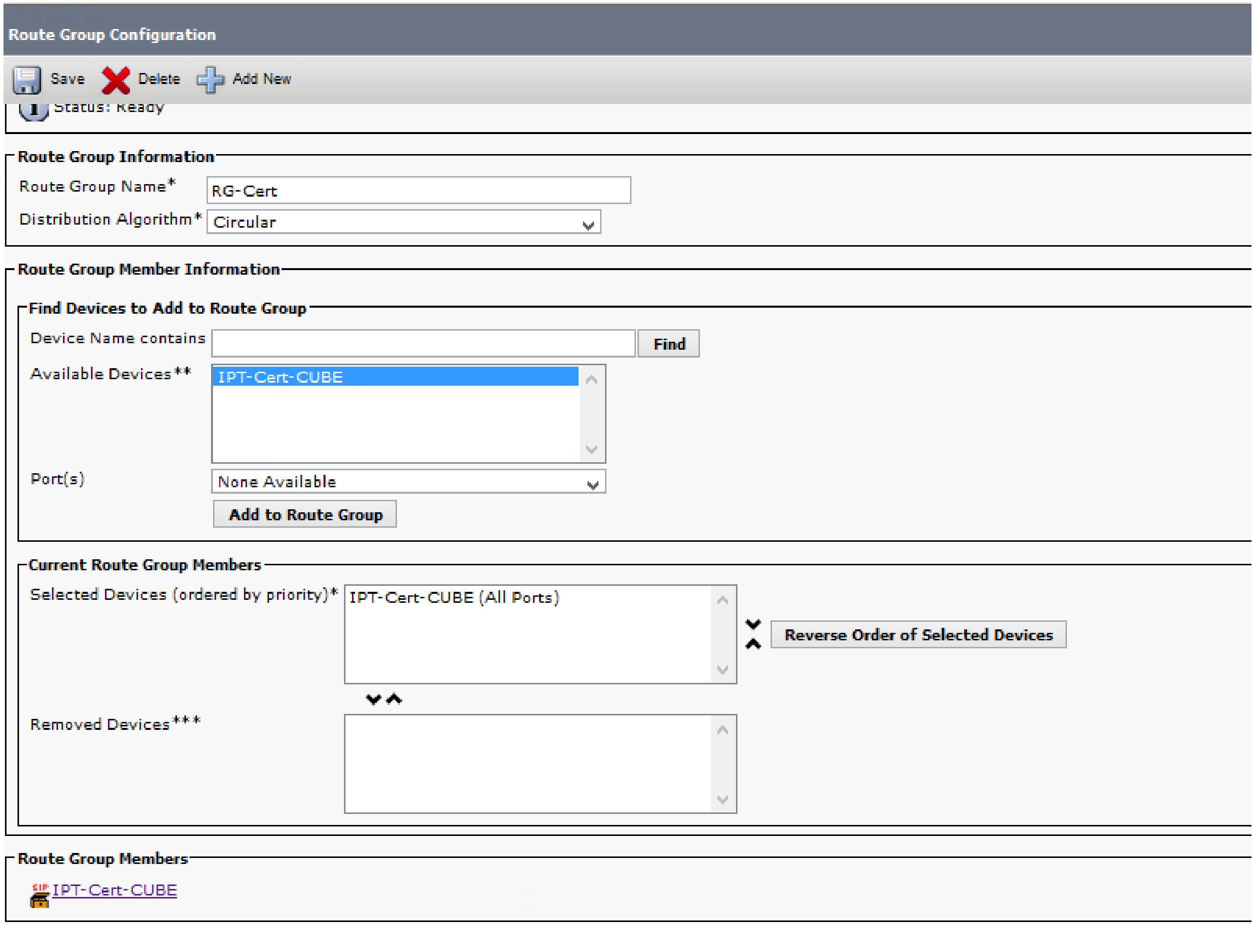
Reroute Incoming Request to new Trunk based on*	Never
RSVP Over SIP*	Local RSVP
Resource Priority Namespace List	< None >

- [x] Fall back to local RSVP

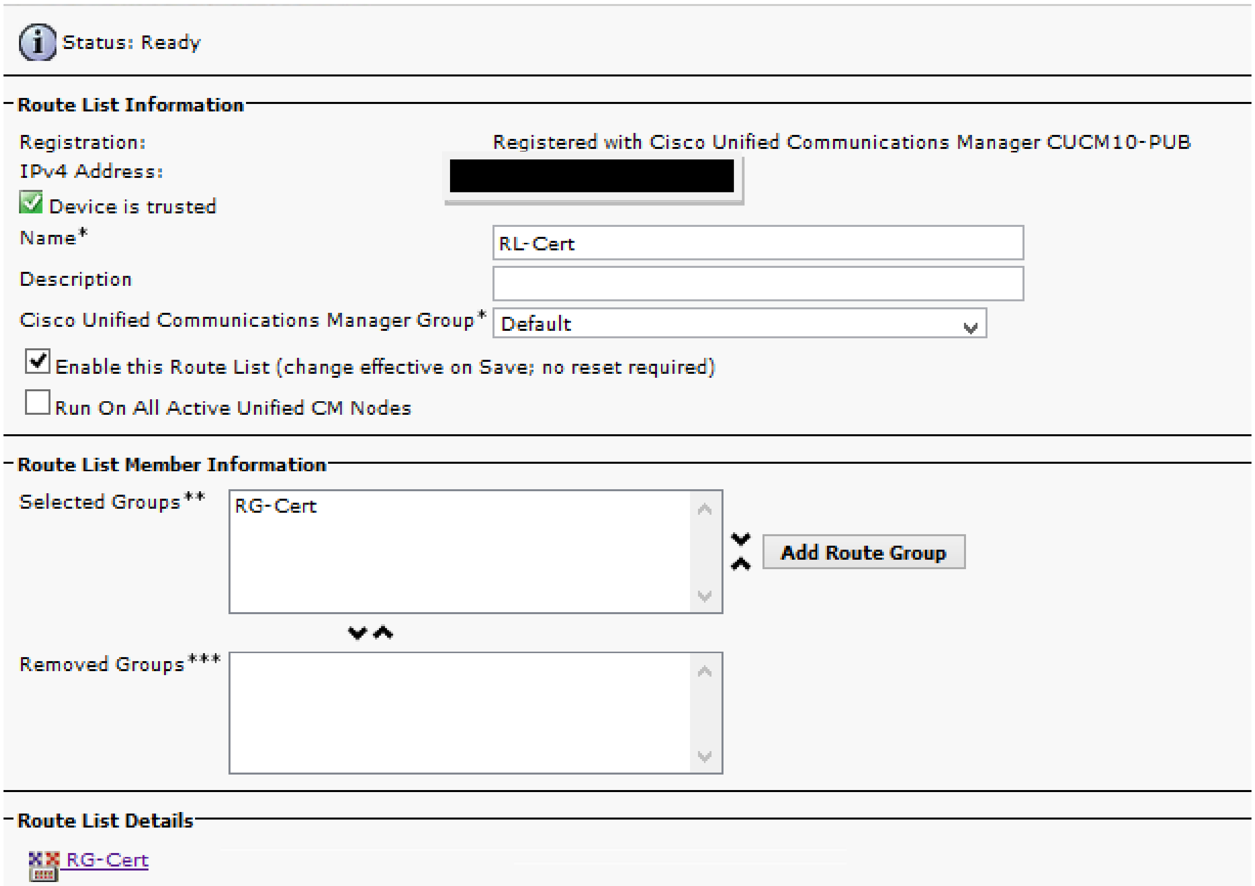
SIP Rel1XX Options*	Send PRACK if 1xx Contains SDP
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default

- [ ] Deliver Conference Bridge Identifier
- [ ] Early Offer support for voice and video calls (insert MTP if needed)
- [ ] Send send-receive SDP in mid-call INVITE
- [ ] Allow Presentation Sharing using BFCP
- [ ] Allow iX Application Media
- [ ] Allow Passthrough of Configured Line Device Caller Information
- [ ] Reject Anonymous Incoming Calls
- [ ] Reject Anonymous Outgoing Calls

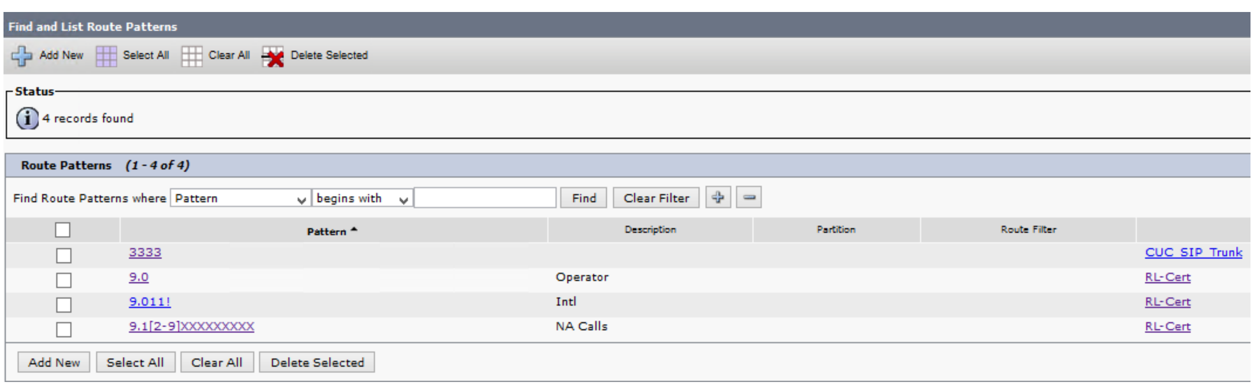
Route Group (SIP Trunk)



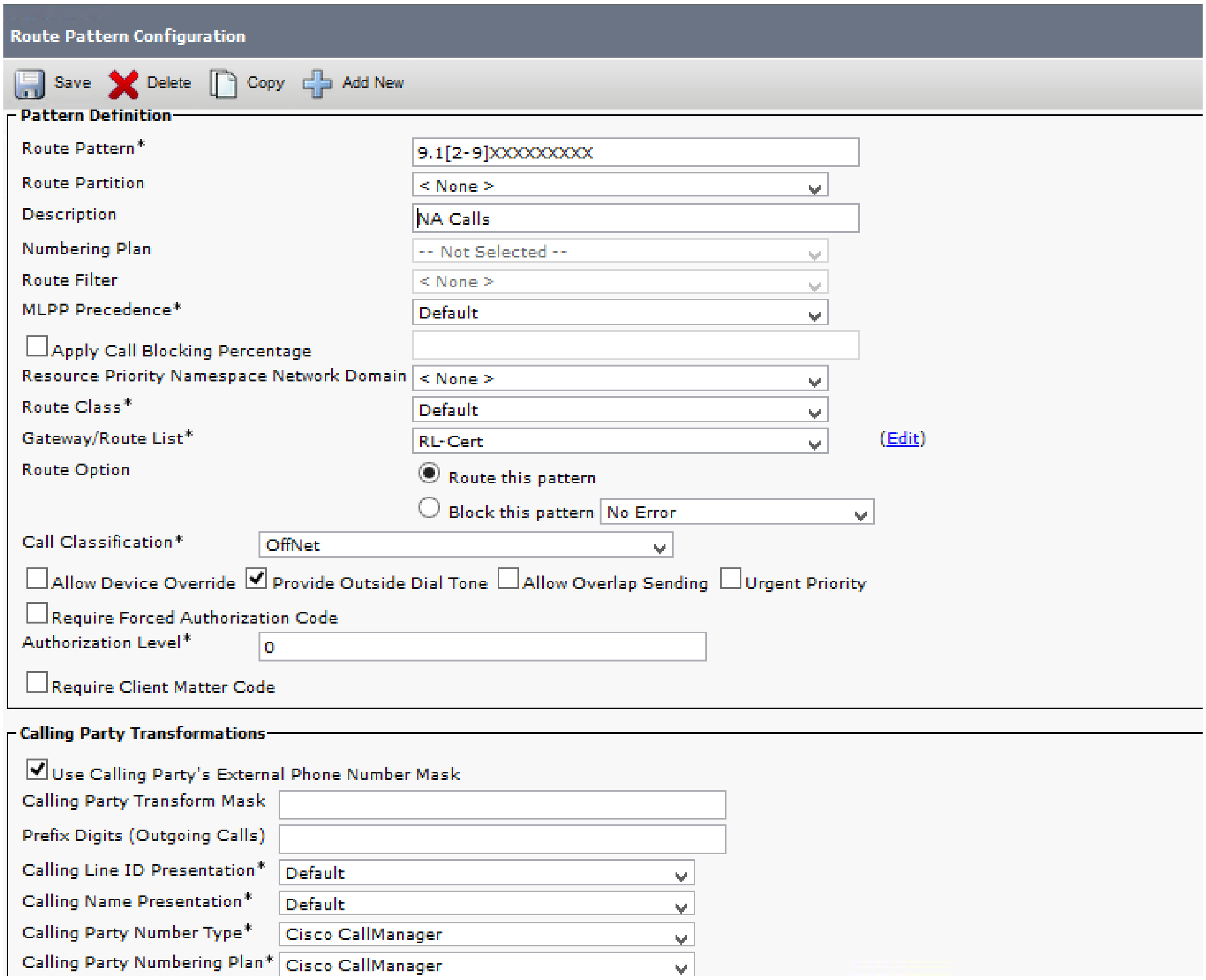
Route List (SIP Trunk)

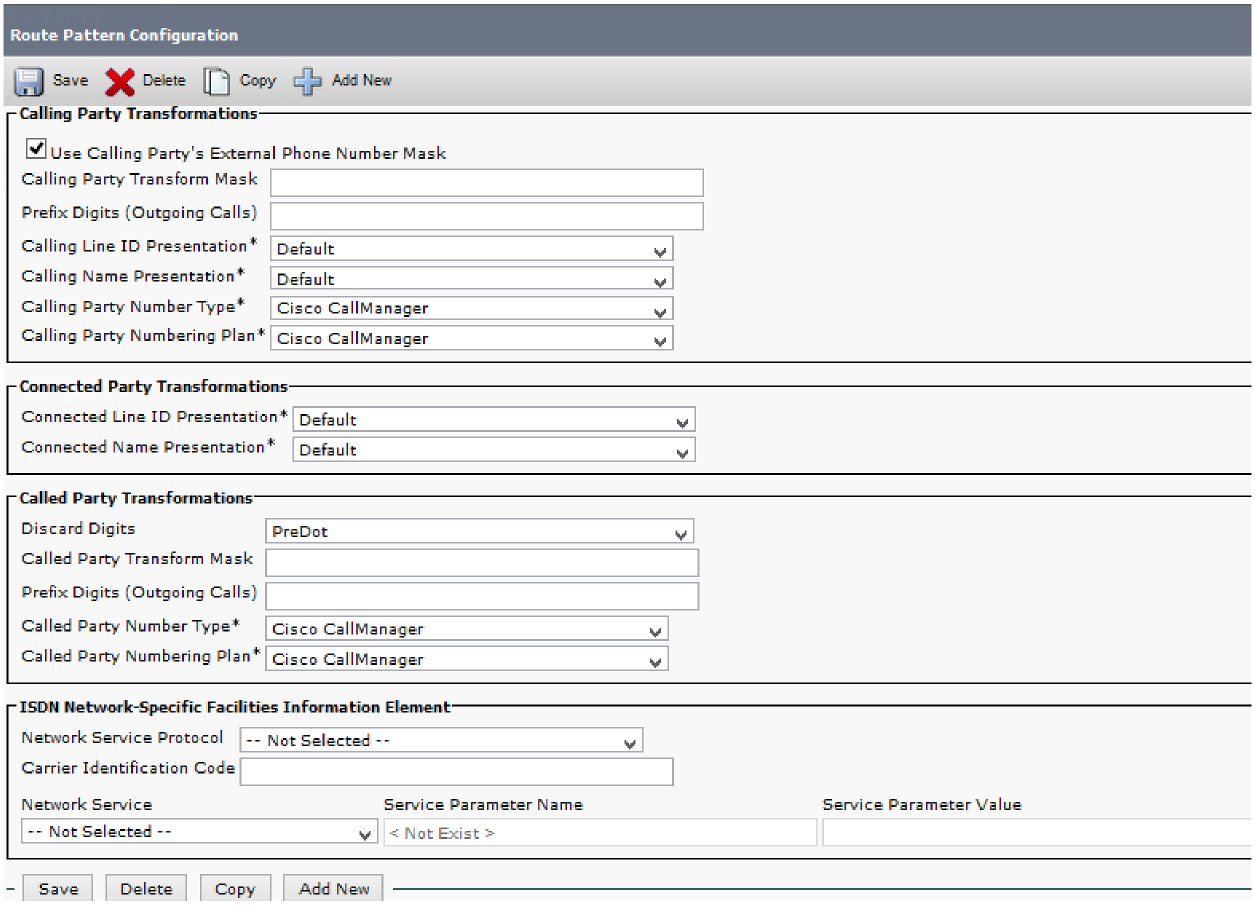


Route Pattern (SIP Trunk)

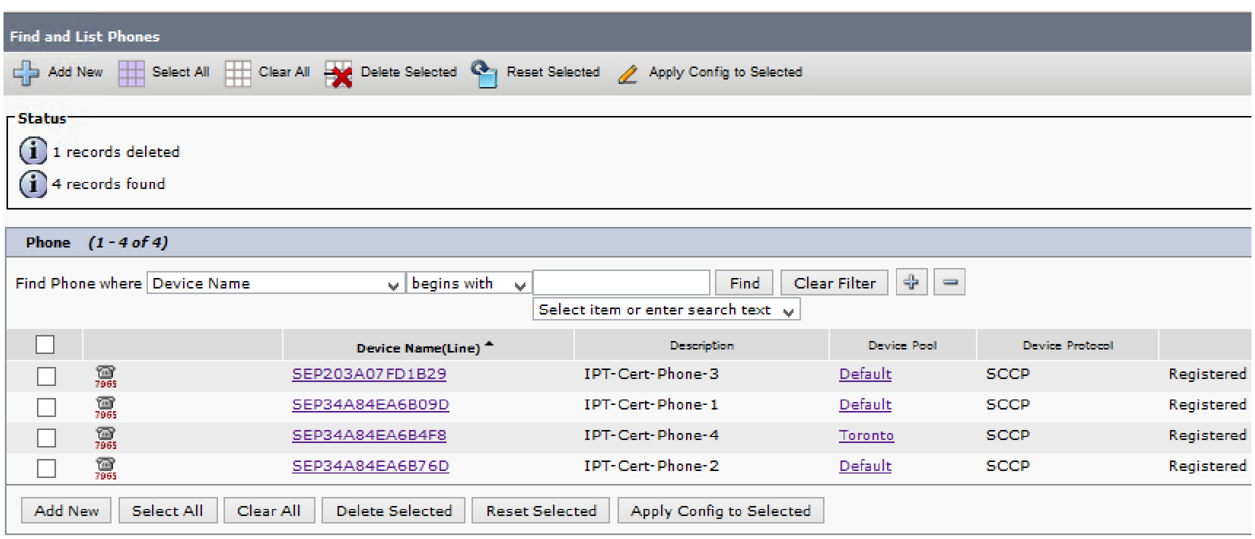


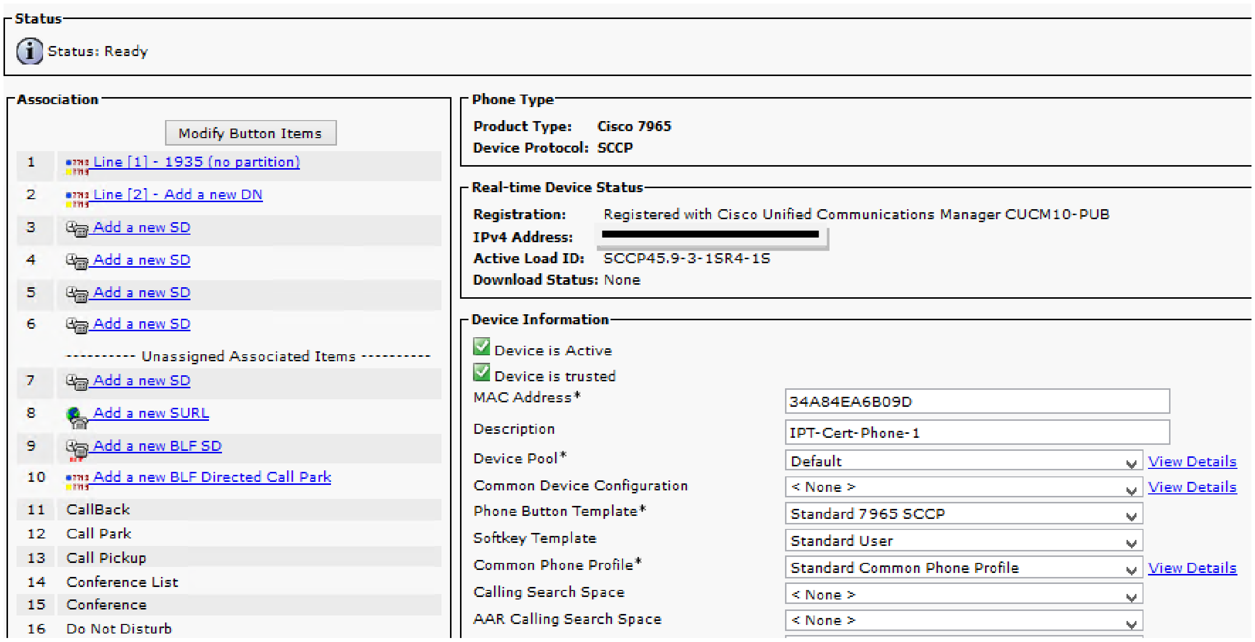
Please note that Digits 9. Predot will be discarded on all patterns on the outgoing SIP trunk. 9 was used only for route selection.

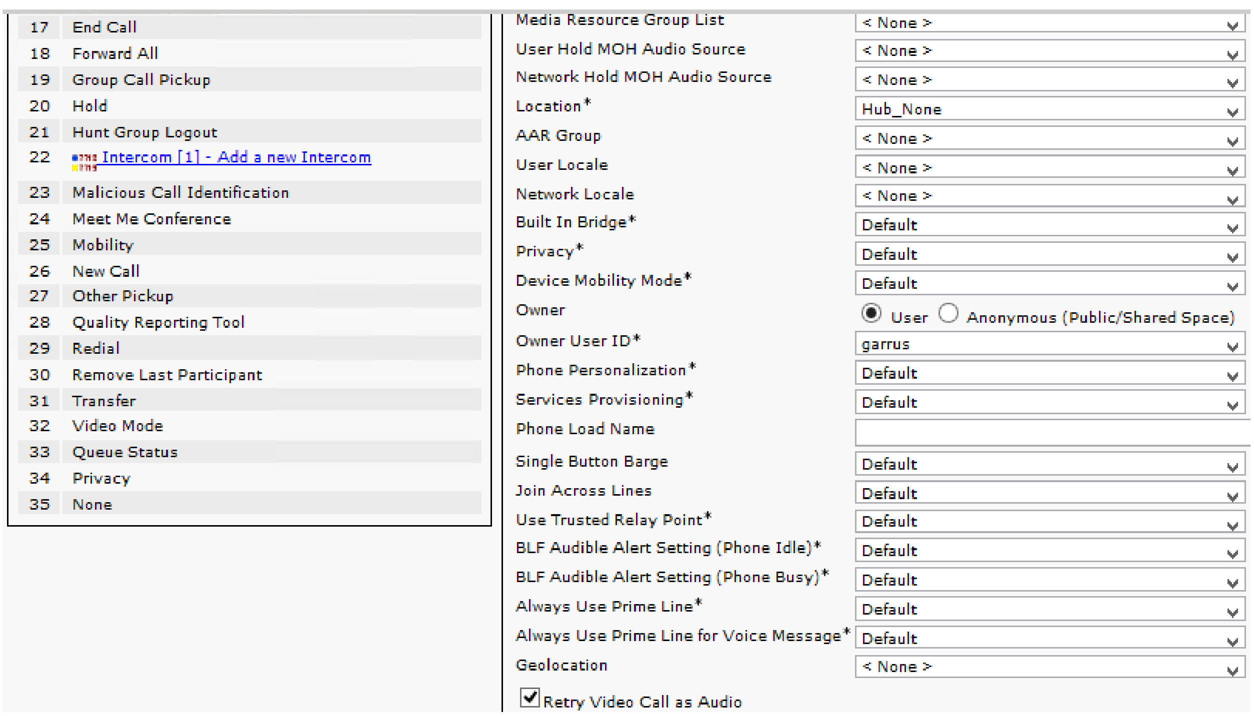

Route Pattern Configuration
Save
Delete
Copy
Add New
Pattern Definition
Route Pattern*
9.1[2-9]XXXXXXXXX
Route Partition
< None >
Description
NA Calls
Numbering Plan
-- Not Selected --
Route Filter
< None >
MLPP Precedence*
Default
Apply Call Blocking Percentage
Resource Priority Namespace Network Domain
< None >
Route Class*
Default
Gateway/Route List*
RL-Cert
(Edit)
Route Option
Route this pattern
Block this pattern
No Error
Call Classification*
OffNet
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

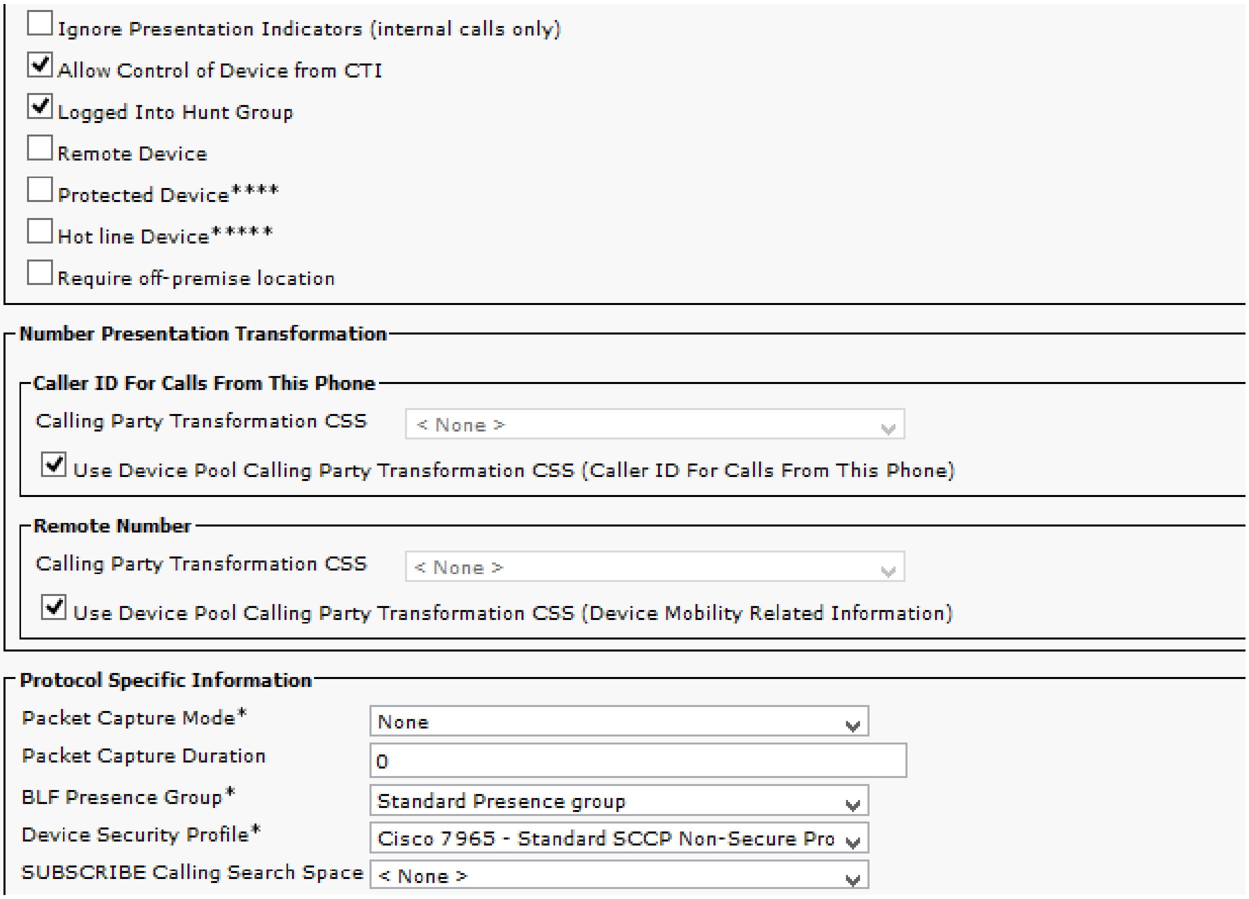


IP phone configuration




Status
Status: Ready
Association
Modify Button Items
1 Line [1] - 1935 (no partition)
2 Line [2] - Add a new DN
3 Add a new SD
4 Add a new SD
5 Add a new SD
6 Add a new SD
---------- Unassigned Associated Items ----------
7 Add a new SD
8 Add a new SURL
9 Add a new BLF SD
10 Add a new BLF Directed Call Park
11 CallBack
12 Call Park
13 Call Pickup
14 Conference List
15 Conference
16 Do Not Disturb
Phone Type
Product Type: Cisco 7965
Device Protocol: SCCP
Real-time Device Status
Registration: Registered with Cisco Unified Communications Manager CUCM10-PUB
IPv4 Address: [redacted]
Active Load ID: SCCP45.9-3-1SR4-1S
Download Status: None
Device Information
Device is Active
Device is trusted
MAC Address* 34A84EA6B09D
Description IPT-Cert-Phone-1
Device Pool* Default View Details
Common Device Configuration < None > View Details
Phone Button Template* Standard 7965 SCCP
Softkey Template Standard User
Common Phone Profile* Standard Common Phone Profile View Details
Calling Search Space < None >
AAR Calling Search Space < None >


17 End Call
18 Forward All
19 Group Call Pickup
20 Hold
21 Hunt Group Logout
22 Intercom [1] - Add a new Intercom
23 Malicious Call Identification
24 Meet Me Conference
25 Mobility
26 New Call
27 Other Pickup
28 Quality Reporting Tool
29 Redial
30 Remove Last Participant
31 Transfer
32 Video Mode
33 Queue Status
34 Privacy
35 None
Media Resource Group List < None >
User Hold MOH Audio Source < None >
Network Hold MOH Audio Source < None >
Location* Hub_None
AAR Group < None >
User Locale < None >
Network Locale < None >
Built In Bridge* Default
Privacy* Default
Device Mobility Mode* Default
Owner User Anonymous (Public/Shared Space)
Owner User ID* garrus
Phone Personalization* Default
Services Provisioning* Default
Phone Load Name
Single Button Barge Default
Join Across Lines Default
Use Trusted Relay Point* Default
BLF Audible Alert Setting (Phone Idle)* Default
BLF Audible Alert Setting (Phone Busy)* Default
Always Use Prime Line* Default
Always Use Prime Line for Voice Message* Default
Geolocation < None >
Retry Video Call as Audio


Ignore Presentation Indicators (internal calls only)
Allow Control of Device from CTI
Logged Into Hunt Group
Remote Device
Protected Device****
Hot line Device*****
Require off-premise location
Number Presentation Transformation
Caller ID For Calls From This Phone
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)
Remote Number
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)
Protocol Specific Information
Packet Capture Mode*
None
Packet Capture Duration
0
BLF Presence Group*
Standard Presence group
Device Security Profile*
Cisco 7965 - Standard SCCP Non-Secure Pro
SUBSCRIBE Calling Search Space
< None >

☐ Unattended Port
☐ Require DTMF Reception
☐ RFC2833 Disabled

Certification Authority Proxy Function (CAPF) Information

Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
Generate String	
Key Size (Bits)*	1024
Operation Completes By	2014 9 30 12 (YYYY:MM:DD:HH)

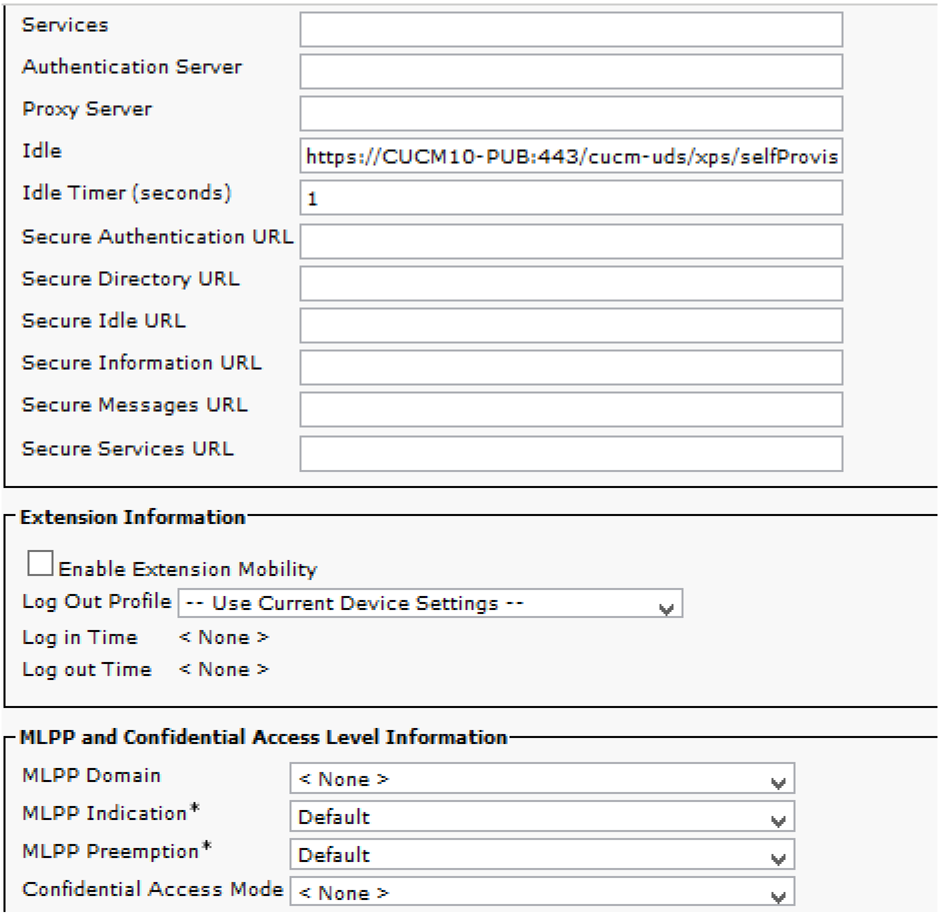
Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.

Expansion Module Information

Module 1	< None >
Module 1 Load Name	
Module 2	< None >
Module 2 Load Name	

External Data Locations Information (Leave blank to use default)

Information	
Directory	
Messages	



Services	
Authentication Server	
Proxy Server	
Idle	https://CUCM10-PUB:443/cucm-uds/xps/selfProvis
Idle Timer (seconds)	1
Secure Authentication URL	
Secure Directory URL	
Secure Idle URL	
Secure Information URL	
Secure Messages URL	
Secure Services URL	

Extension Information

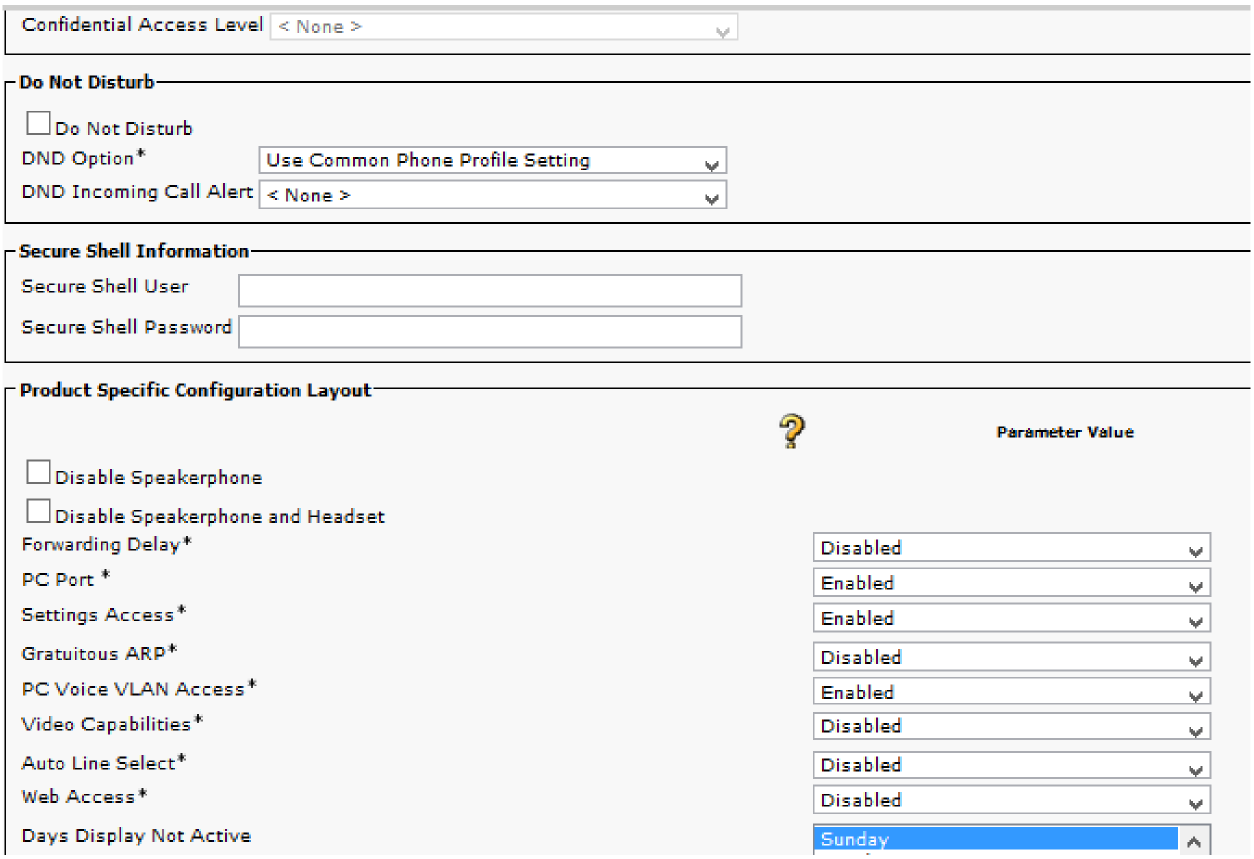
☐ Enable Extension Mobility

Log Out Profile	-- Use Current Device Settings --
Log in Time	< None >
Log out Time	< None >

MLPP and Confidential Access Level Information

MLPP Domain	< None >
MLPP Indication*	Default
MLPP Preemption*	Default
Confidential Access Mode	< None >

cisco

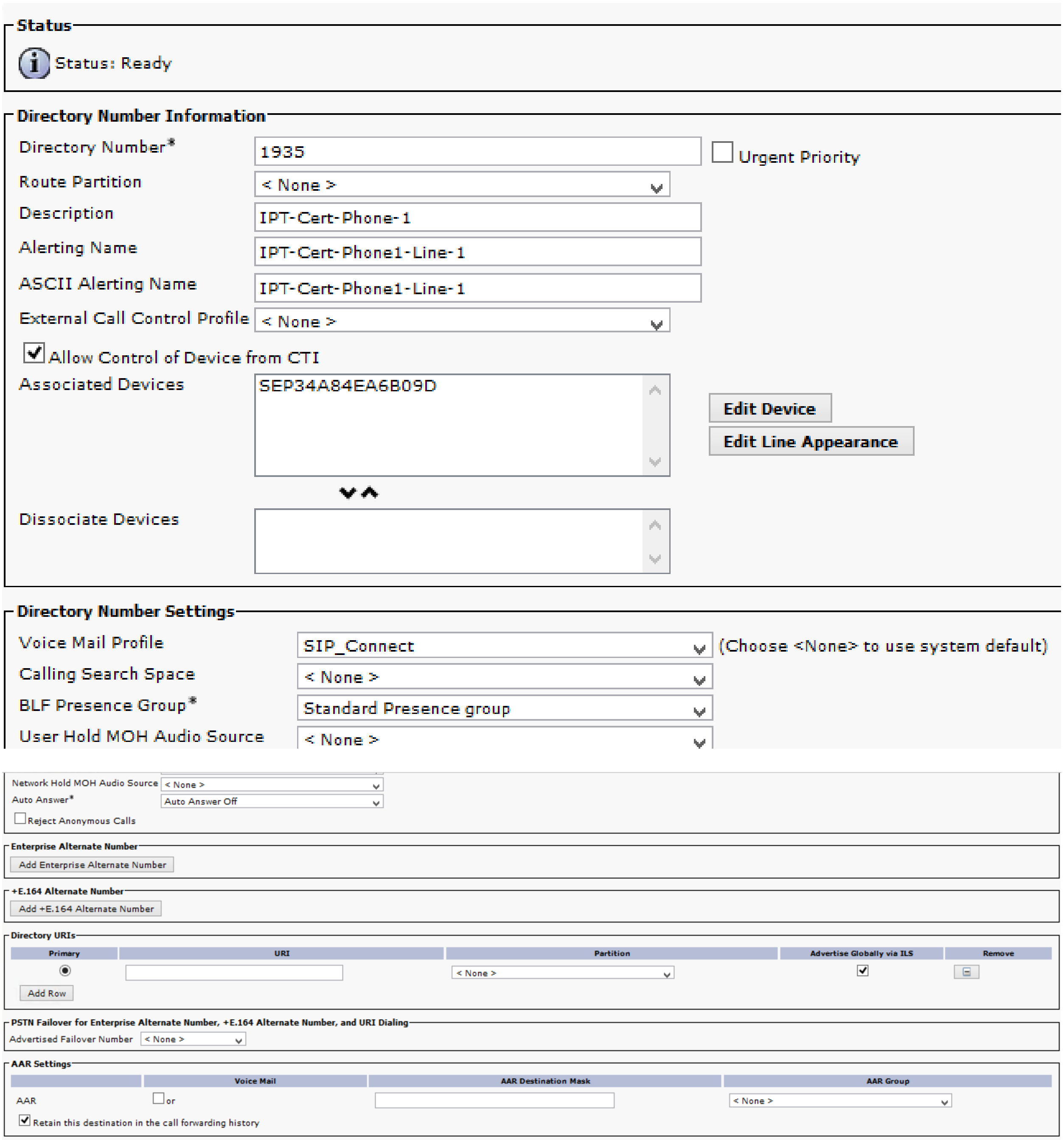

Confidential Access Level
< None >
Do Not Disturb
Do Not Disturb
DND Option*
Use Common Phone Profile Setting
DND Incoming Call Alert
< None >
Secure Shell Information
Secure Shell User
Secure Shell Password
Product Specific Configuration Layout
Parameter Value
Disable Speakerphone
Disable Speakerphone and Headset
Forwarding Delay*
Disabled
PC Port *
Enabled
Settings Access*
Enabled
Gratuitous ARP*
Disabled
PC Voice VLAN Access*
Enabled
Video Capabilities*
Disabled
Auto Line Select*
Disabled
Web Access*
Disabled
Days Display Not Active
Sunday

Display On Duration	10:30
Display Idle Timeout	01:00
Enable Power Save Plus	Sunday Monday Tuesday
Phone On Time	00:00
Phone Off Time	24:00
Phone Off Idle Timeout*	60
☐ Enable Audible Alert	
EnergyWise Domain	
EnergyWise Endpoint Security Secret	
☐ Allow EnergyWise Overrides	
Span to PC Port*	Disabled
Logging Display*	PC Controlled
Load Server	
Recording Tone*	Disabled
Recording Tone Local Volume*	100
Recording Tone Remote Volume*	50
Recording Tone Duration	
Display On When Incoming Call*	Disabled
RTCP*	Disabled
"more" Soft Key Timer	5
Auto Call Select*	Enabled

Log Server	
Advertise G.722 Codec*	Use System Default
Wideband Headset UI Control*	Enabled
Wideband Headset*	Enabled
Peer Firmware Sharing*	Enabled
Cisco Discovery Protocol (CDP): Switch Port*	Enabled
Cisco Discovery Protocol (CDP): PC Port*	Enabled
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled
LLDP Asset ID	
LLDP Power Priority*	Unknown
Wireless Headset Hookswitch Control*	Disabled
IPv6 Load Server	
IPv6 Log Server	
802.1x Authentication*	User Controlled
Detect Unified CM Connection Failure*	Normal
Minimum Ring Volume*	0-Silent
Headset Sidetone Level*	Default
Headset Send Gain*	Default
HTTPS Server*	http and https Enabled
Handset/Headset Monitor*	Enabled
Headset Recording*	Disabled

Enbloc Dialing*	Enabled
Switch Port Remote Configuration*	Disabled
PC Port Remote Configuration*	Disabled
Automatic Port Synchronization*	Disabled
SSH Access*	Disabled
LOGIN Access*	Enabled
FIPS Mode*	Disabled
80-bit SRTCP*	Disabled

IP phone DN configuration



Call Forward and Call Pickup Settings

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	☐ or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	☐ or		< None >
Forward Busy External	☐ or		< None >
Forward No Answer Internal	☐ or		< None >
Forward No Answer External	☐ or		< None >
Forward No Coverage Internal	☐ or		< None >
Forward No Coverage External	☐ or		< None >
Forward on CTI Failure	☐ or		< None >
Forward Unregistered Internal	☐ or		< None >
Forward Unregistered External	☐ or		< None >

No Answer Ring Duration (seconds)

Call Pickup Group < None >

Park Monitoring

	Voice Mail	Destination
Park Monitoring Forward No Retrieve Destination External	☐ or	
Park Monitoring Forward No Retrieve Destination Internal	☐ or	

Park Monitoring Reversion Timer A blank value will use value set in Park Monito

MLPP Alternate Party And Confidential Access Level Settings

Target (Destination)	
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	
Confidential Access Mode	< None >
Confidential Access Level	< None >
Call Control Agent Profile	< None >

Line Settings for All Devices

Hold Reversion Ring Duration (seconds)		Setting the Hold Reversion Ring I
Hold Reversion Notification Interval (seconds)		Setting the Hold Reversion Notifi
Party Entrance Tone*	Default	

Line 1 on Device SEP34A84EA6B09D

Display (Caller ID) IPT-Cert-Phone1-Line-1 Display text for a line appearance is i
number, the person receiving a call may not see the proper identity of the caller.

ASCII Display (Caller ID)	IPT-Cert-Phone1-Line-1
Line Text Label	1935
External Phone Number Mask	+15877569935
Visual Message Waiting Indicator Policy*	Use System Policy
Audible Message Waiting Indicator Policy*	Default
Ring Setting (Phone Idle)*	Ring
Ring Setting (Phone Active)	Use System Default — Applies to this line when an
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default
Call Pickup Group Audio Alert Setting(Phone Active)	Use System Default
Recording Option*	Call Recording Disabled
Recording Profile	< None >
Recording Media Source*	Gateway Preferred
Monitoring Calling Search Space	< None >

- [x] Log Missed Calls

Multiple Call/Call Waiting Settings on Device SEP34A84EA6B09D

Note:The range to select the Max Number of calls is: 1-200

Maximum Number of Calls*	4
Busy Trigger*	2 (Less

Forwarded Call Information Display on Device SEP34A84EA6B09D

- [x] Caller Name
- [ ] Caller Number
- [ ] Redirected Number
- [x] Dialed Number

Users Associated with Line

Associate End Users

Save | Delete | Reset | Apply Config | Add New

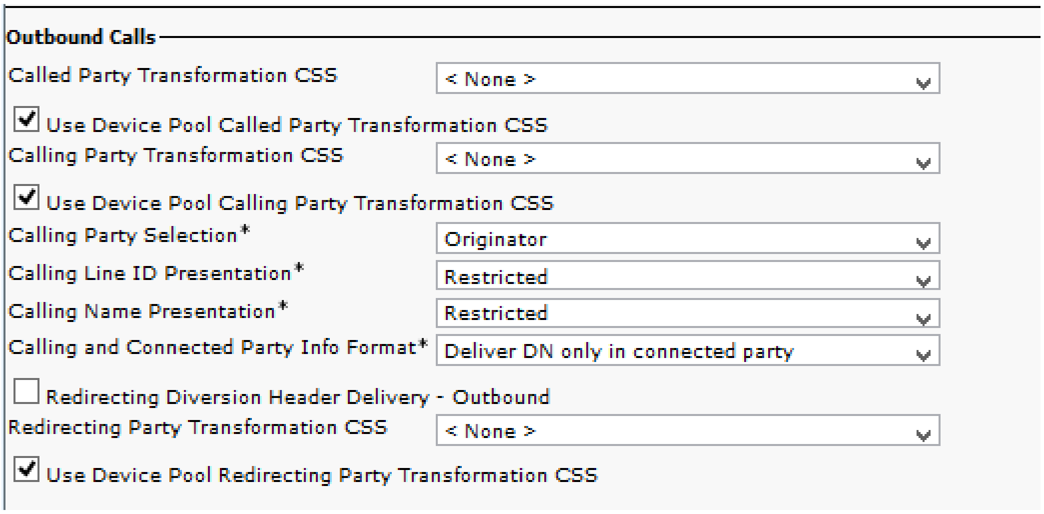
*- indicates required item.

**- Changes to Line or Directory Number settings require restart.

Privacy configuration

CUCM configuration privacy

Calling Line and Calling Name should be set to Restricted, this will enable the privacy: id



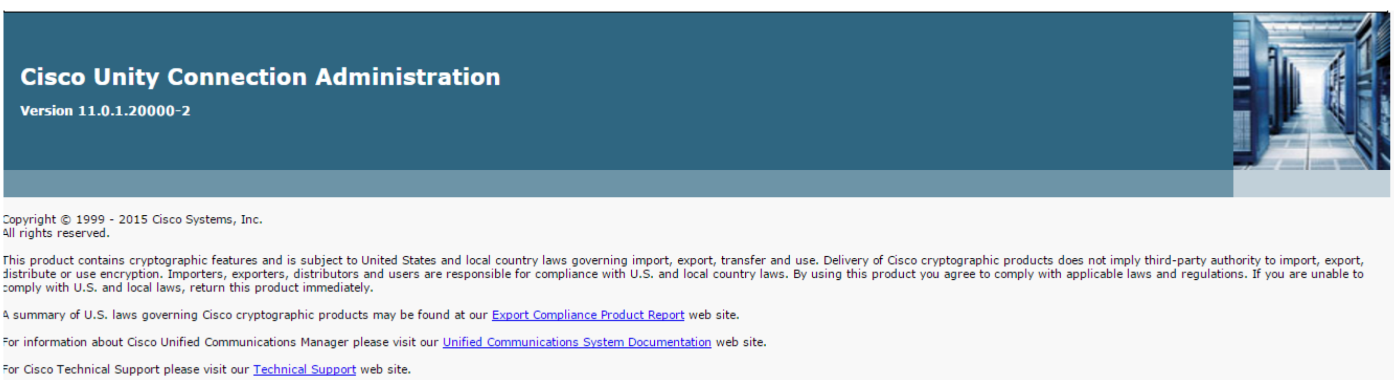
CUBE configuration privacy

SIP header should be modified in order to replace the anonymous with valid information as the SBC expect valid information in the INVITE header. If a valid header is not present then the SBC will use the Parent DN when the call is sent out to other switches (for tracking purposes). PAI header is being used for billing and routing on the SBC.

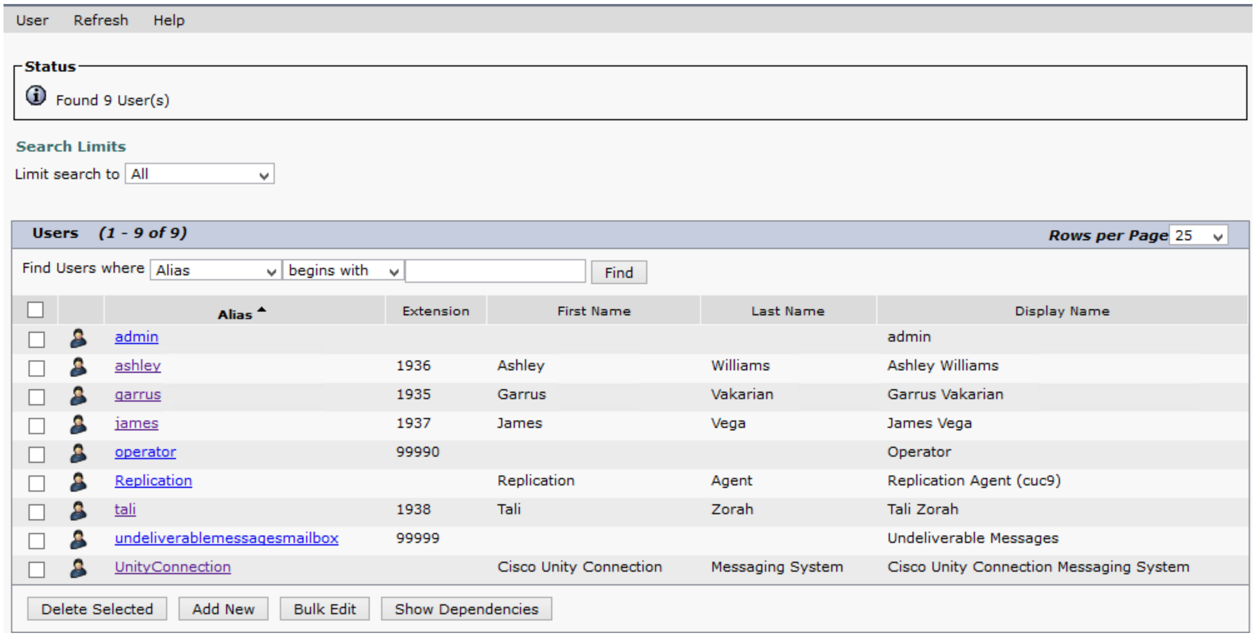
```
voice class sip-profiles 150
 request INVITE sip-header From modify "Anonymous" "IPT-Cert-Phone3-Line-1"
 request INVITE sip-header From modify "<sip:anonymous@anonymous.invalid>" "<sip:+15877569937@172.24.0.120>"
dial-peer voice 2000 voip
  voice-class sip profiles 150
```

Configuring the Cisco Unity Connection

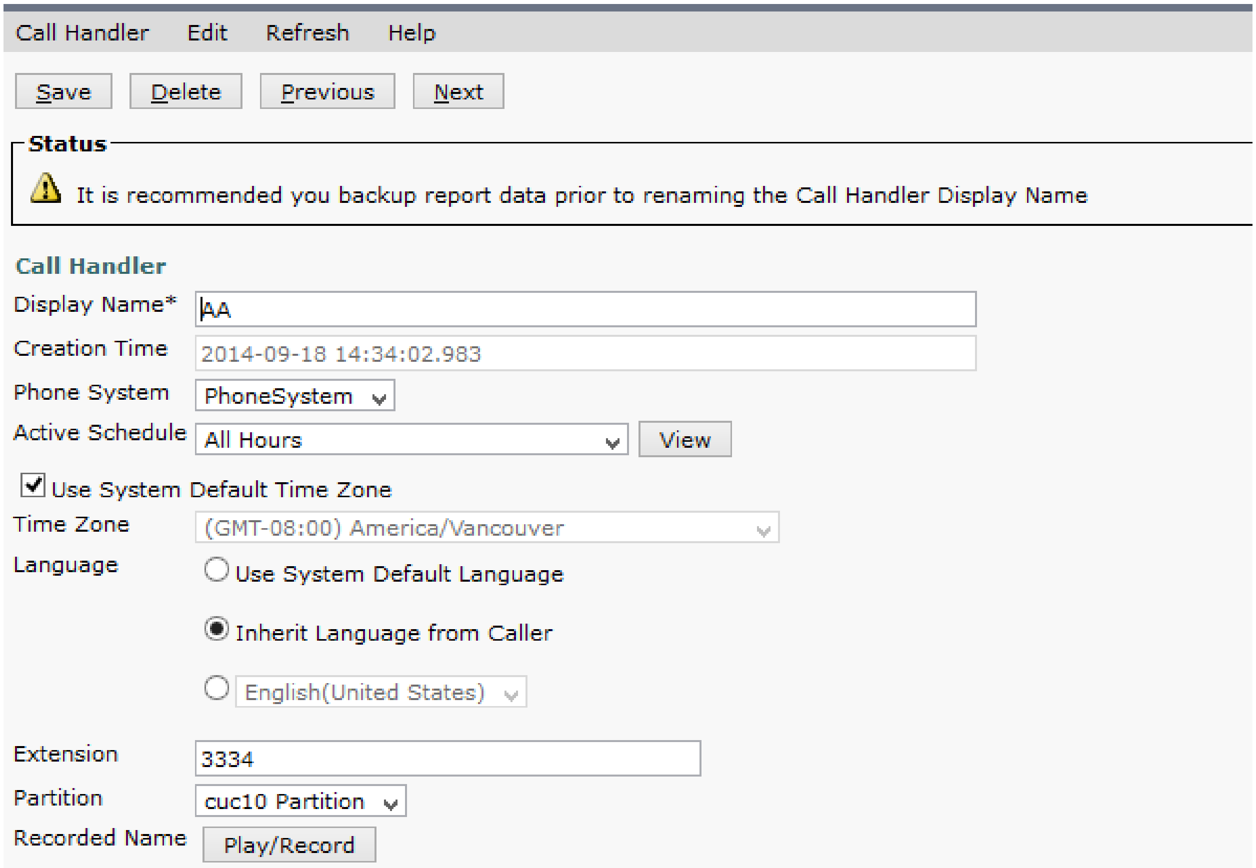
System Version



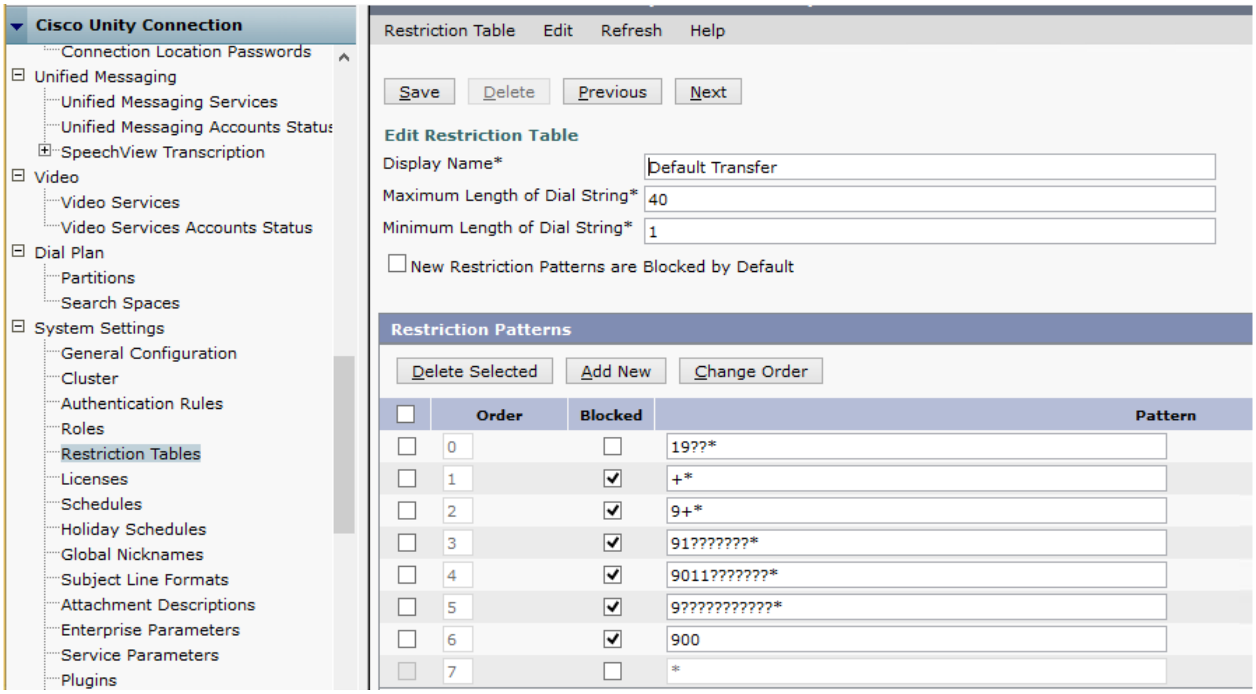
User configuration



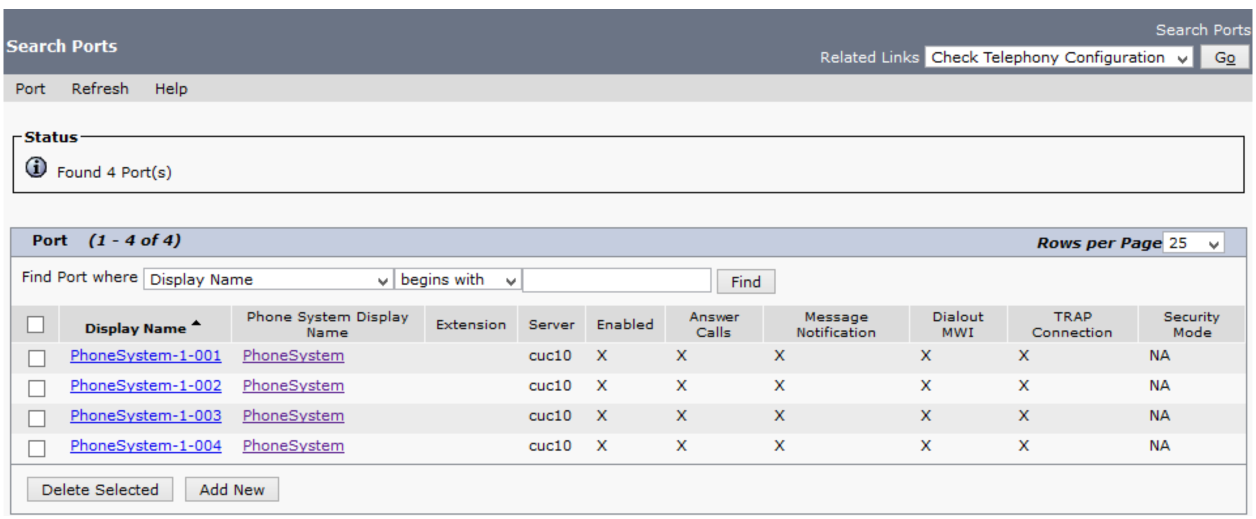
Call Handler for Auto Attendant



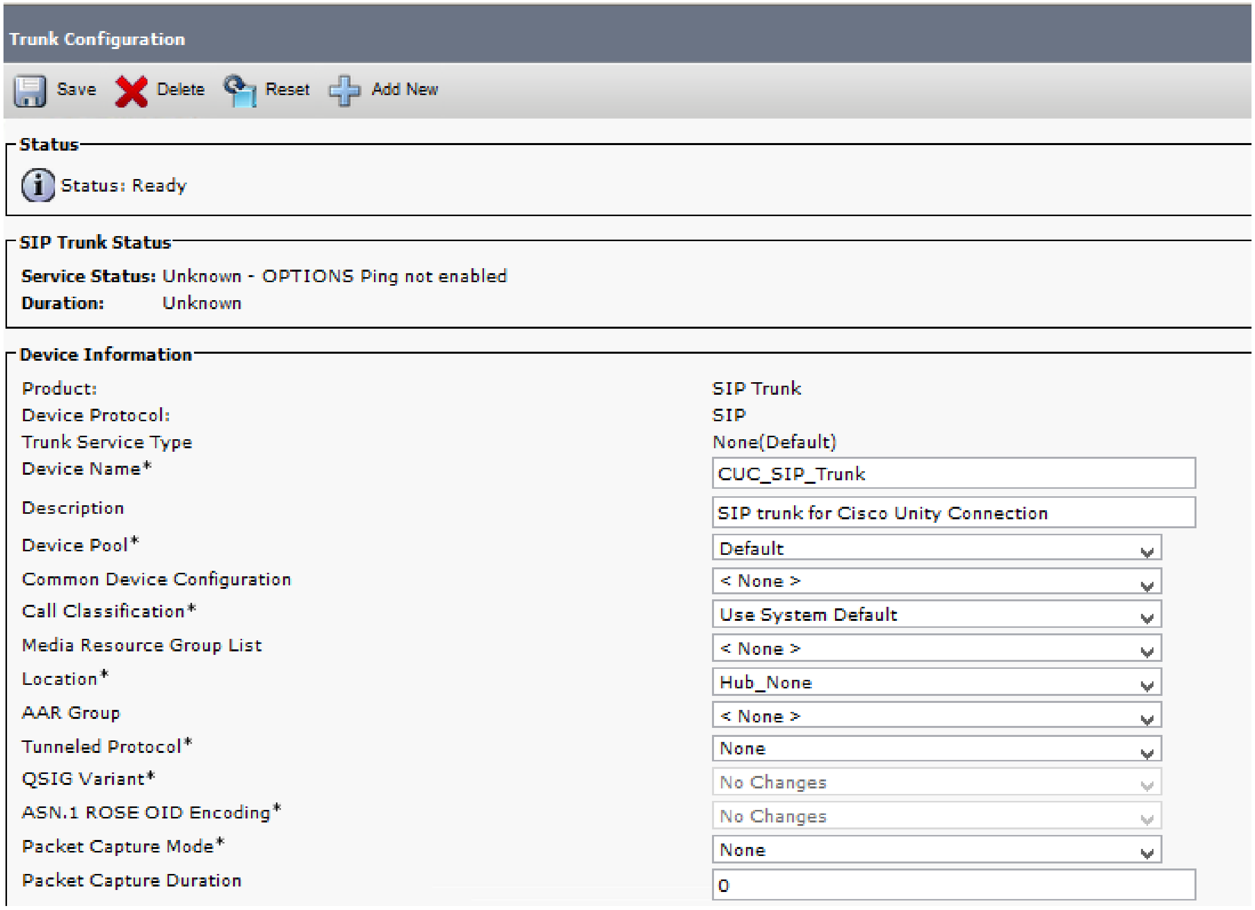
Call Routing for user extension



Cisco Unity Connection Ports



Cisco Unity Connection SIP Trunk



Trunk Configuration

Save | Delete | Reset | Add New

Status

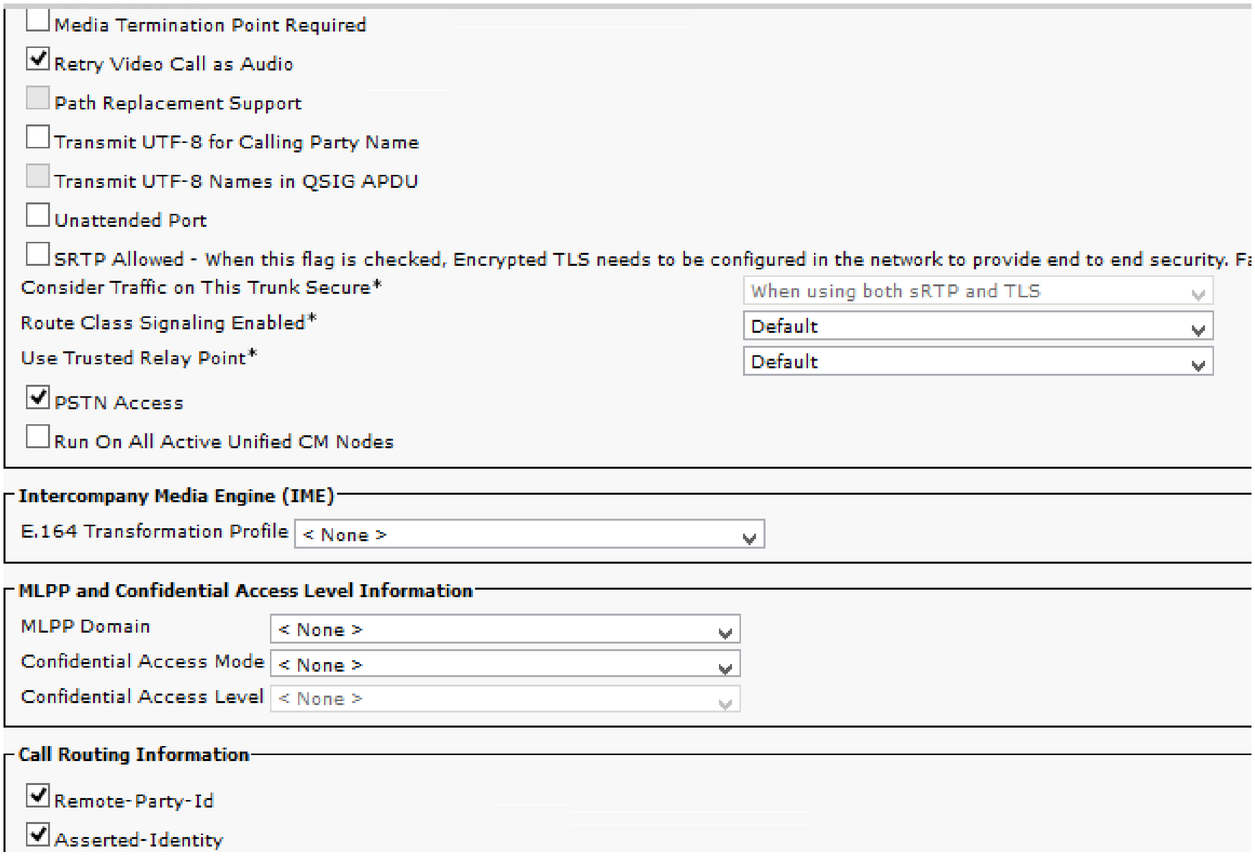
Status: Ready

SIP Trunk Status

Service Status: Unknown - OPTIONS Ping not enabled
Duration: Unknown

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	CUC_SIP_Trunk
Description	SIP trunk for Cisco Unity Connection
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
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. F

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
- [ ] Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

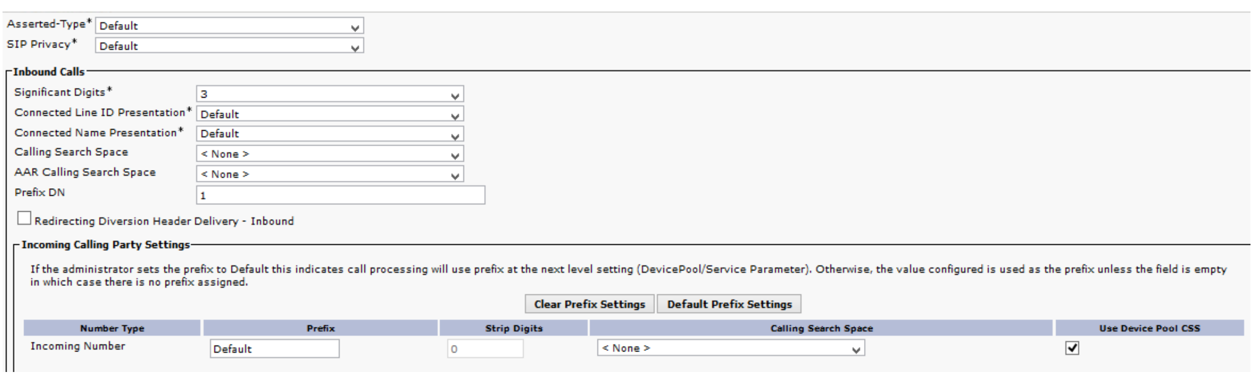
E.164 Transformation Profile	< None >

MLPP and Confidential Access Level Information

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

Call Routing Information

- [x] Remote-Party-Id
- [x] Asserted-Identity



Asserted-Type*	Default
SIP Privacy*	Default

Inbound Calls

Significant Digits*	3
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	1

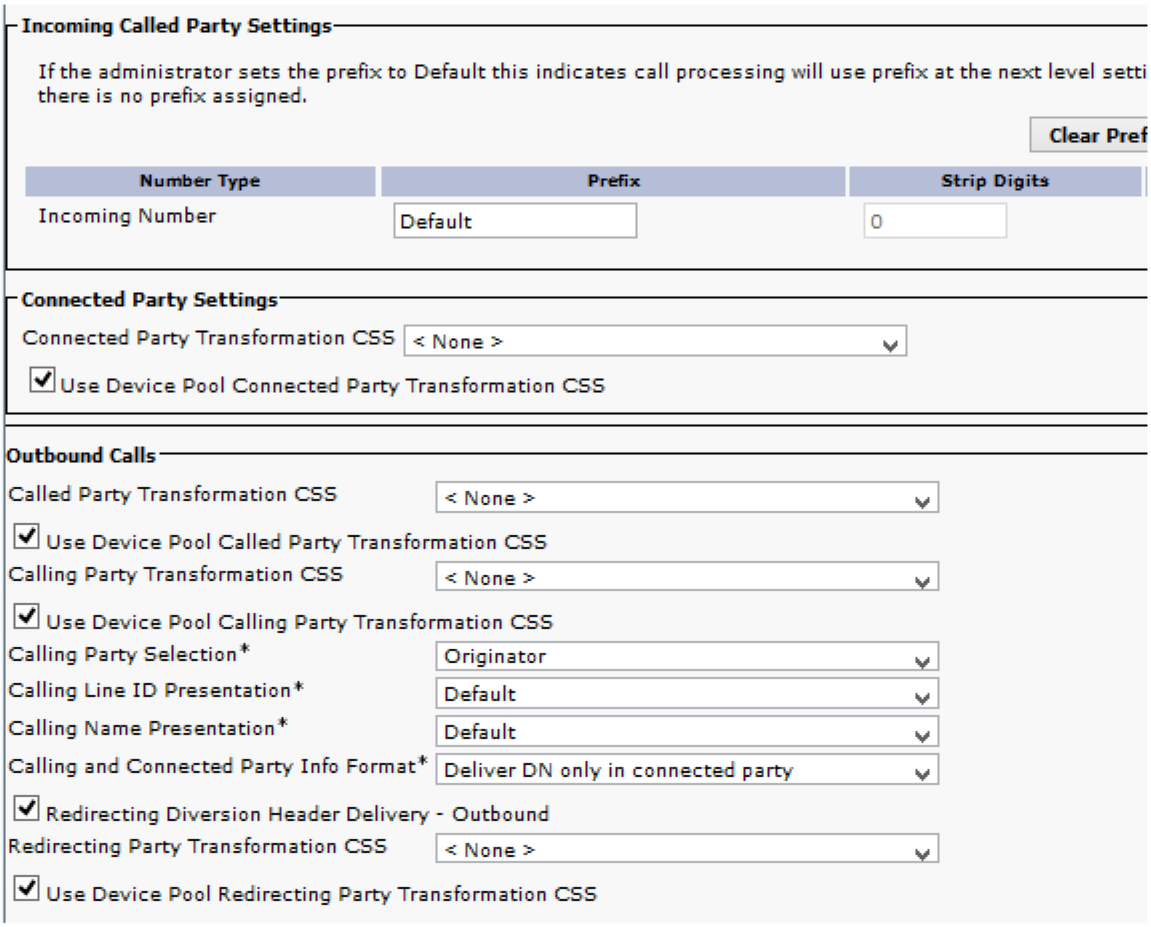
- [ ] Redirecting Diversion Header Delivery - Inbound

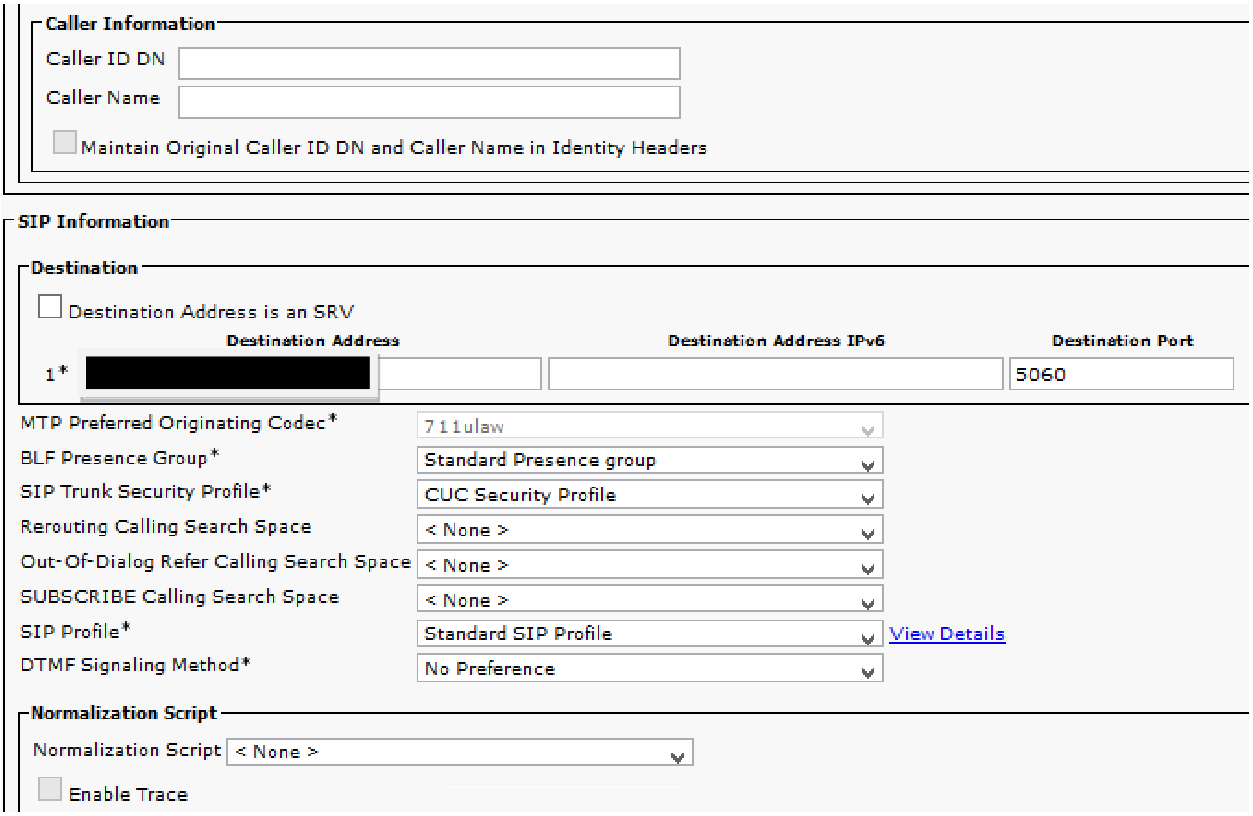
Incoming Calling Party Settings

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 setti
there is no prefix assigned.
Clear Pref
Number Type
Prefix
Strip Digits
Incoming Number
Default
0
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

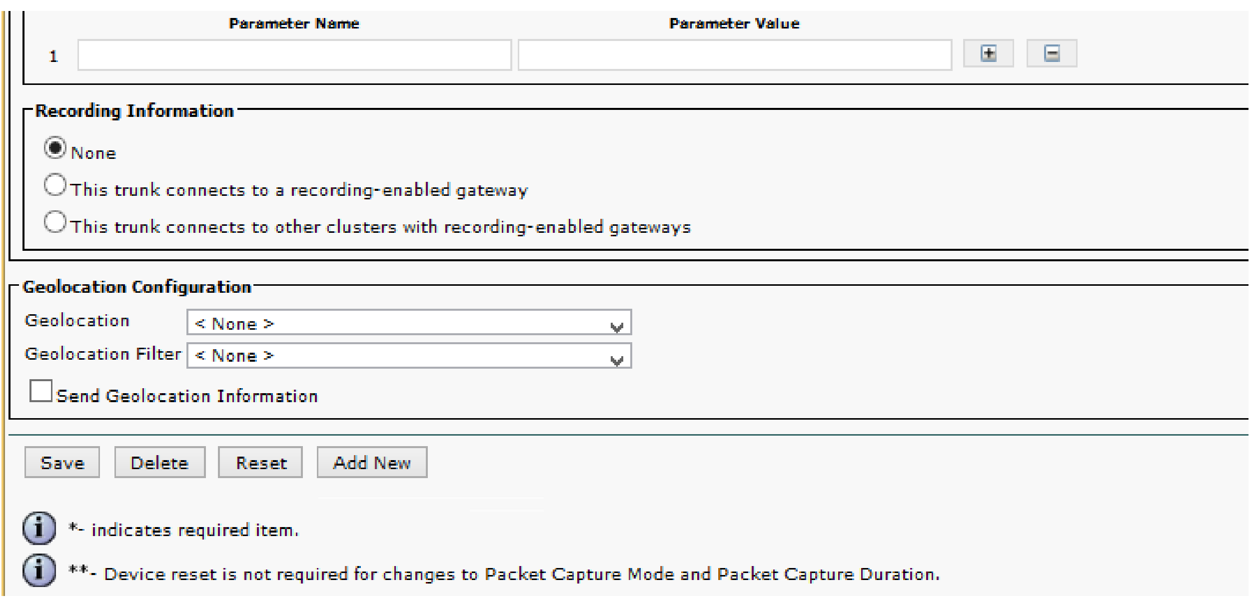
	Destination Address	Destination Address IPv6	Destination Port
1*	[redacted]		5060

MTP Preferred Originating Codec*	711ulaw
BLF Presence Group*	Standard Presence group
SIP Trunk Security Profile*	CUC Security Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile View Details
DTMF Signaling Method*	No Preference

Normalization Script

Normalization Script < None >

Enable Trace



	Parameter Name	Parameter Value
1		

Recording Information

None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation < None >

Geolocation Filter < None >

Send Geolocation Information

Save Delete Reset Add New

*- indicates required item.

**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Configuring Cisco Voice Gateway VG204XM

Show Version

```
Cisco IOS Software, VG20XXM Software (VG20XXM-IPVOICE-M), Version 15.3(2)T, RELEASE SOFTWARE (fc3)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2013 by Cisco Systems, Inc.
Compiled Thu 28-Mar-13 14:21 by prod_rel_team

ROM: System Bootstrap, Version 12.4(20r)YA2, RELEASE SOFTWARE (fc1)

VG204XM uptime is 1 hour, 1 minute
System returned to ROM by power-on
System image file is "flash:vg20xxm-ipvoice-mz.153-2.T.bin"
Last reload type: Normal Reload
Last reload reason: power-on

Cisco VG204XM (MPC8300) processor (revision 0x100) with 249856K/12288K bytes of memory.
Processor board ID FCH1807S0N8
MPC8300 CPU Rev: Part Number 0x8062, Revision ID 0x11
2 FastEthernet interfaces
4 Voice FXS interfaces
256K bytes of non-volatile configuration memory.
125496K bytes of ATA CompactFlash (Read/Write)
```

Show Running-Configuration

```
Building configuration...

Current configuration : 1942 bytes
!
! Last configuration change at 18:47:02 UTC Sun Mar 3 2002
version 15.3
no service pad
service tcp-keepalives-in
service tcp-keepalives-out
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
!
hostname VG204XM
!
boot-start-marker
boot-end-marker
!
enable secret 5 YYYYY
enable password YYYY
!
no aaa new-model
!
ip domain name ipt.local
ip cef
no ipv6 cef
!
voice-card 0
!
interface FastEthernet0/0
 ip address dhcp
 duplex auto
 speed auto
!
interface FastEthernet0/1
 ip address dhcp
 shutdown
 speed auto
 half-duplex
!
```

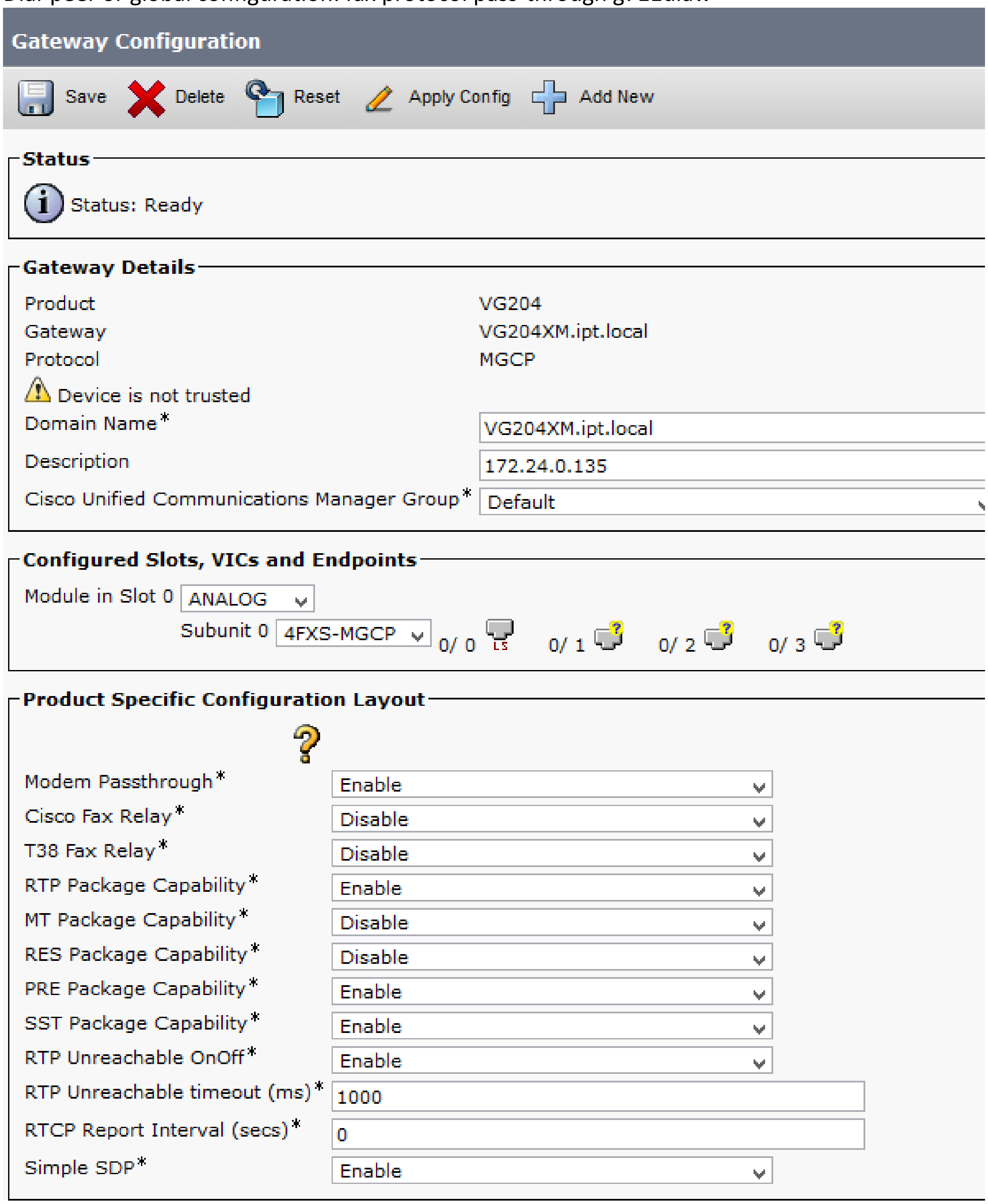
```
ip forward-protocol nd
!
no ip http server
!
control-plane
!
voice-port 0/0
 cptone CA
!
voice-port 0/1
 cptone CA
!
voice-port 0/2
 cptone CA
!
voice-port 0/3
 cptone CA
!
ccm-manager redundant-host Y.Y.Y.Y
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server X.X.X.X
ccm-manager config
!
mgcp
mgcp call-agent X.X.X.X 2427 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp bind control source-interface FastEthernet0/0
mgcp bind media source-interface FastEthernet0/0
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 1 pots
 service mgcpapp
 port 0/0
!
dial-peer voice 2 pots
 service mgcpapp
 port 0/1
!
dial-peer voice 3 pots
 service mgcpapp
 port 0/2
!
dial-peer voice 4 pots
 service mgcpapp
 port 0/3
!
line con 0
 no modem enable
line aux 0
line vty 0 4
 password YYYY
 login
 transport input all
```

!
end

Fax mode

Pass- through mode

Dial-peer or global configuration: fax protocol pass-through g711ulaw



Acronyms

Acronym	Definitions
CODEC	Coder-Decoder (in this document a device used to digitize and un-digitize voice signals)
CUBE	Cisco Unified Border Element
CUCM	Cisco Unified Communications Manager
DN	Directory Number
IP	Internet Protocol
MGCP	Media Gateway Control Protocol
MPLS	Multiprotocol Label Switching
PSTN	Public switched telephone network
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol
SP	Service Provider
TDM	Time-division multiplexing
VG	Voice Gateway
VPN	Virtual Private Network

Important Information

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Note: Testing was conducted in Telus labs

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