Application Note

TELUS SIP Trunking:

Connecting Cisco Unified Communication Manager 12.5 via the virtual Cisco Unified Border Element (vCUBE)12.6 using SIP (IP Authentication)

July 12, 2020

Note: Testing was conducted in Telus labs

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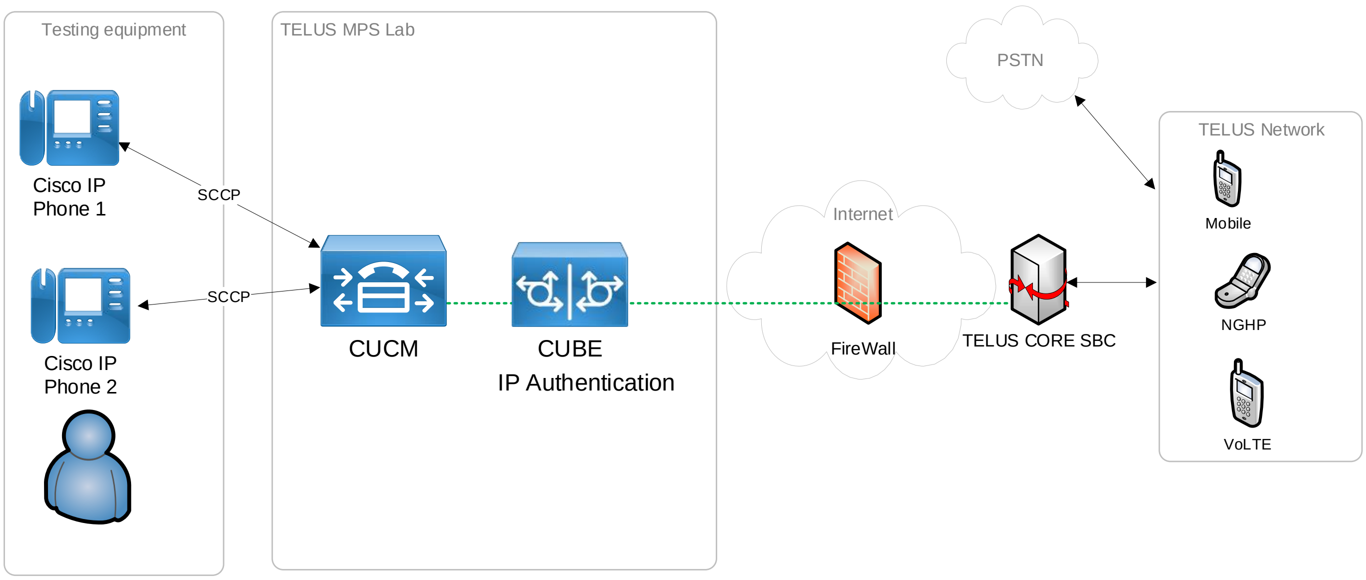
Introduction

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 SIP 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 Communication Manager (CUCM) 12.5.1.11900-146 with a Cisco Unified Border Element (CUBE) 12.6 for connectivity to TELUS SIP Trunking 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 UCSB-B200-M3 with Hypervisor VMware ESXi, 6.5.0, 5969303
- CUCM, UC, CUP and CUBE are virtual appliance
- 2 Cisco Unified IP Phones (7965)

Software Requirements

The following software is required:

- Cisco Unified Communications Manager Release 12.5. This solution was tested with 12.5.1.11900-146
- Cisco Unity Connection Release 11. This solution was tested with 12.5.1.11900-57
- Cisco Unified Border Element Release 12.6 with IOS XE version 16.11.01a release. This configuration was tested with Cisco CSR1000V (VXE)

Features

Features Tested

- 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 Tested

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

Configuration

Configuring Cisco Unified Border Element (CUBE)

Show Version

```
Cisco IOS XE Software, Version 16.11.011
Cisco IOS Software [Gibraltar], Virtual XE Software (X86_64_LINUX_IOSD-UNIVERSALK9-
M), Version 16.11.1b, RELEASE SOFTWARE (fc2)
cisco CSR1000V (VXE) processor (revision VXE) with 2168340K/3075K bytes of memory.
Processor board ID 9SMSTATZQT1
3 Gigabit Ethernet interfaces
32768K bytes of non-volatile configuration memory.
3984452K bytes of physical memory.
7774207K bytes of virtual hard disk at bootflash:.
0K bytes of WebUI ODM Files at webui:.
Configuration register is 0x2102
```

Show Running-Configuration

version 16.11
service timestamps debug datetime msec
service timestamps log datetime msec
service call-home
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
platform console virtual
hostname iptc2020cube12
boot-start-marker
boot-end-marker
no aaa new-model

ip host siptrunking.telus.com A.A.A.A[9]

login on-success log
subscriber templating
multilink bundle-name authenticated

voice service voip[1]
 mode border-element license capacity 25
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 no supplementary-service h450.2
 no supplementary-service h450.3
 no supplementary-service sip moved-temporarily
 no supplementary-service sip refer
 no supplementary-service sip handle-replaces
 redirect ip2ip
 sip
  header-passing
  referto-passing
  asserted-id pai[2]

```
 asymmetric payload dtmf[3]
 early-offer forced[4]
 midcall-signaling passthru[5]
 privacy-policy passthru[6]
 options-ping 60
 g729 annexb-all
 sip-profiles 100
 support-path-header
!
voice class sip-profiles 100

 request REGISTER sip-header To modify "<sip:(.*)@(.*)" "<sip:\1@siptrunking.telus.com>"

 request REGISTER sip-header From modify "<sip:(.*)@.*>(.*)" "<sip:\1@siptrunking.telus.com>\2"

!

voice class sip-options-keepalive 200
description PING -CUCM
down-interval 60
up-interval 30
retry 2
transport udp


voice class sip-profiles 300[11]

 request INVITE sip-header P-Asserted-Identity modify "<sip:(.*)@(.*)" "<sip:5555555555[10]@siptrunking.telus.com>"

!

dial-peer voice 100 voip
 description 10D-Dialing
 destination-pattern [2-9]..[2-9]......
 session protocol sipv2
 session target ipv4:A.A.A.A[9]
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip early-offer forced
 voice-class sip options-keepalive profile 200
voice-class sip profiles 300

 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 101 voip
 description 11D-LD-Dialing
 destination-pattern 1[2-9]..[2-9]......
 session protocol sipv2
session target ipv4:A.A.A.A[9]
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip early-offer forced
 voice-class sip options-keepalive profile 200
```

voice-class sip profiles 300

```
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 10000 voip
 description ***INBOUND***
 preference 1
 destination-pattern xxxxxxxxx[x-x]
signaling forward unconditional[8]
 session protocol sipv2
session target ipv4:B.B.B.B[7]
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive profile 200
 dtmf-relay rtp-nte
```

---

[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.

[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.

[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

[10] Trunk pilot number

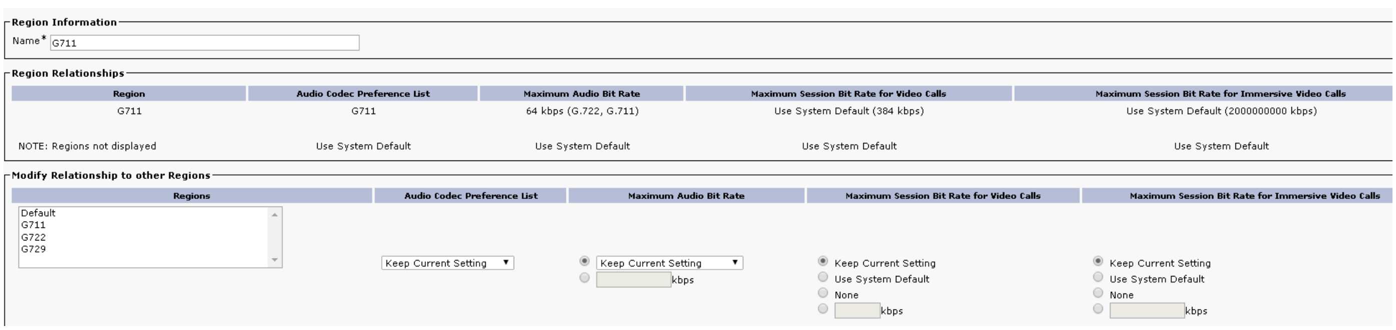
[11] In order to make outgoing call you need change PAI to pilot number!

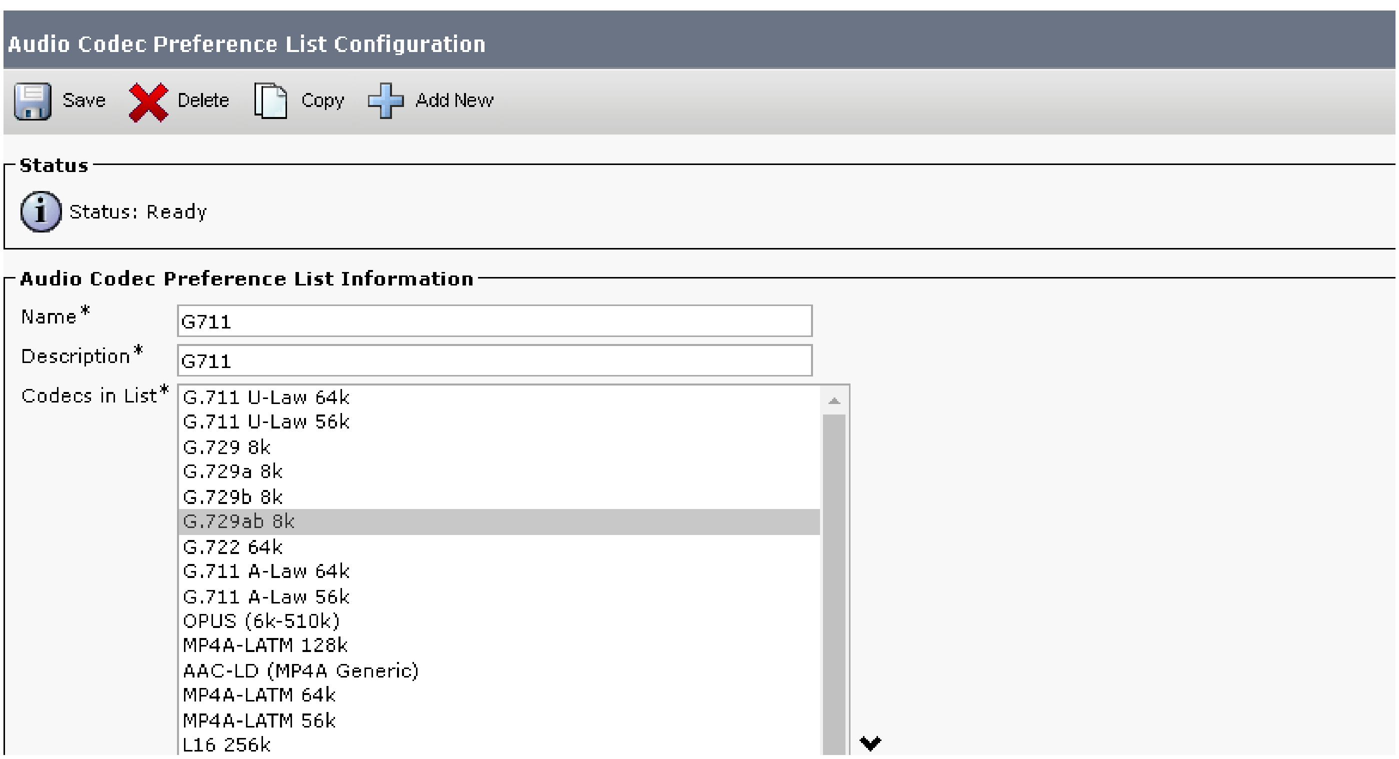
[12] Customer DID range

Configuring the Cisco Unified Communications Manager

System Version

Region (Codec settings)





Device Pool

Status

Status: Ready

Device Pool Information

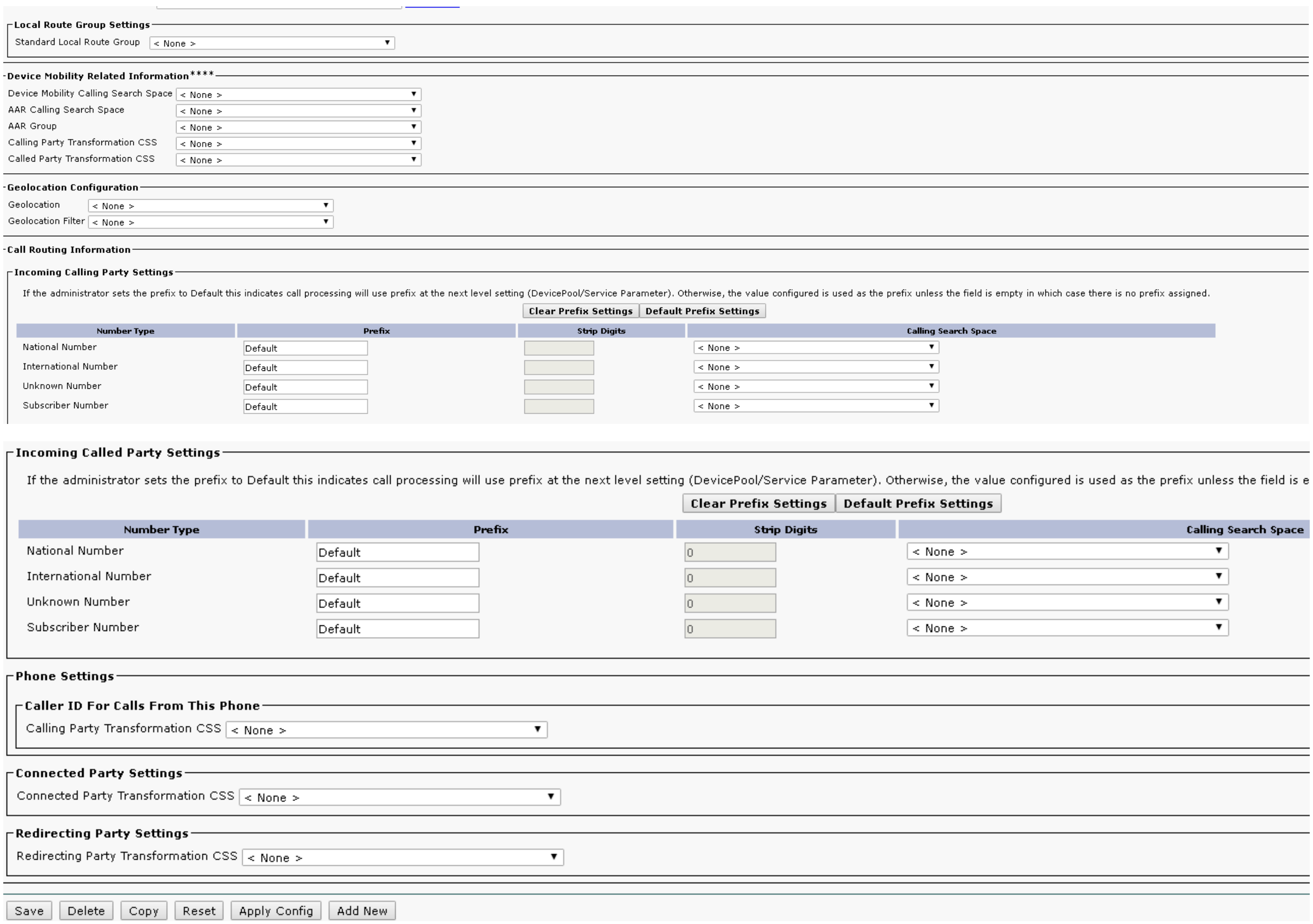
Device Pool: G711_pool (12 members**)

Device Pool Settings

Device Pool Name*	G711_pool
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	< None >
Adjunct CSS	< None >
Reverted Call Focus Priority	Default
Intercompany Media Services Enrolled Group	< None >
MRA Service Domain	< None >

Roaming Sensitive Settings

Date/Time Group*	CMLocal
Region*	G711
Media Resource Group List	MRGL
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


Local Route Group Settings
Standard Local Route Group < None >
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 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
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 the prefix unless the field is e
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 >
Phone Settings
Caller ID For Calls From This Phone
Calling Party Transformation CSS < None >
Connected Party Settings
Connected Party Transformation CSS < None >
Redirecting Party Settings
Redirecting Party Transformation CSS < None >
Save
Delete
Copy
Reset
Apply Config
Add New

SIP Trunk Cisco Unified Communications Manager

Status

Status: Ready

SIP Trunk Status

Service Status: Full Service
Duration: Time In Full Service: 8 days 2 hours 50 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	IOT22
Description	IOT22
Device Pool*	G711_pool
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
- [x] 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 >

Call Routing Information

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

Asserted-Type*	PAI
SIP Privacy*	Default
Trust Received Identity*	Trust All (Default)

Inbound Calls

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

- [ ] Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	✔

Incoming Called Party Settings

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

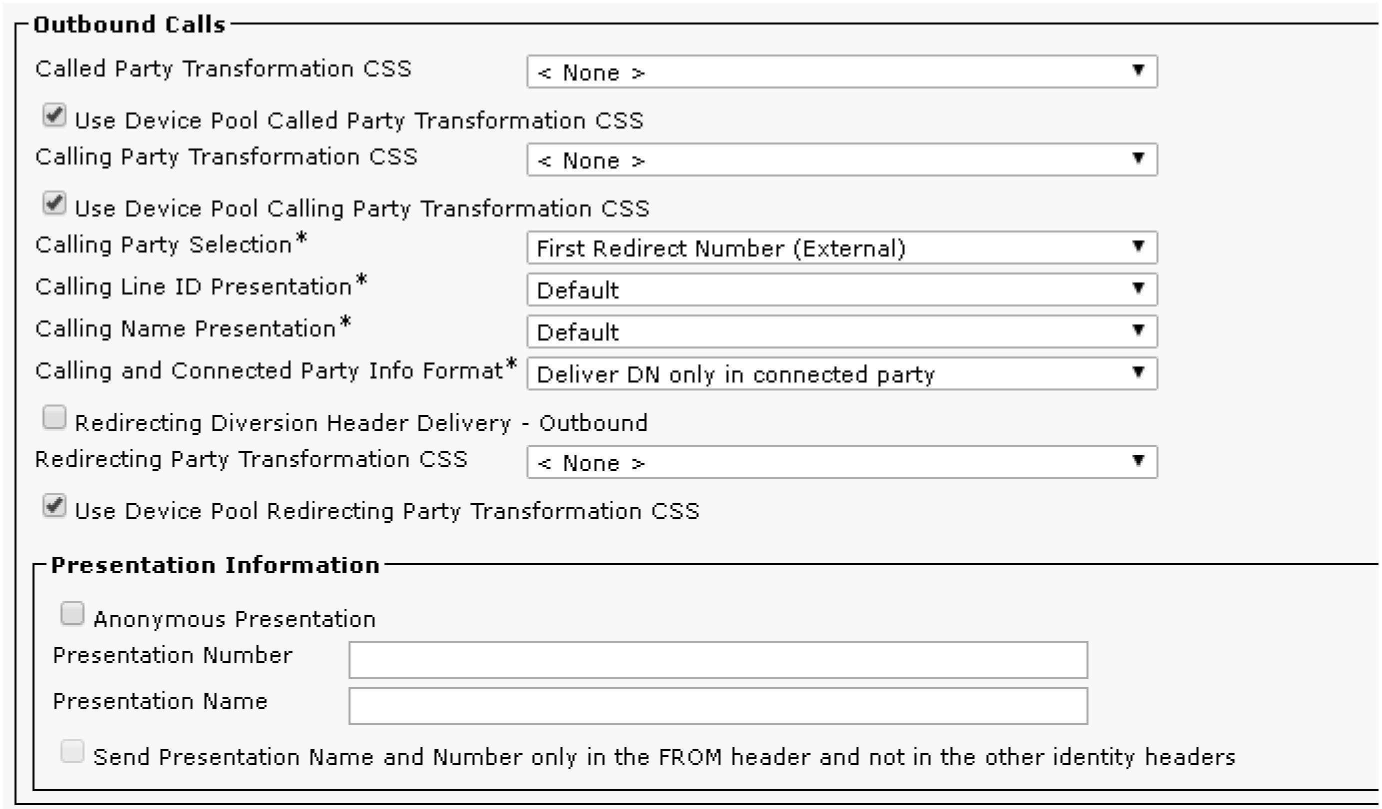
Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	✔

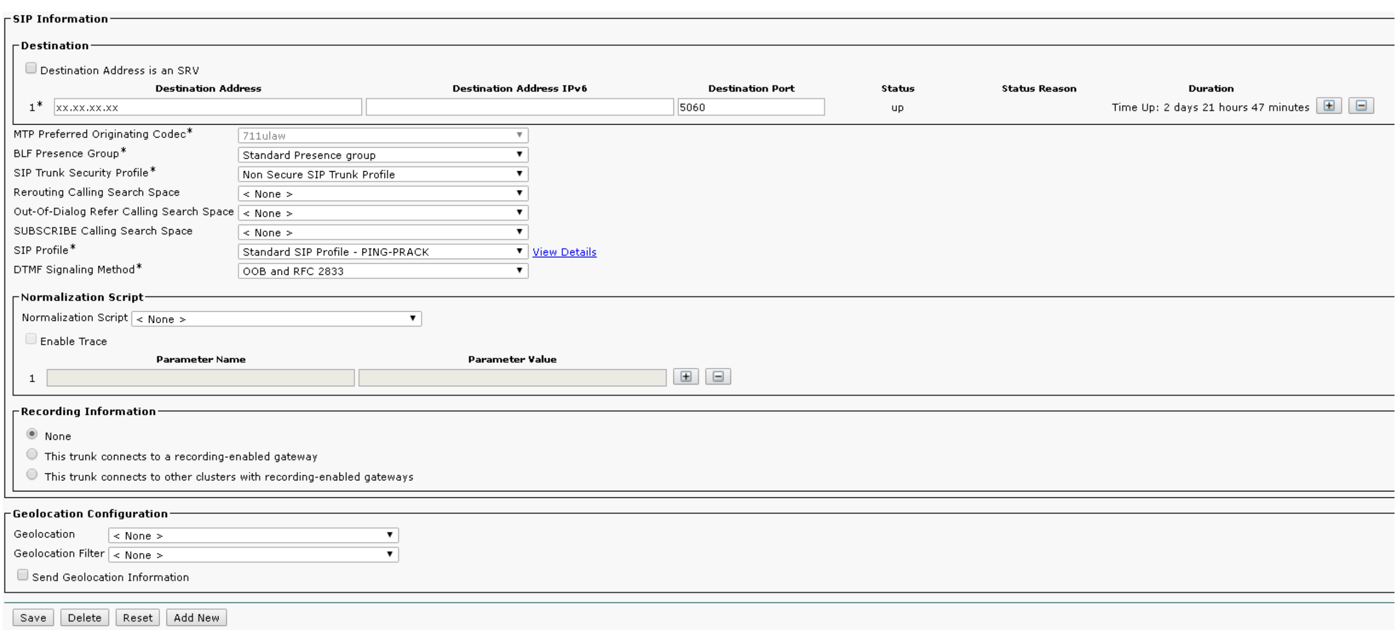
Connected Party Settings

Connected Party Transformation CSS	< None >

- [x] 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*
First Redirect Number (External)
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
Presentation Information
Anonymous Presentation
Presentation Number
Presentation Name
Send Presentation Name and Number only in the FROM header and not in the other identity headers



SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
Status
Status Reason
Duration
1*
xx.xx.xx.xx
5060
up
Time Up: 2 days 21 hours 47 minutes
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 - PING-PRACK
View Details
DTMF Signaling Method*
OOB and RFC 2833
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

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)

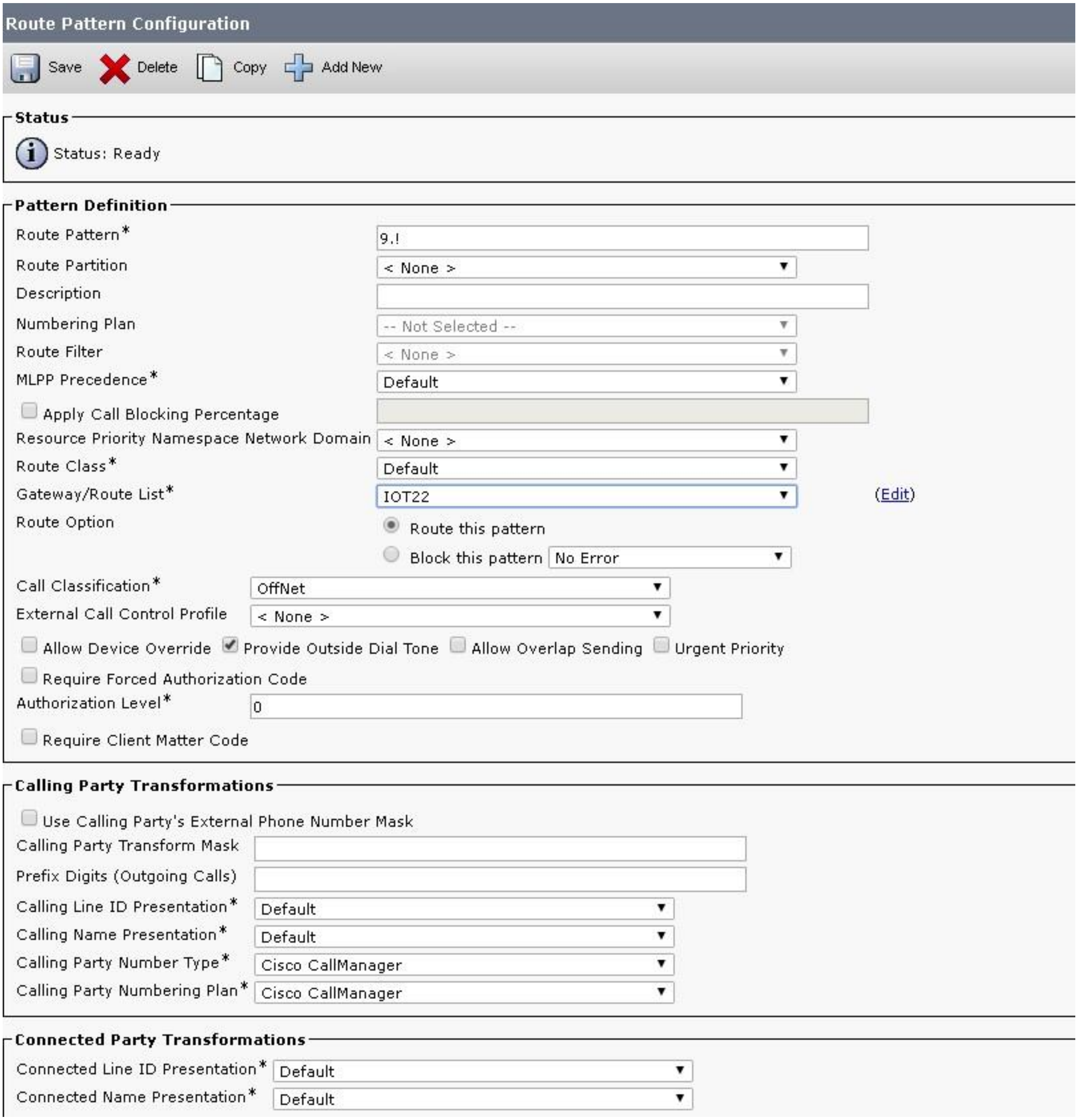
SIP Interoperability Enabled *	True	True
Retry Count for SIP Bye *	10	10
Retry Count for SIP Cancel *	10	10
Retry Count for SIP Invite *	6	6
Retry Count for SIP PRACK *	6	6
Retry Count for SIP Rel1XX *	10	10
Retry Count for SIP Publish *	6	6
Retry Count for SIP Response *	6	6
SIP Connect Timer *	500	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

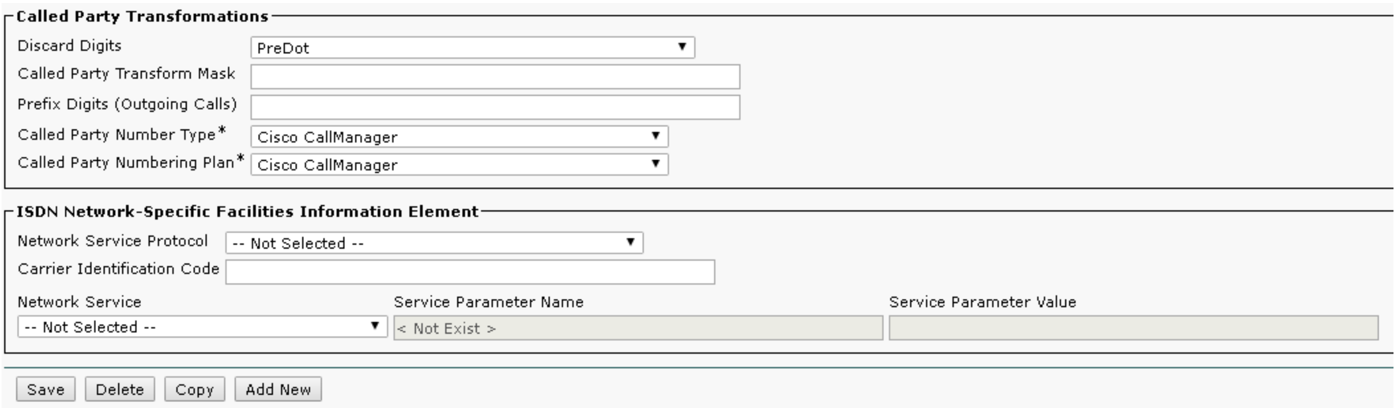
Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*	Never
Resource Priority Namespace List	< None >
SIP Rel1XX Options*	Send PRACK for all 1xx Messages
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Mandatory (insert MTP if needed)

- [ ] Enable ANAT
- [ ] Deliver Conference Bridge Identifier
- [ ] Enable External Presentation Name and Number
- [ ] Reject Anonymous Incoming Calls
- [ ] Reject Anonymous Outgoing Calls
- [ ] Send ILS Learned Destination Route String
- [ ] Connect Inbound Call before Playing Queuing Announcement

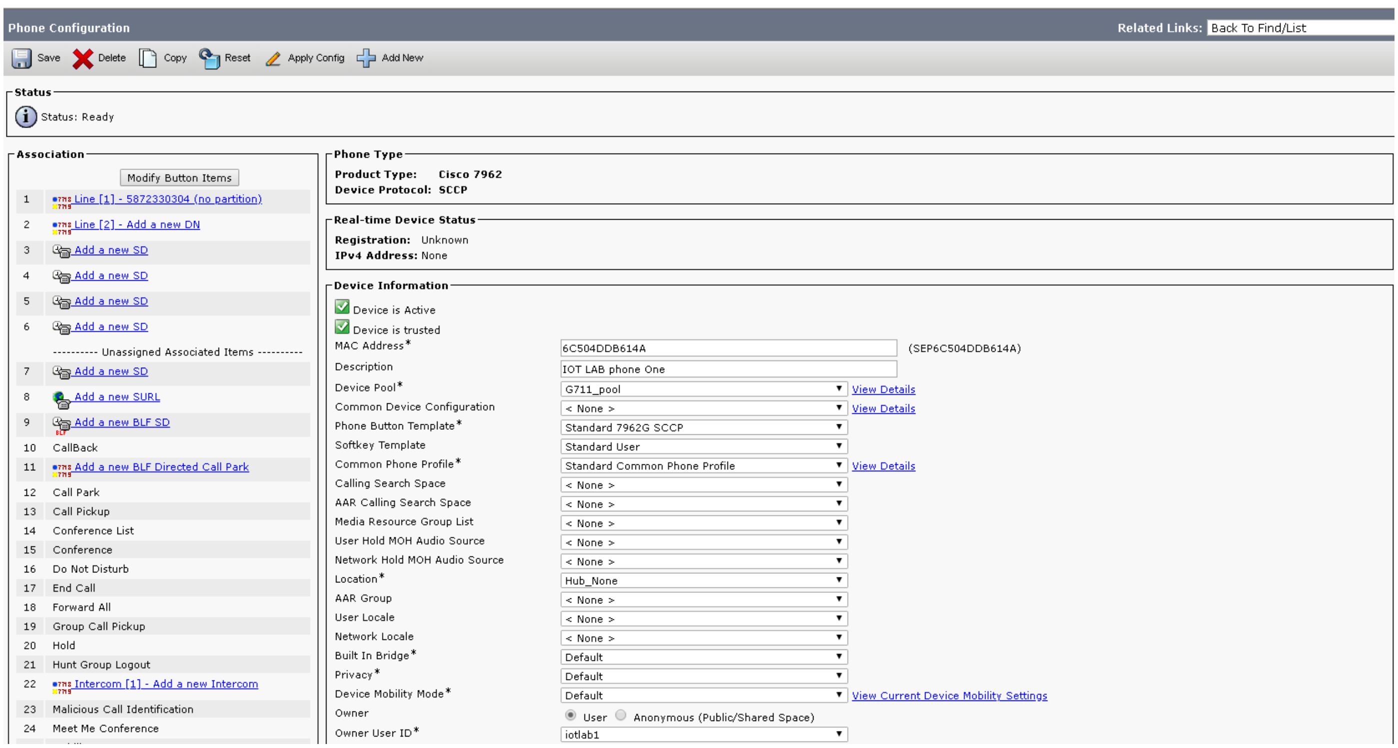
Route Pattern (SIP Trunk)

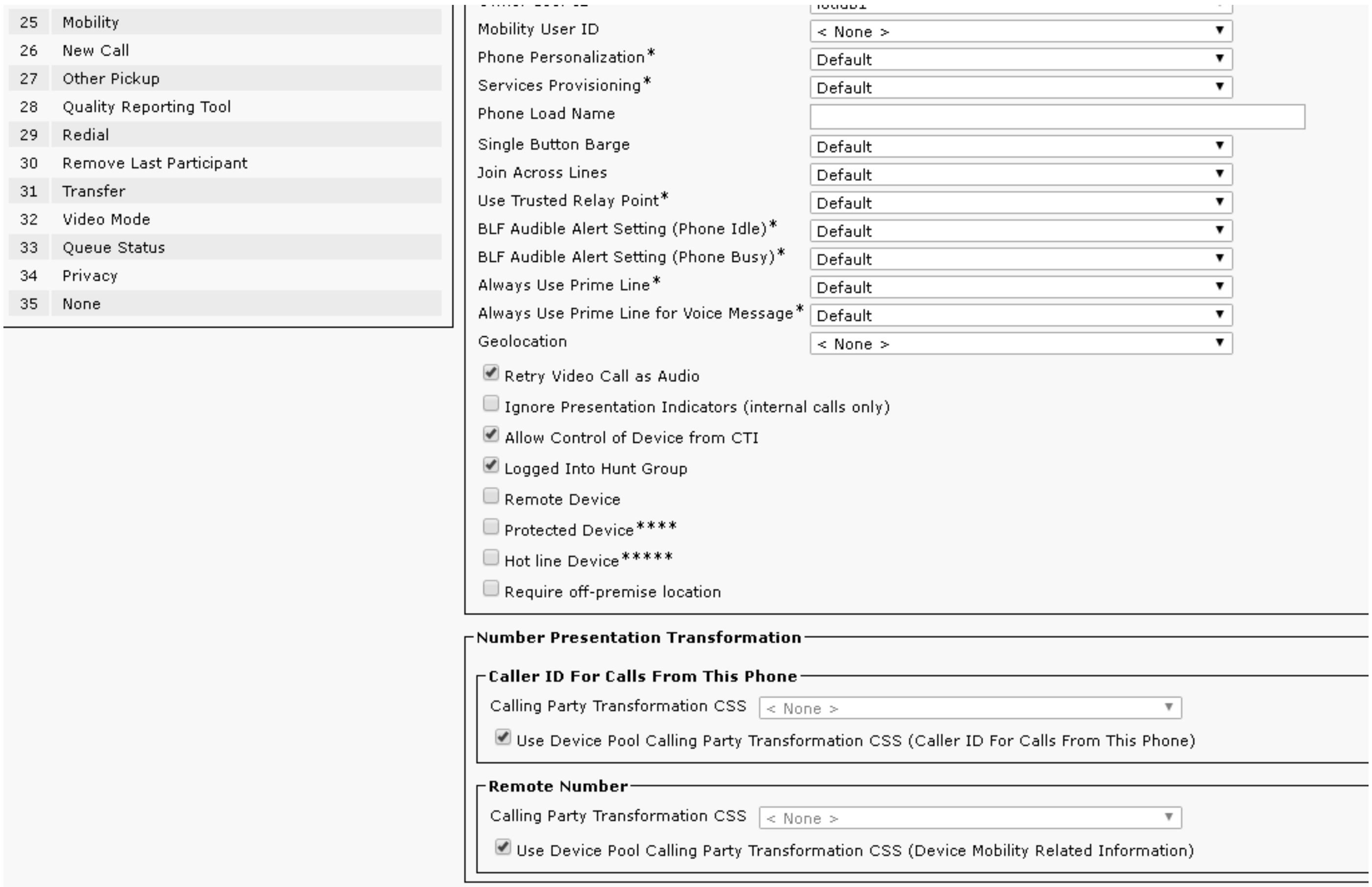




Please note that Digits 9. Predot will be discarded on all patterns on the outgoing SIP trunk. 9 was used only for route selection.

IP phone configuration




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
Mobility User ID
< None >
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 7962 - Standard SCCP Non-Secure Profile
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 Order*	RSA Only
RSA Key Size (Bits)*	2048
EC Key Size (Bits)	
Operation Completes By	2019 : 03 : : 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	
Idle Timer (seconds)	
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 >
Confidential Access Level	< None >

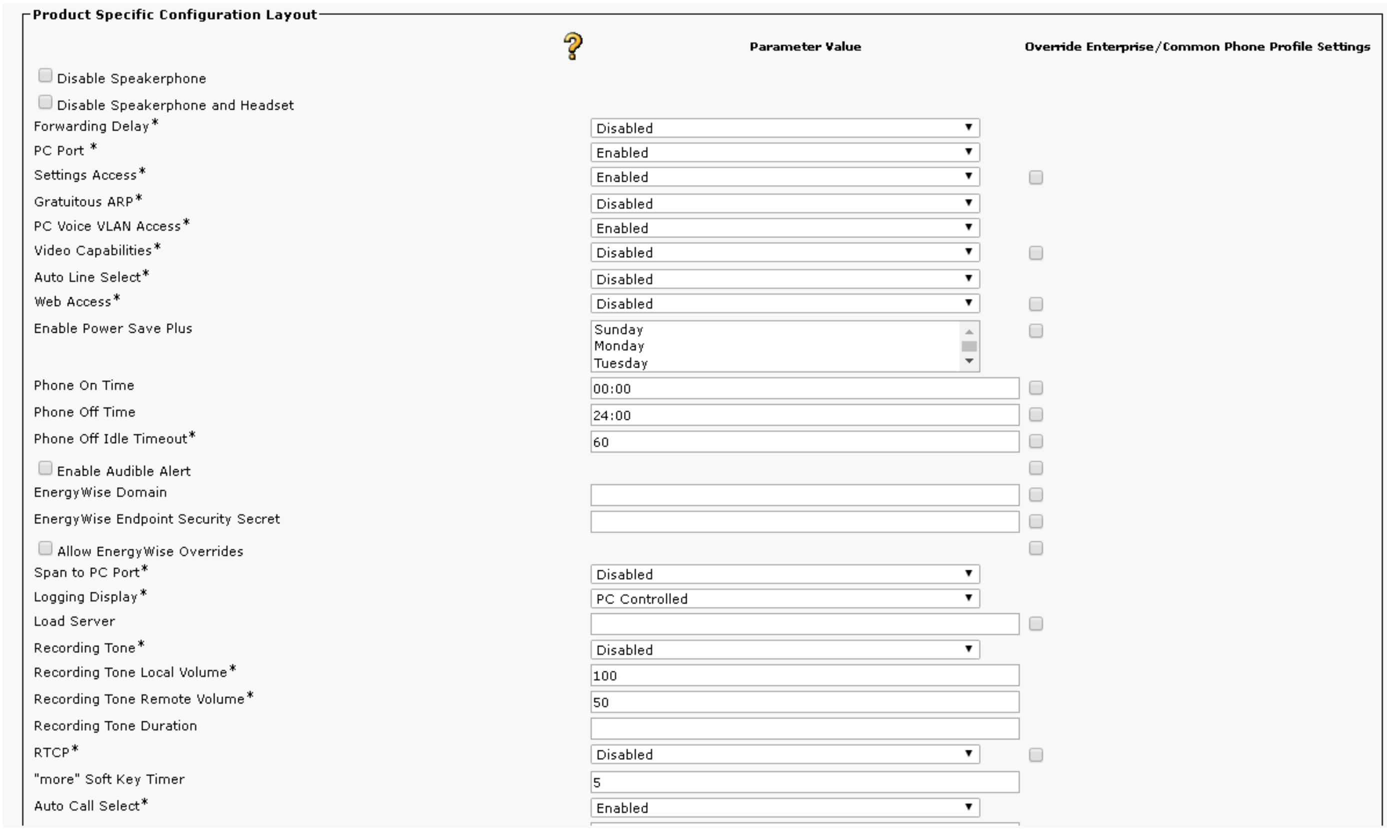
Do Not Disturb

☐ Do Not Disturb

DND Option*	Use Common Phone Profile Setting
DND Incoming Call Alert	< None >

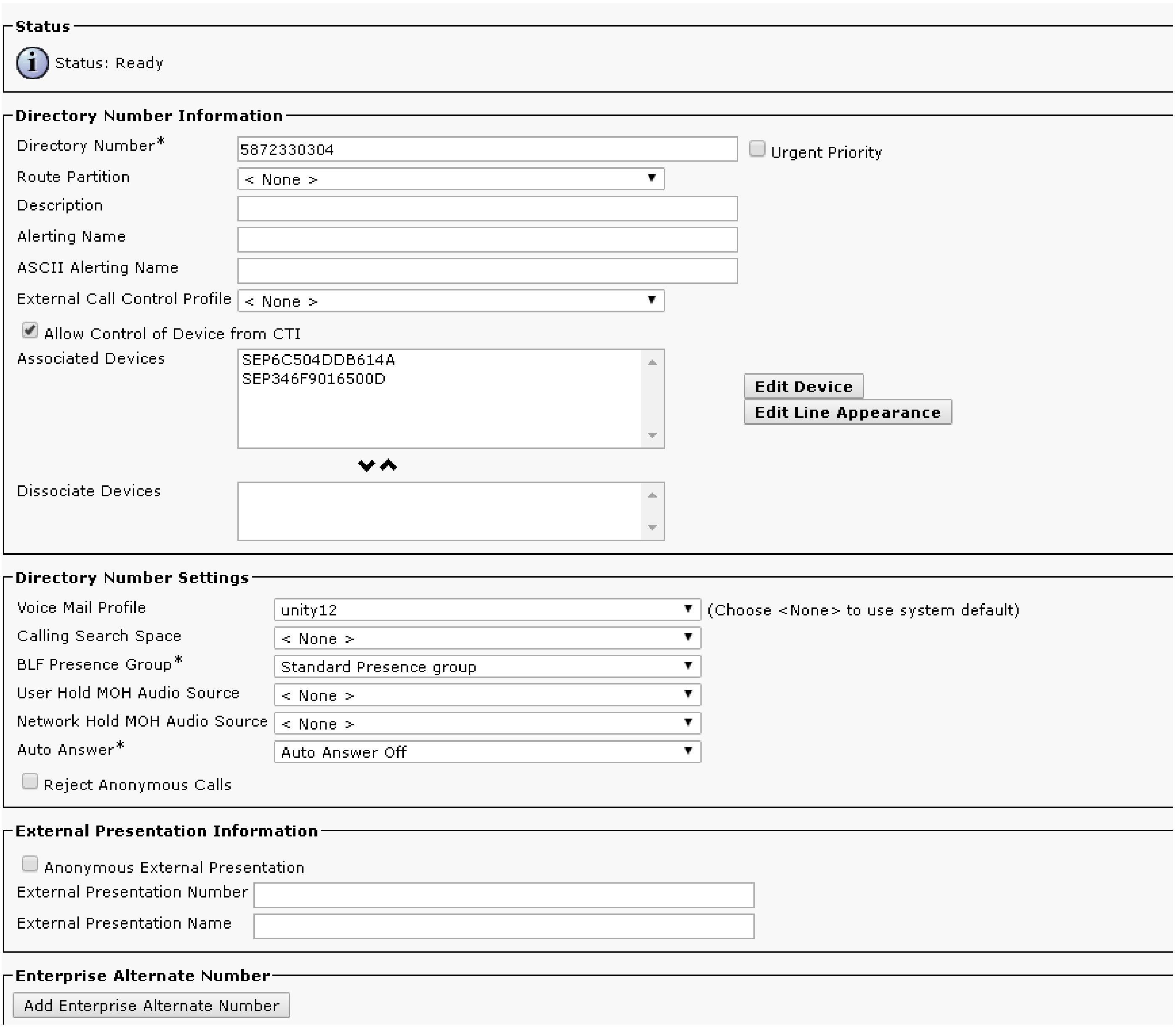
Secure Shell Information

Secure Shell User	
Secure Shell Password	

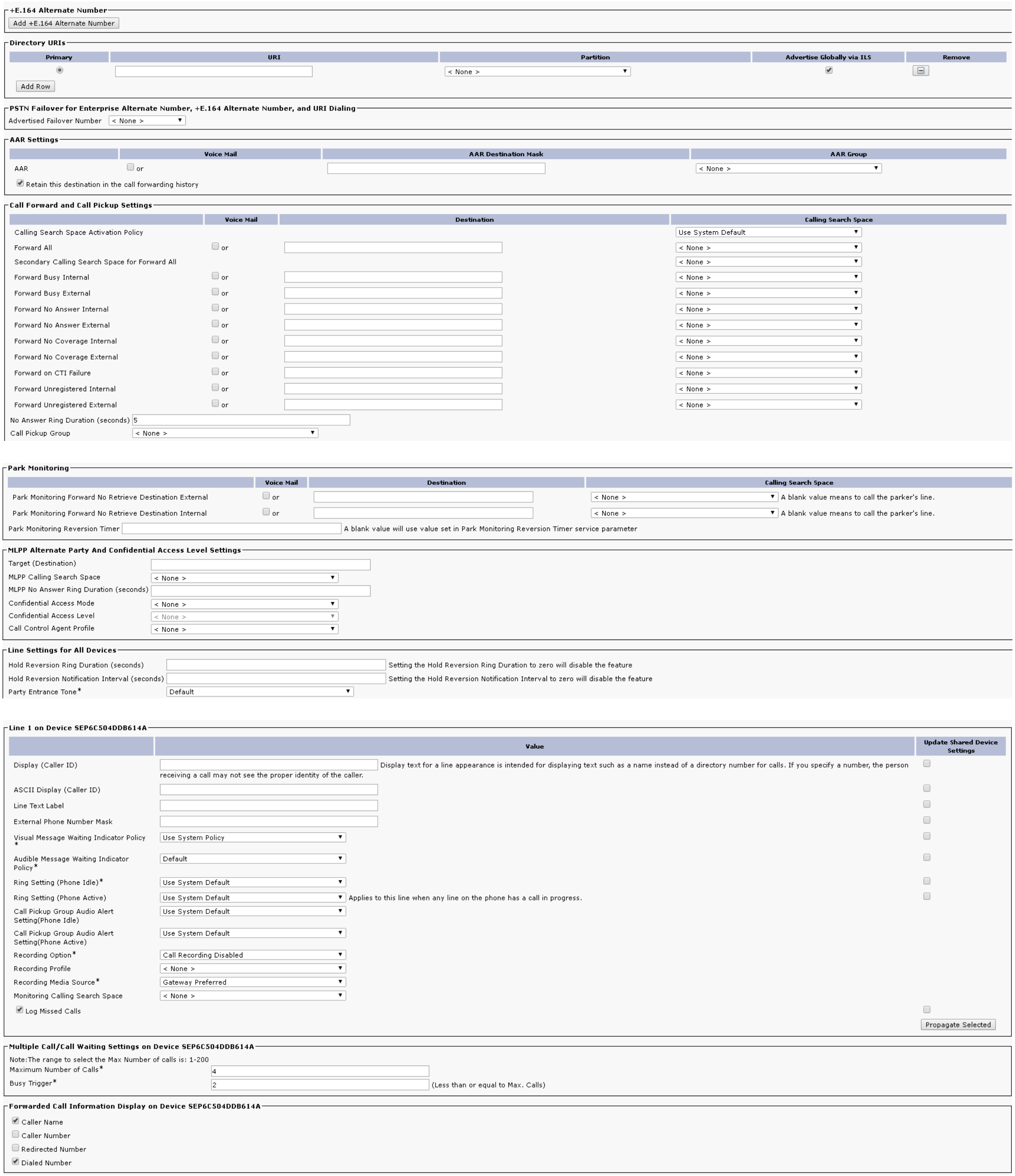

Product Specific Configuration Layout
Parameter Value
Override Enterprise/Common Phone Profile Settings
Disable Speakerphone
Disable Speakerphone and Headset
Forwarding Delay* Disabled
PC Port * Enabled
Settings Access* Enabled
Gratuitous ARP* Disabled
PC Voice VLAN Access* Enabled
Video Capabilities* Disabled
Auto Line Select* Disabled
Web Access* Disabled
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
RTCP* Disabled
"more" Soft Key Timer 5
Auto Call Select* Enabled

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
Display Refresh Rate*	Normal
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
Customer Support Use	

IP phone DN configuration



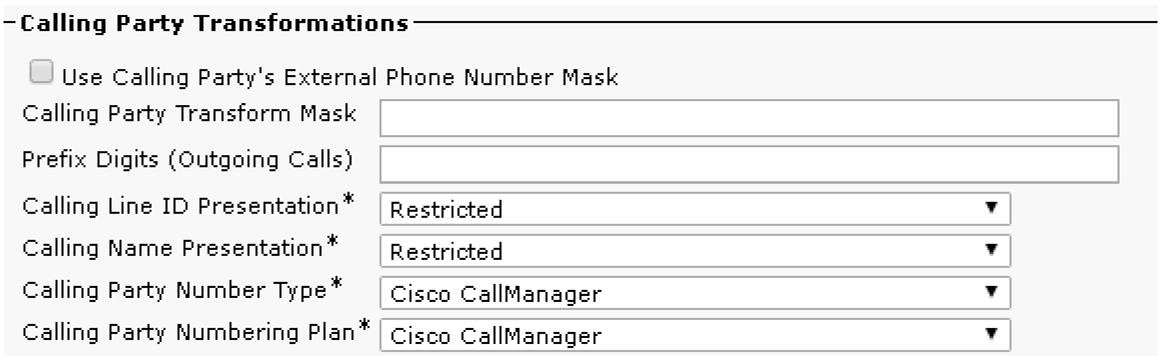
![Cisco logo]


+E.164 Alternate Number
Add +E.164 Alternate Number
Directory URIs
Primary
URI
Partition
Advertise Globally via ILS
Remove
< None >
Add Row
PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing
Advertised Failover Number
< None >
AAR Settings
Voice Mail
AAR Destination Mask
AAR Group
AAR
or
< None >
Retain this destination in the call forwarding history
Call Forward and Call Pickup Settings
Voice Mail
Destination
Calling Search Space
Calling Search Space Activation Policy
Use System Default
Forward All
Secondary Calling Search Space for Forward All
Forward Busy Internal
Forward Busy External
Forward No Answer Internal
Forward No Answer External
Forward No Coverage Internal
Forward No Coverage External
Forward on CTI Failure
Forward Unregistered Internal
Forward Unregistered External
No Answer Ring Duration (seconds)
5
Call Pickup Group
< None >
Park Monitoring
Voice Mail
Destination
Calling Search Space
Park Monitoring Forward No Retrieve Destination External
Park Monitoring Forward No Retrieve Destination Internal
A blank value means to call the parker's line.
Park Monitoring Reversion Timer
A blank value will use value set in Park Monitoring Reversion Timer service parameter
MLPP Alternate Party And Confidential Access Level Settings
Target (Destination)
MLPP Calling Search Space
MLPP No Answer Ring Duration (seconds)
Confidential Access Mode
Confidential Access Level
Call Control Agent Profile
Line Settings for All Devices
Hold Reversion Ring Duration (seconds)
Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)
Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*
Default
Line 1 on Device SEP6C504DDB614A
Value
Update Shared Device Settings
Display (Caller ID)
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)
Line Text Label
External Phone Number Mask
Visual Message Waiting Indicator Policy*
Use System Policy
Audible Message Waiting Indicator Policy*
Default
Ring Setting (Phone Idle)*
Use System Default
Ring Setting (Phone Active)
Use System Default
Applies to this line when any line on the phone has a call in progress.
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 >
Log Missed Calls
Propagate Selected
Multiple Call/Call Waiting Settings on Device SEP6C504DDB614A
Note:The range to select the Max Number of calls is: 1-200
Maximum Number of Calls*
4
Busy Trigger*
2
(Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP6C504DDB614A
Caller Name
Caller Number
Redirected Number
Dialed Number

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

Cisco Unity Connection Administration

Version 12.5.1.11900-57

User configuration

Status

Found 7 User(s)

Search Limits

Limit search to All

Users *(1 - 7 of 7)* — *Rows per Page* 25

Find Users where Alias begins with [ ] Find

		Alias	Extension	First Name	Last Name	Display Name
		0377	5872330377		Certification user	vanlab1
		iotlab1	5872330304		iot lab	iotlab1
		operator	99990			Operator
		Replication		Replication	Agent	Replication Agent (iptc2019cuc12)
		ucadmin				ucadmin
		undeliverablemessagesmailbox	99999			Undeliverable Messages
		UnityConnection		Cisco Unity Connection	Messaging System	Cisco Unity Connection Messaging System

Delete Selected | Add New | Bulk Edit | Show Dependencies

Key:
Local User
Remote User
Cisco Unity User

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

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Note: Testing was conducted in Telus labs

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