Application Note

Bell Canada SIP Trunking Service: Connecting Cisco Unified Communications Manager 10.5 via the Cisco Unified Border Element 10.5 [IOS 15.4(3)M] using SIP

December 2014

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Note: Testing was conducted in (*Bell Canada*) labs

Introduction

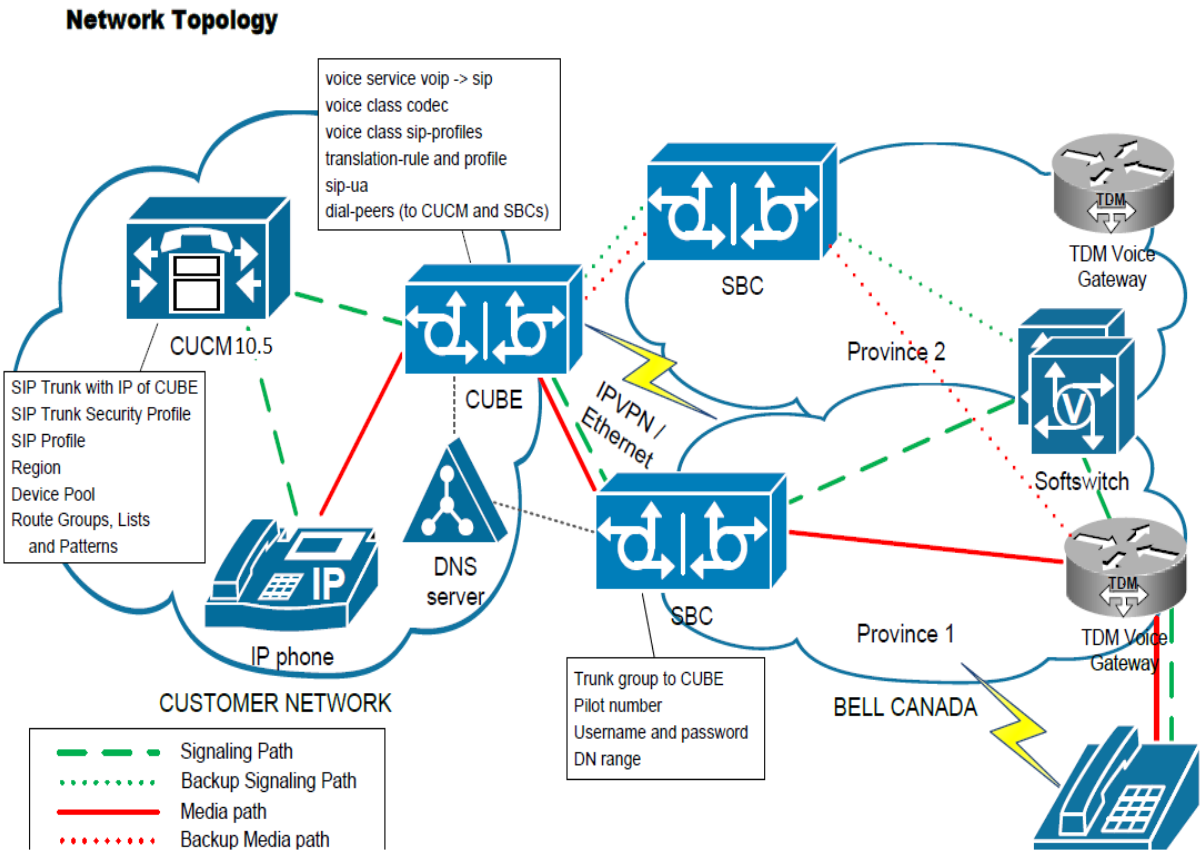
Service Providers today, such as Bell Canada, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. Bell Canada is a SP offering that 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) 10.5.1 with a Cisco Unified Border Element (CUBE) for connectivity to Bell Canada SIP Trunking Service SIP trunk service. The deployment model covered in this application note is Cisco CUCM 10.5.1 / CUBE to PSTN Bell Canada SIP trunk service. This document does not address 911 emergency outbound calls. For 911 feature service details contact Bell Canada directly.
- Testing was performed in accordance to Bell Canada SIP Trunking Test plan and all features were verified. Key features verified are:

  Inbound and outbound basic call (including international calls), calling name delivery, calling number and name restriction, DNIS translations, CODEC negotiation, advanced 8XX call prompter, intra-site transfers, intra-site conferencing, call hold and resume, call forward (forward all, busy and no answer), leaving and retrieving voicemail (Cisco Unity Connection), auto-attendant, fax using G.711 (G3 and SG3 speeds), teleconferencing, Simultaneous and Sequential Ring, failover of unresponsive SIP network to PSTN and outbound/inbound calls to/from TDM networks.
- 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 Bell Canada's 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 interoperating with Bell Canada's 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 link below: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/6_0_1/ccmsys/a03ptcss.html

Network Topology

Figure 1. Basic Call Setup



System Components

Hardware Components

Hardware Components

- CUCM V 10.5.1
- CUBE 10.5
- VG224

Software Requirements

Bell service is compatible with CUCM v10.5.1 + CUBE on Cisco IOS Software, C2900 Software (c2900-universalk9-mz.SPA.154-3.M.bin), Version 15.4(3) M, and above. The following list summarizes the version and patch levels of the hardware that has been validated as compatible in the Bell lab:

CUCM v 10.5.1

CUBE on IOS 15.4(3) M

Features

Features Supported

- Basic Call using G.729 or G.711ulaw
- Calling Party Number Presentation and Restriction
- Calling Name
- Intra-site Call Transfer
- Intra-site Conference
- Advanced 8XX Call Prompter
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- Fax using G.711 pass-through
- CUBE: performs Early-Offer-to initial SIP INVITE with SDP
- Incoming DNIS Translation and Routing
- Outbound calls to IP and TDM networks
- Auto-attendant transfer-to service
- CPE voicemail managed service (Cisco Unity Connection), leave and retrieve voice message via incoming Bell SIP Trunk
- Auto-attendant transfer-to service
- Network-based Call Forward Unconditional, Busy, No Answer and Not Reachable

Features Not Supported

- T.38 Fax protocol
- Modem transmissions

Caveats

- It is recommended to have a transponder resource if the customer network will support more than one codec.
- When a PSTN to CPE call is transferred by the CPE to a second PSTN number, the Caller ID displayed on the transfer target is the CPE DID number. It does not update to the original PSTN calling party number when the transfer is completed
- BELL SIP Trunking Service only supports G711ulaw codec for FAX calls. G.711 fax passthrough is the supported FAX transport method for the BELL SIP Trunking Service. See configuration section for details

Configuration

Configuring Cisco Unified Border Element (CUBE)

This section covers the base configuration of the customer CUBE required to set it up as the customers interface for Bell Canada's SIP Trunking service. The CUBE acts as the mediation point between the customer's CUCM PBX and the session border controllers that provide access to Bell Canada's SIP Trunking Service.

The "show version" and "show running config" output from the CUBE in Bell's verification lab is shown below for reference. The running configuration excerpt has been truncated and abbreviated for clarity, and explanatory annotations have been added to explain the intent of each important entry.

NOTE: The recommended version for the CISCO IOS load on the customer CUBE is 15.4(3) M or later. Earlier versions might not support some of the configuration entries displayed below. Pay special attention to the bolded entries, and their annotation numbers, which map to the explanatory footnotes at the end of the configuration. Extraneous empty lines (marked with "!") have been added to the excerpt to keep configuration sections in one piece for ease of reading.

```
BITS_SIP_TRUNK_2921#show version
Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.4(3)M, RELEASE
SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2014 by Cisco Systems, Inc.

ROM: System Bootstrap, Version 15.0(1r)M9, RELEASE SOFTWARE (fc1)

BITS_SIP_TRUNK_2921 uptime is 7 weeks, 3 days, 31 minutes
System returned to ROM by reload at 11:15:42 EDT Thu Oct 9 2014
System image file is "flash:c2900-universalk9-mz.SPA.154-3.M.bin"
Last reload type: Normal Reload
Last reload reason: Reload Command

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 CISCO2921/K9 (revision 1.0) with 1519616K/53248K bytes of memory.
Processor board ID FHK1448F0UZ
6 Gigabit Ethernet interfaces
3 terminal lines
1 Virtual Private Network (VPN) Module
1 Internal Services Module (ISM) with Services Ready Engine (SRE)
DRAM configuration is 64 bits wide with parity enabled.
255K bytes of non-volatile configuration memory.
2048256K bytes of ATA System CompactFlash 0 (Read/Write)
```

<<CUBE Configuration excerpt BEGIN>>

```
voice-card 0
 dspfarm
 dsp services dspfarm
!
voice service voip
 address-hiding
 mode border-element
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 fax protocol pass-through g711ulaw
 sip
  asserted-id pai
  localhost dns:simcoe-siptcube.on.bell.ca
  options-ping 60
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
!
voice class sip-profiles 416
 request INVITE sip-header To modify "@.*>" "@siptrunking.bell.ca>"
 request INVITE sip-header From modify "(@.*)>" "@simcoe-siptcube.on.bell.ca;user=phone>"
 request INVITE sip-header Diversion modify "<sip:.1(.*)@(.*)>" "<sip:\1@simcoe-siptcube.on.bell.ca;user=phone>"
 request INVITE sip-header SIP-Req-URI modify "10.220.254.228" "siptrunking.bell.ca"
 request INVITE sip-header P-Asserted-Identity modify "(@.*)>" "@simcoe-siptcube.on.bell.ca;user=phone>"
```

```
 request INVITE sip-header Contact modify "@" ";tgrp=simcoe_01_4163532154_admin;trunk-context=siptrunking.bell.ca@"
!
voice translation-rule 1
 rule 1 /^9/ //
!
voice translation-rule 2
 rule 1 /^\+1/ //
!
voice translation-profile remove+1
 translate calling 2
!
voice translation-profile remove9
 translate called 1
!
!
sccp local GigabitEthernet0/2
sccp ccm 10.220.22.161 identifier 1 version 7.0
sccp
!
sccp ccm group 1
 bind interface GigabitEthernet0/2
 associate ccm 1 priority 1
 associate profile 2 register Transcoder
 associate profile 1 register Conference
!
!
!
dspfarm profile 2 transcode
 codec g729abr8
 codec g729ar8
 codec g711alaw
 codec g711ulaw
 maximum sessions 20
 associate application SCCP
!
dspfarm profile 1 conference
```

```
 codec g729br8
 codec g729r8
 codec g729abr8
 codec g729ar8
 codec g711alaw
 codec g711ulaw
 maximum sessions 4
 associate application SCCP
!
dial-peer voice 41635321 voip
 description outgoing leg to SIP trunk for default trunk
 translation-profile outgoing remove9
 destination-pattern 9T
 session protocol sipv2
 session target ipv4:10.220.254.228
 voice-class codec 1
 no voice-class sip localhost
 voice-class sip profiles 416
 voice-class sip options-keepalive down-interval 20
 dtmf-relay rtp-nte
!
dial-peer voice 41635322 voip
 description outgoing leg to SIP trunk for Load Balancer
 translation-profile outgoing remove9
 preference 1
 destination-pattern 9T
 session protocol sipv2
 session target ipv4:10.220.254.231
 voice-class codec 1
 no voice-class sip localhost
 voice-class sip profiles 416
 voice-class sip options-keepalive down-interval 20
!
dial-peer voice 41635323 voip
 description outgoing leg to SIP trunk for Load Balancer
```

```
 translation-profile outgoing remove9
 preference 2
 destination-pattern 9T
 session protocol sipv2
 session target ipv4:10.220.254.234
 voice-class codec 1
 no voice-class sip localhost
 voice-class sip profiles 416
 voice-class sip options-keepalive down-interval 20
 dtmf-relay rtp-nte
!
dial-peer voice 416353 voip
 description outgoing leg to CUCM-10.5 for 41635321..
 translation-profile incoming remove+1
 answer-address +1T
 destination-pattern 41635321..
 session protocol sipv2
 session target ipv4:10.220.22.161
 voice-class codec 1
 dtmf-relay rtp-nte
!
sip-ua
 keepalive target dns:simcoe-siptcube.on.bell.ca
 authentication username Bell password 7 123B001B1E5A5BB7E
 no remote-party-id
 retry invite 2
 retry response 3
 retry bye 3
 retry prack 6
```

<<CUBE Configuration excerpt END>>

Configuration Annotations:

- These settings are global SIP settings. Depending on your specific requirements, these parameters may also be provisioned specifically in dial-peer entries for more refined control granularity.
- The company domain agreed upon during service setup with Bell Canada. This parameter is used in the From: and P-Asserted-Identity: headers as the host part of the address URI.
- "early-offer-forced" is used to force an early offer of media codec support (via SDP message body) in outgoing INVITE messages. Bell Canada SIP Trunking requires this early offer.
- This privacy-policy setting (passthru) is used to indicate that the session target is trusted and it will maintain privacy requests. This setting is new in IOS version 15 for the CUBE.
- Voice codec class setup, this example shows that both G711ulaw and g729 are defined for this class. Use of this voice class will supply both codec in the offer during media negotiation.
- Modification of Diversion header to allow for Call Forwarding from CUCM v10.5.1. The modification simply replaces the host portion of the URI with the Bell Canada SIP Trunking service domain. It also adds the required (mandatory) "user=phone" parameter.
- The contact header is modified to supply a tgrp parameter (SIP RFC 4904) to select a specific trunk for outbound calls.
- A further addition of mandatory user=phone parameter to the From header.
- A further addition of mandatory user=phone parameter to the P-Asserted Identity header.
- Specifies the use of the codec class defined in "codec class 1"
- Specifies the sip profile to use. "The specified profile provides trunk selection via trgp specification.
- Specifies DTMF signaling type for SIP Trunking service.
- This Dial peer is used as a default outgoing dial peer for outgoing calls.
- Another Dial peer Specifies the address of the customer's CUCM v10.5.1 PBX. This session target is the target to be used for inbound calls, which obviously need to terminate on the customer's CUCM.
- This Dial peer is used as a default incoming dial peer from the customer's SIP Trunking service. The E.164 prefix "+1" will be stripped before sending the call to the CUCM v10.5.1.
- Specifies the DIGEST credentials to be used for authentication challenges to outbound SIP requests. These credentials are provided by Bell Canada.
- Bell Canada SIP Trunking only supports UDP transport.

Configuring the Cisco Unified Communications Manager

This section covers the base configuration of the CUCM to communicate with a Cisco Unified Border Element (CUBE). The CUBE acts as the customer interface to Bell Canada's SIP Trunking Service.
The customer administrator will use the CUCM Administration Web interface to implement the following CUCM configuration examples:

Configure a SIP Profile

Navigate to Device → Device Settings → SIP Profile

- Press the Find button to display existing SIP Profile entries.
- Copy the existing Default SIP Profile using the Copy icon on the right hand side (See **NOTE** below).
- Name the New SIP Profile.
- There are no changes made to the settings of the CUCM Default SIP Profile for the CUCM SIP Profile.
- Save the configuration

NOTE: Cisco strongly suggests that each configuration profile component be created as a copy of the default version, rather than using the default version as is, or altering the default profile and re-saving it. The default profile provides a clean base reference point for future profiles, or to revert to in the event that a configuration becomes corrupted.

SIP Profile Information

Name*	BELL SIP Profile
Description	BELL SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agen
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, anc
Confidential Access Level Headers*	Disabled

- [ ] Redirect by Application
- [ ] Disable Early Media on 180
- [ ] Outgoing T.38 INVITE include audio mline
- [ ] Use Fully Qualified Domain Name in SIP Requests
- [ ] Assured Services SIP conformance

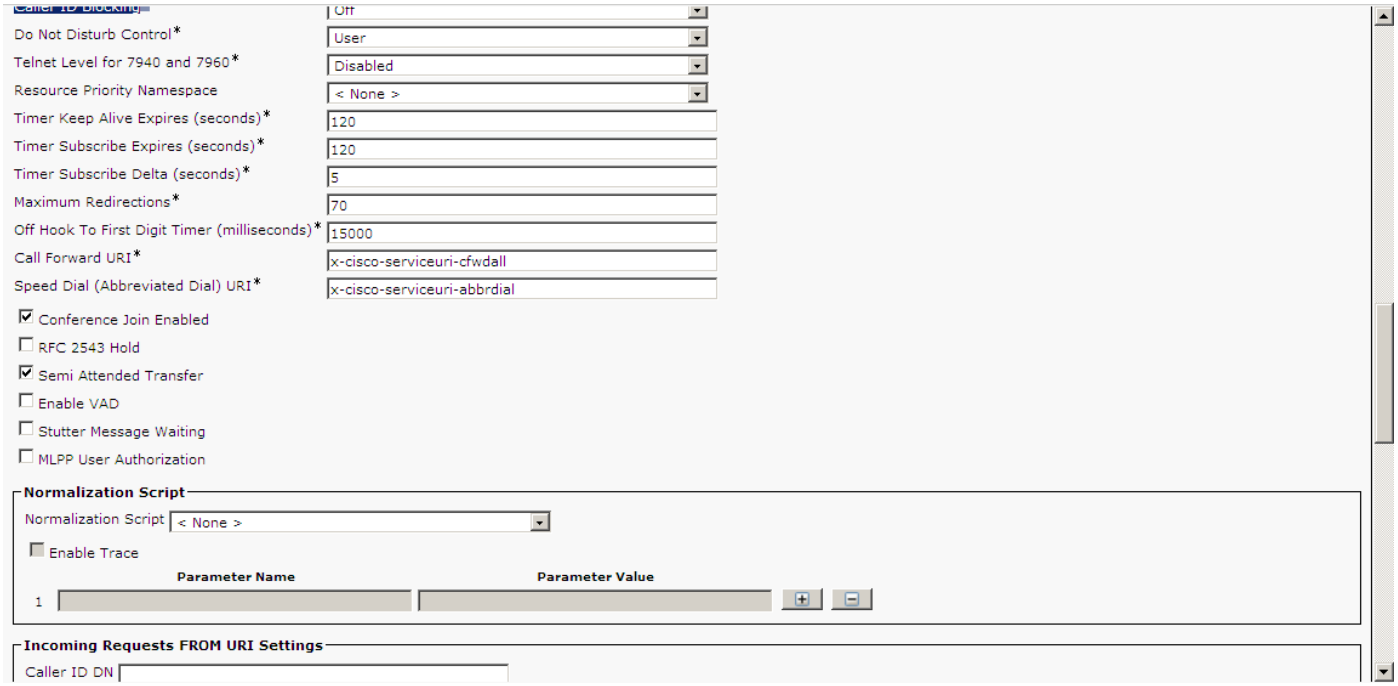
SDP Information

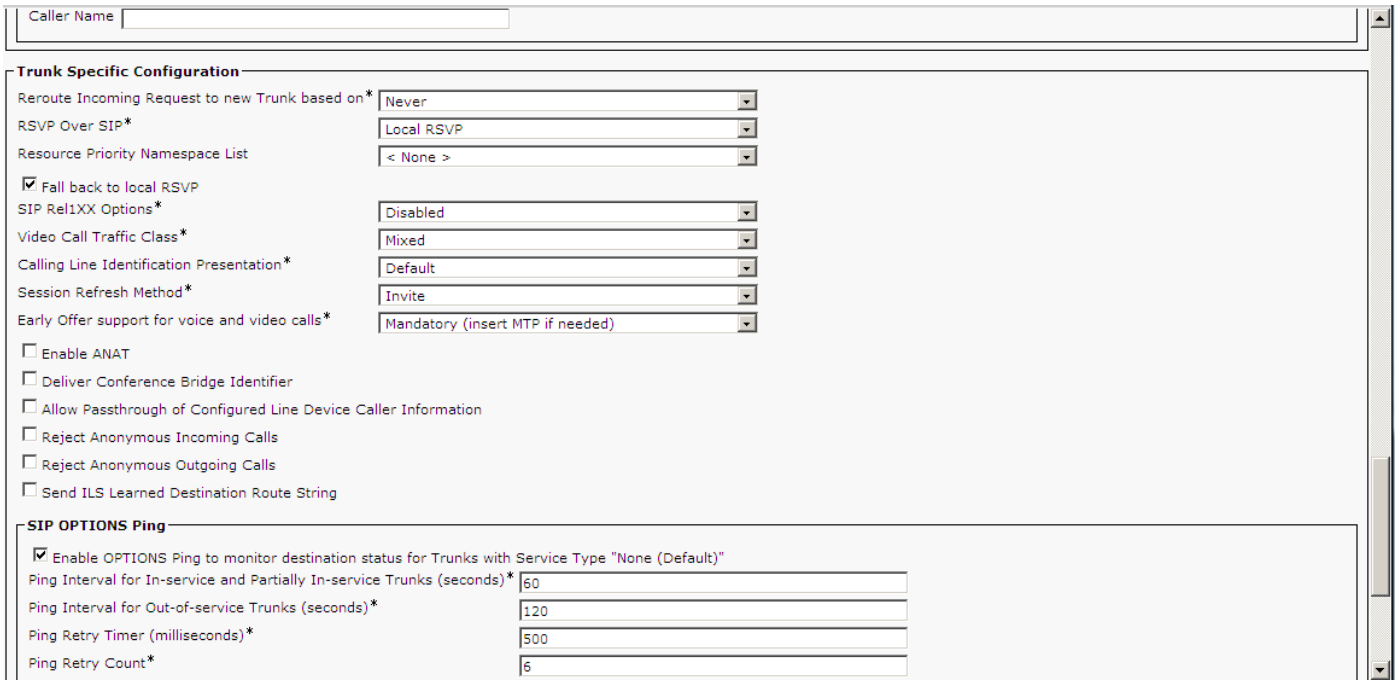
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default

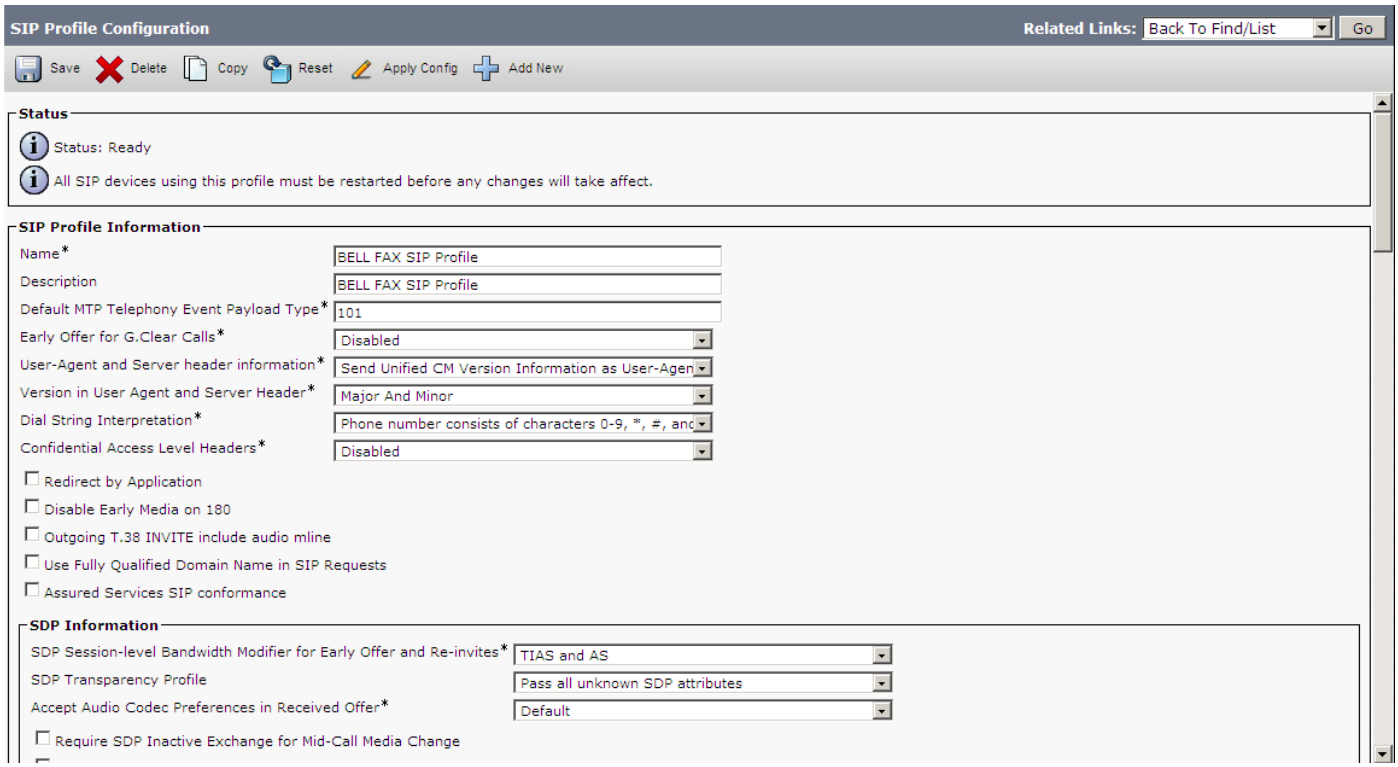
- [ ] Require SDP Inactive Exchange for Mid-Call Media Change
- [ ] Allow RR/RS bandwidth modifier (RFC 3556)

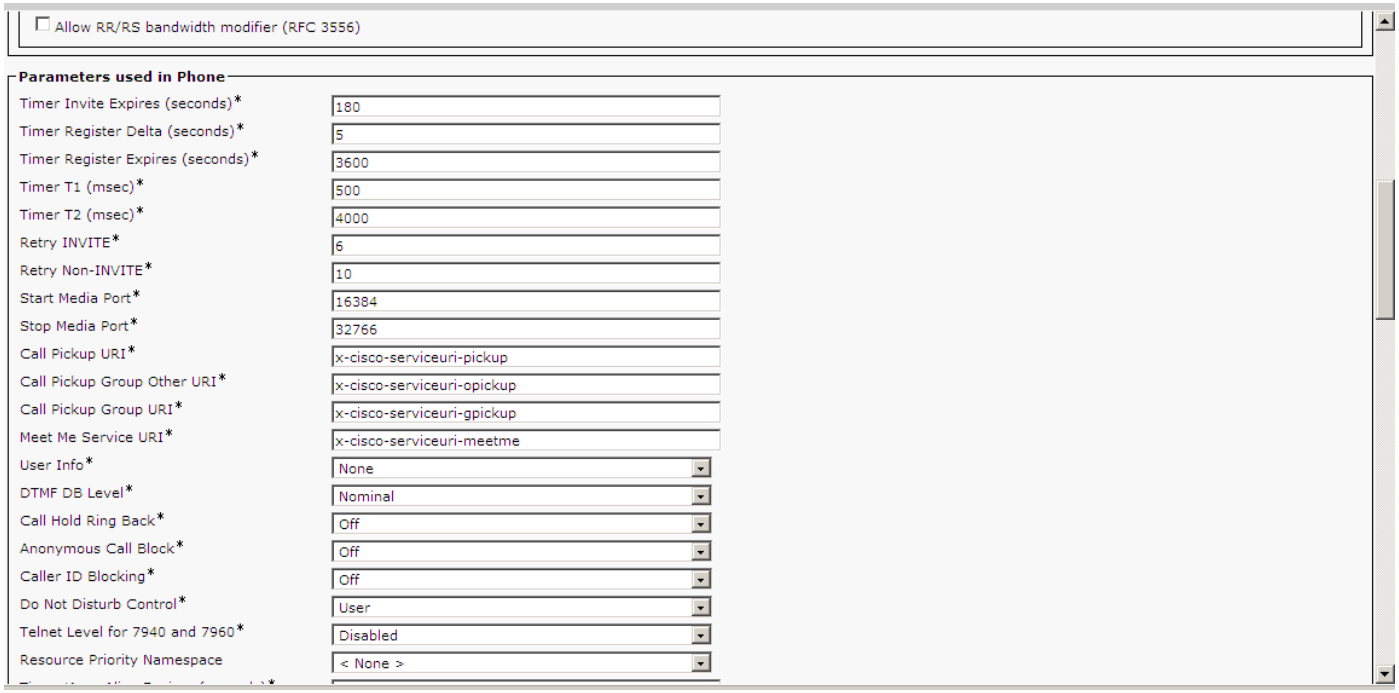
Parameters used in Phone

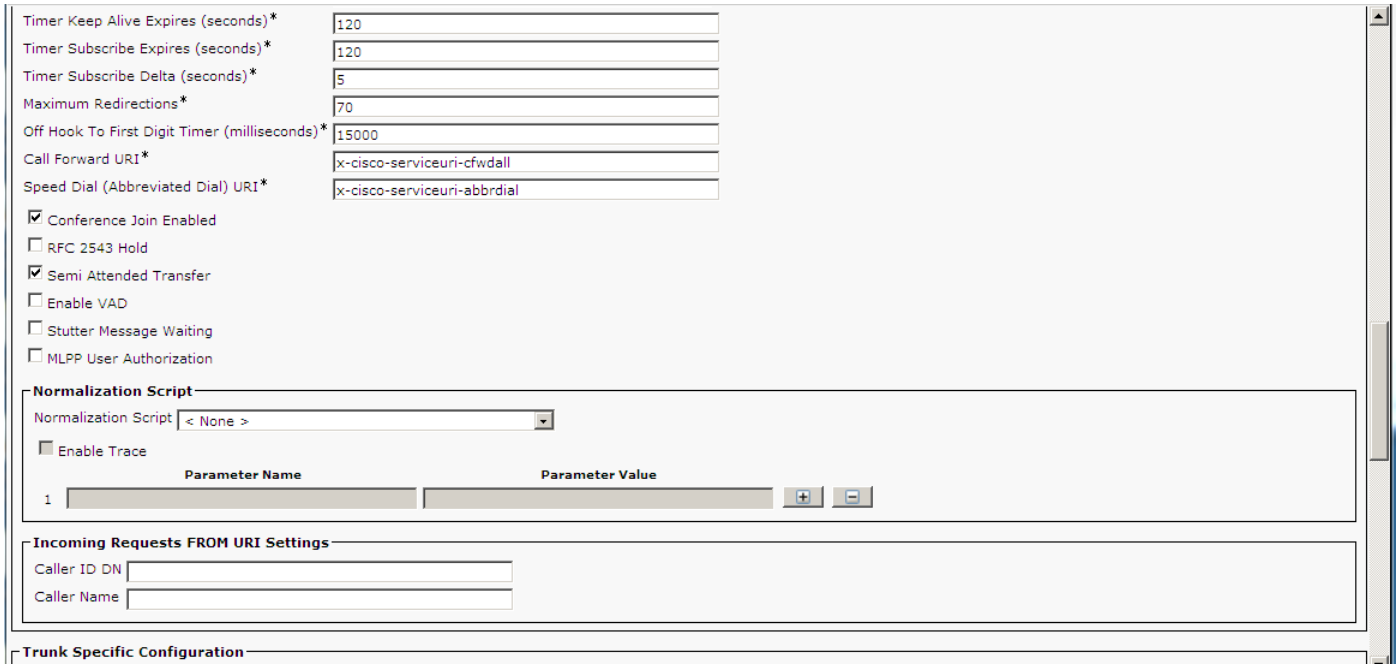
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off

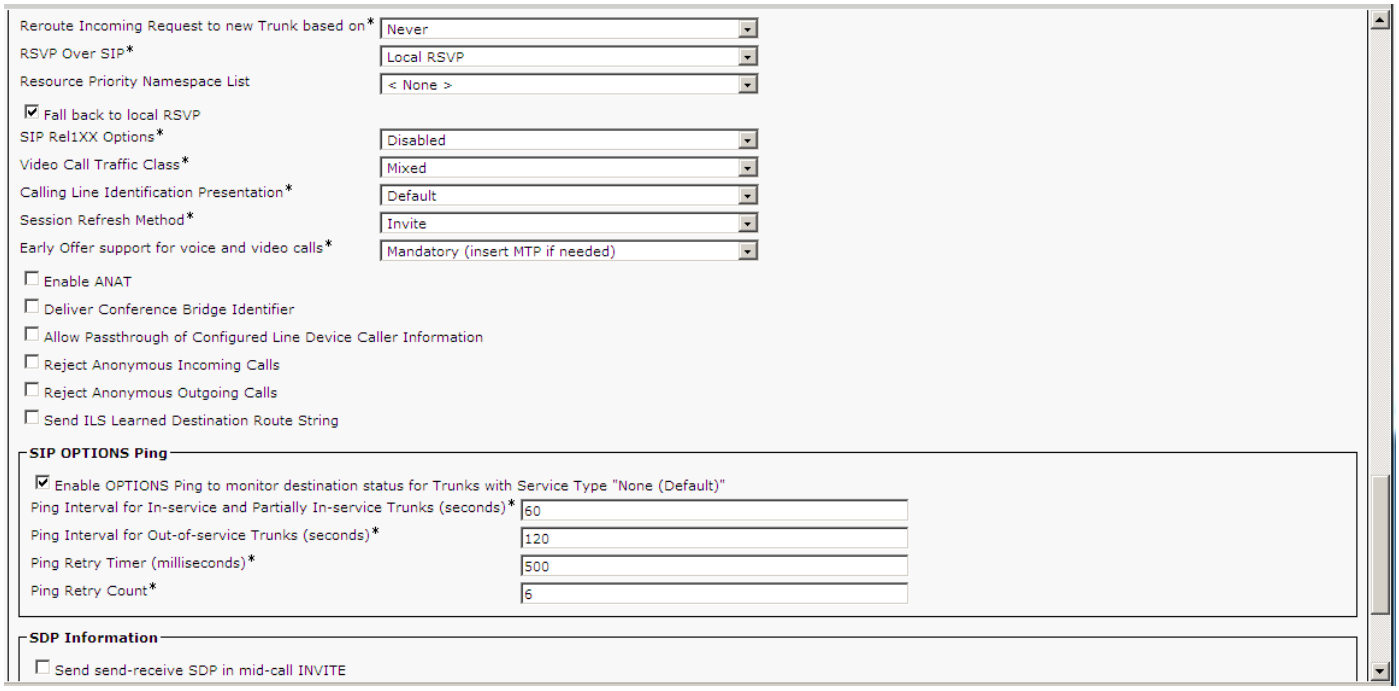


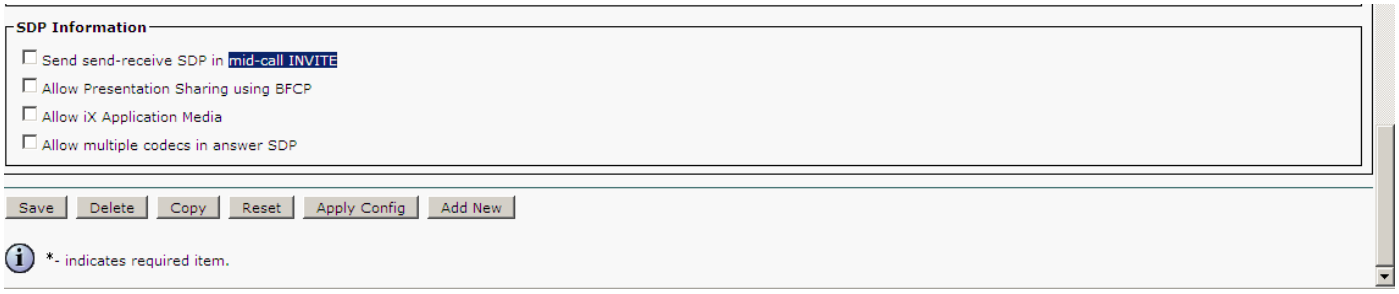








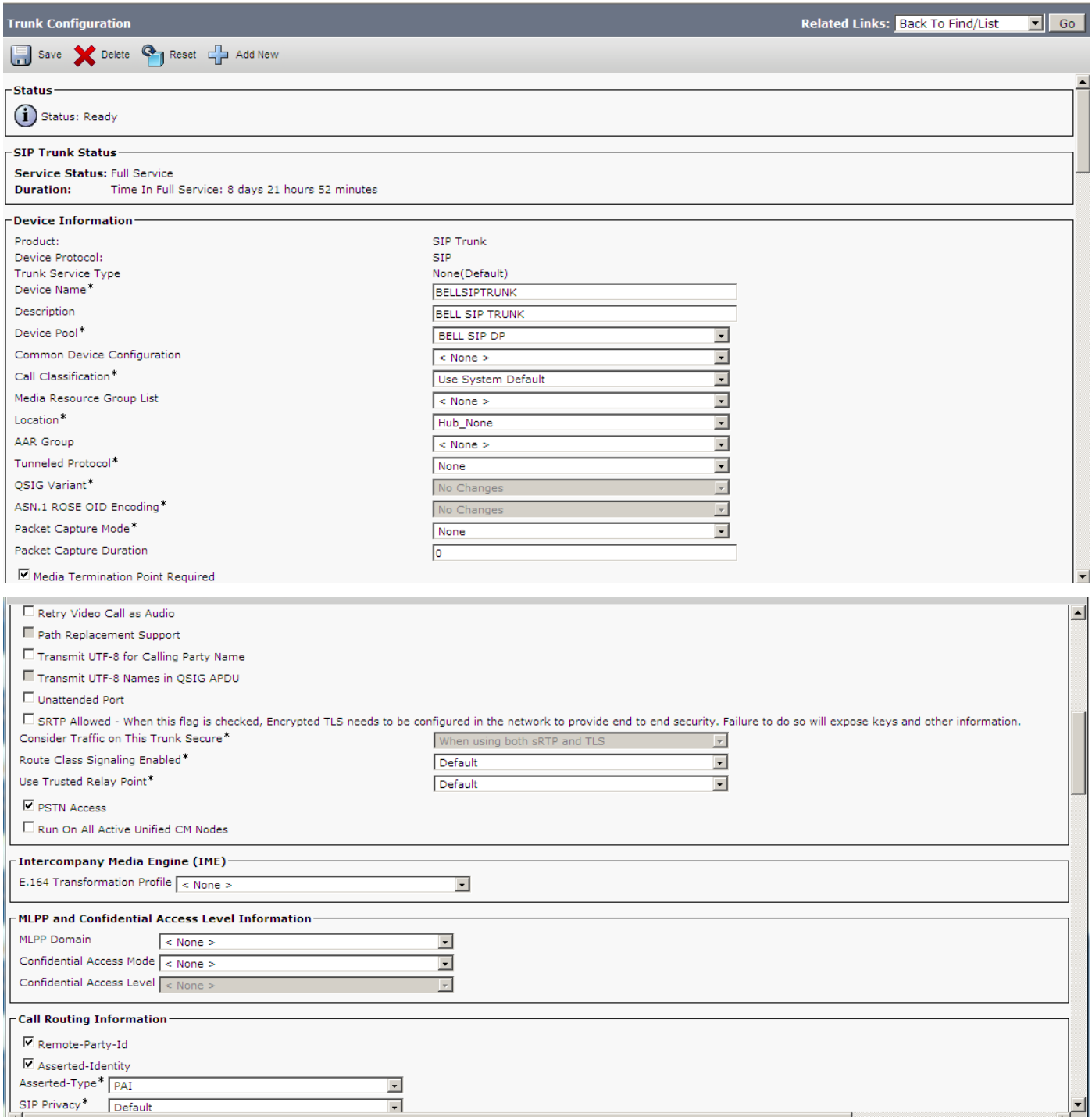


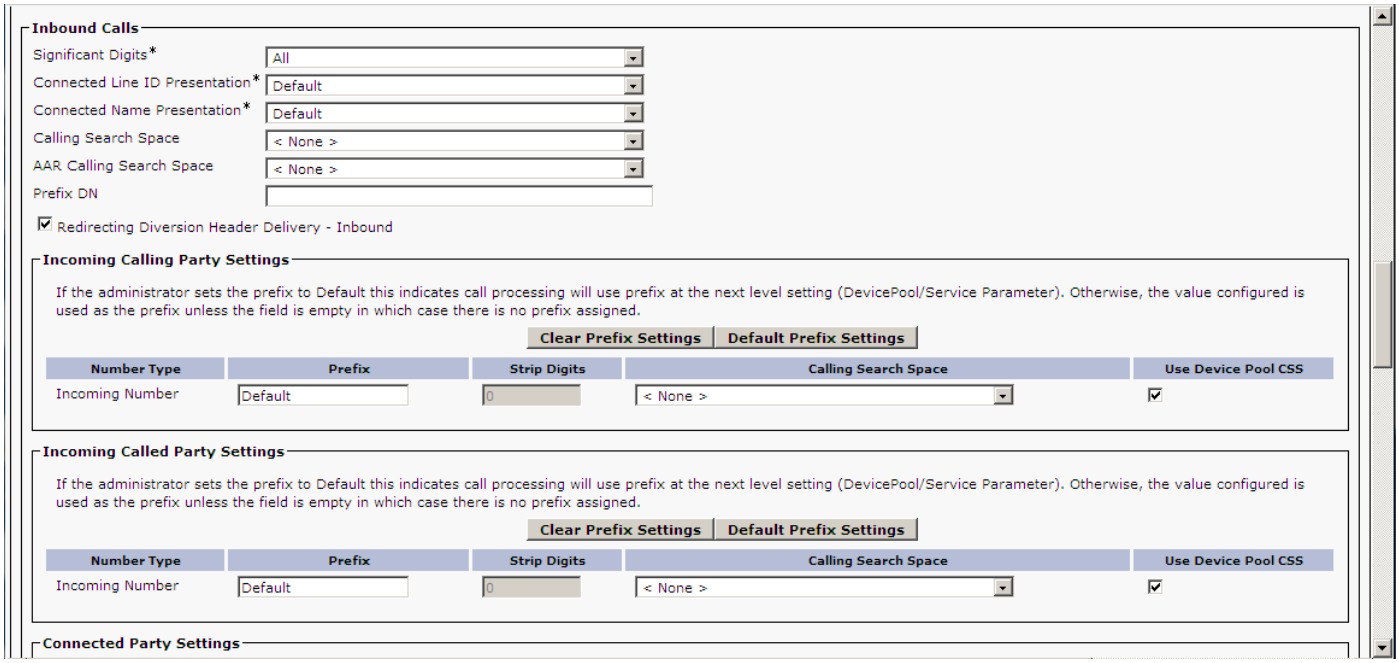


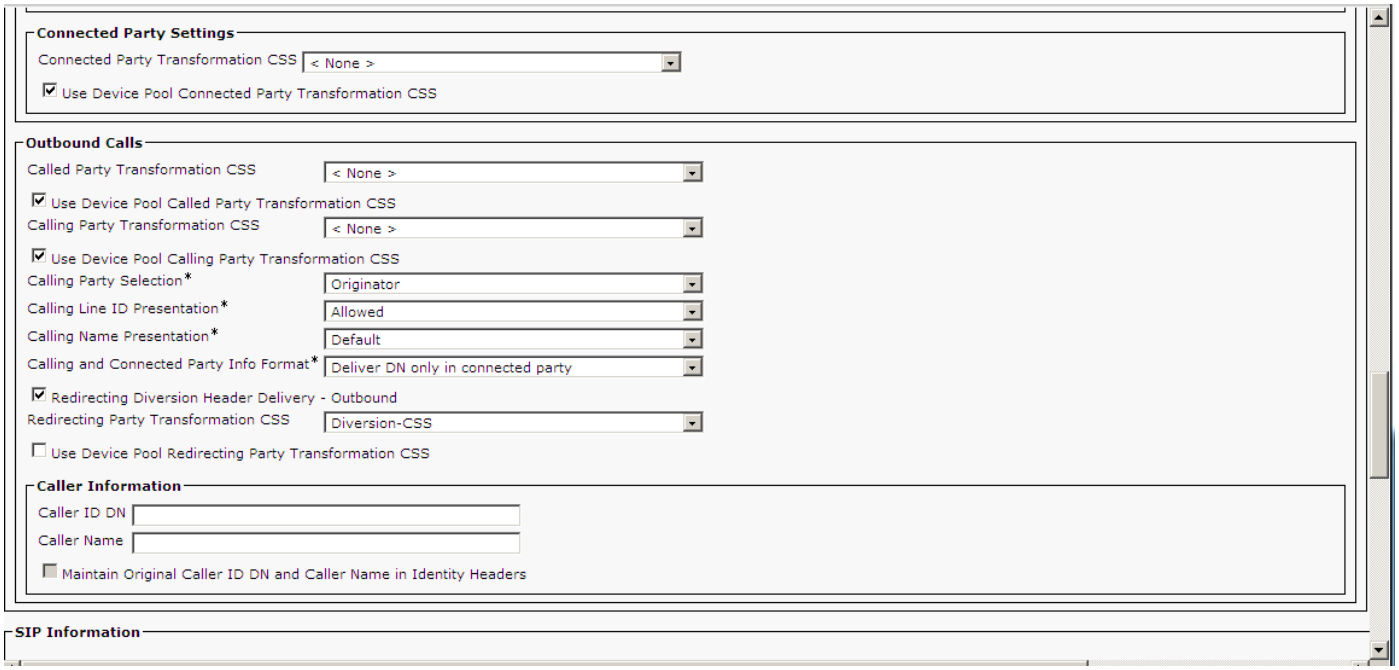
Configure a SIP Trunk

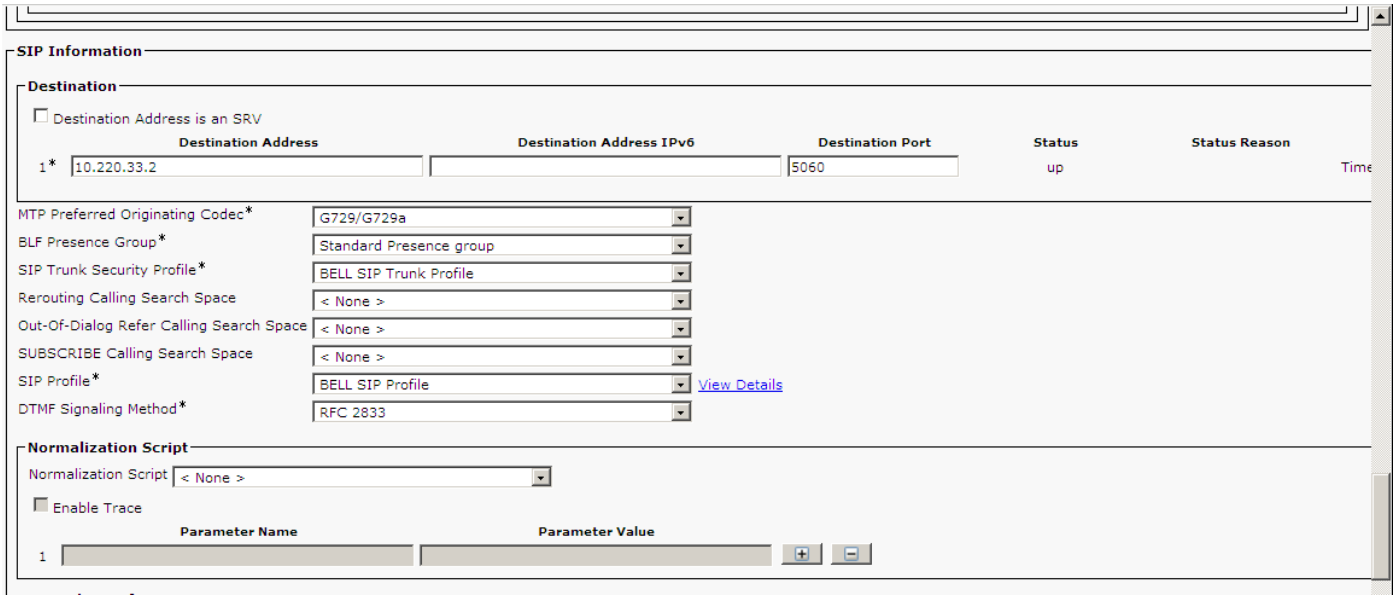
Navigate to the Device → Trunk page in the CUCM portal.

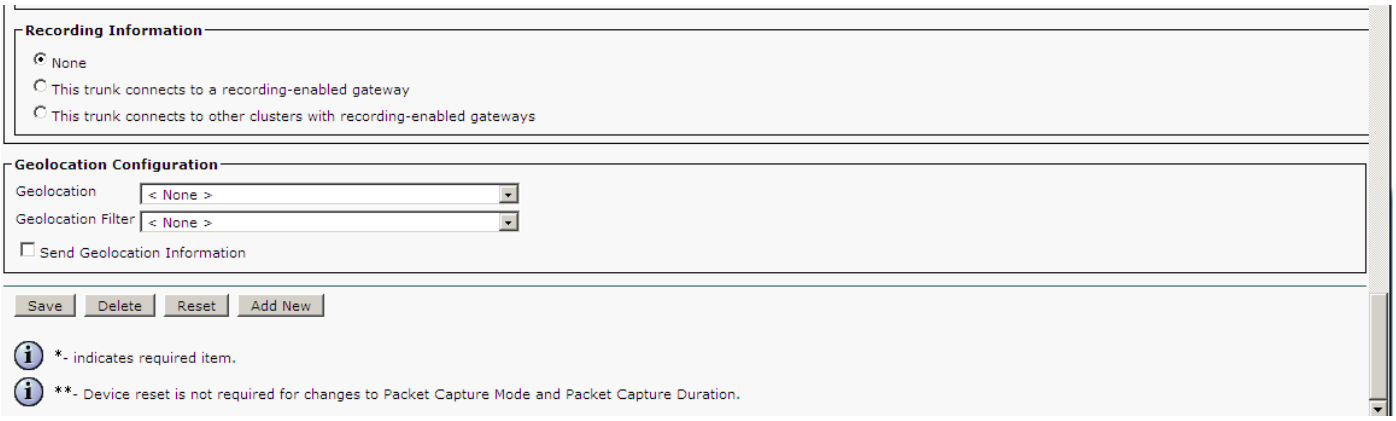
- Click on "Add New"
- Select "SIP Trunk" as the trunk type.
- The Device Protocol will automatically be populated with "SIP", just click Next
- Configure as below screen shot.
- Save the configuration.

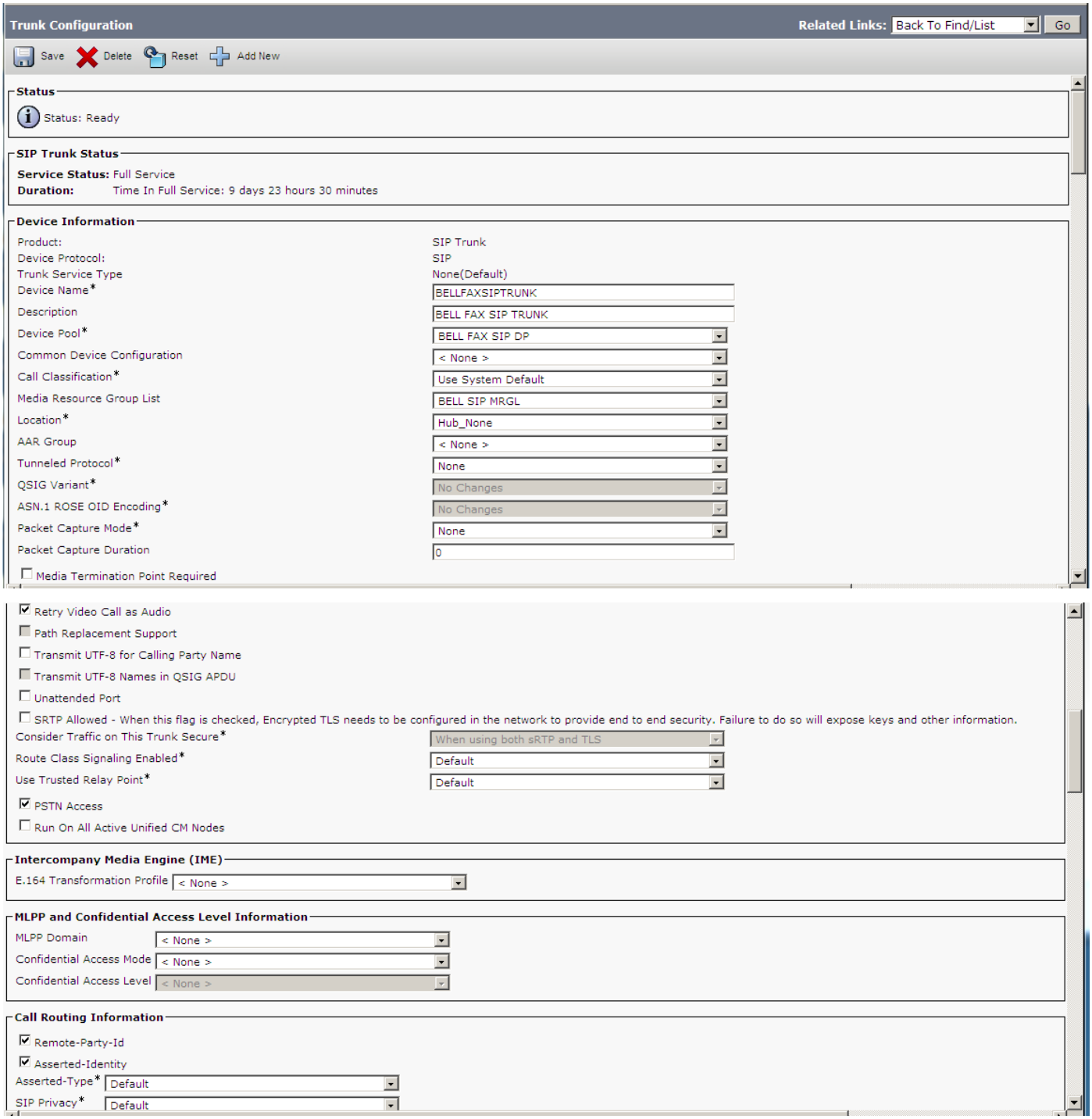












Inbound Calls

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 Settings

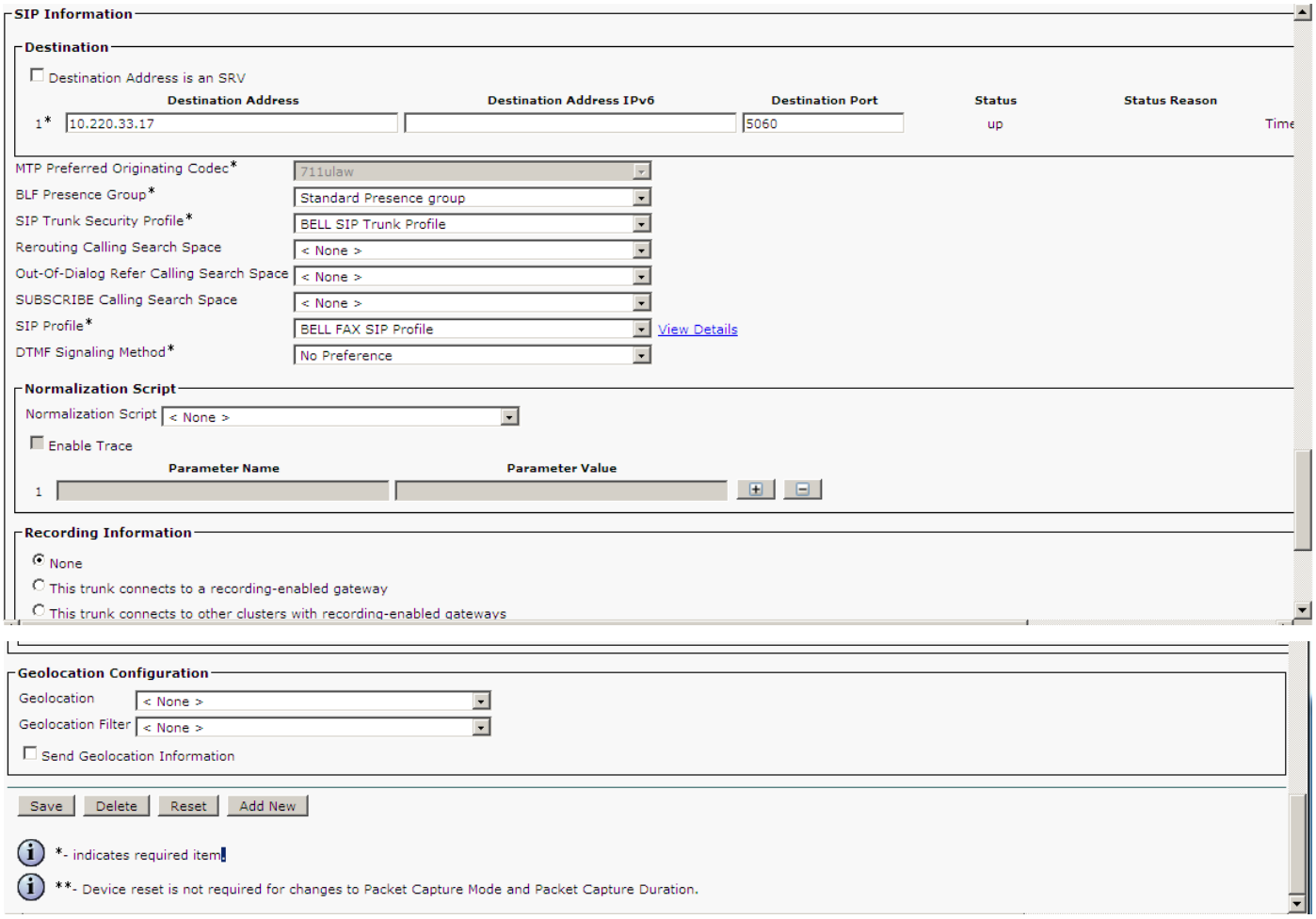
Connected Party Transformation CSS < None >
☑ Use Device Pool Connected Party Transformation CSS

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

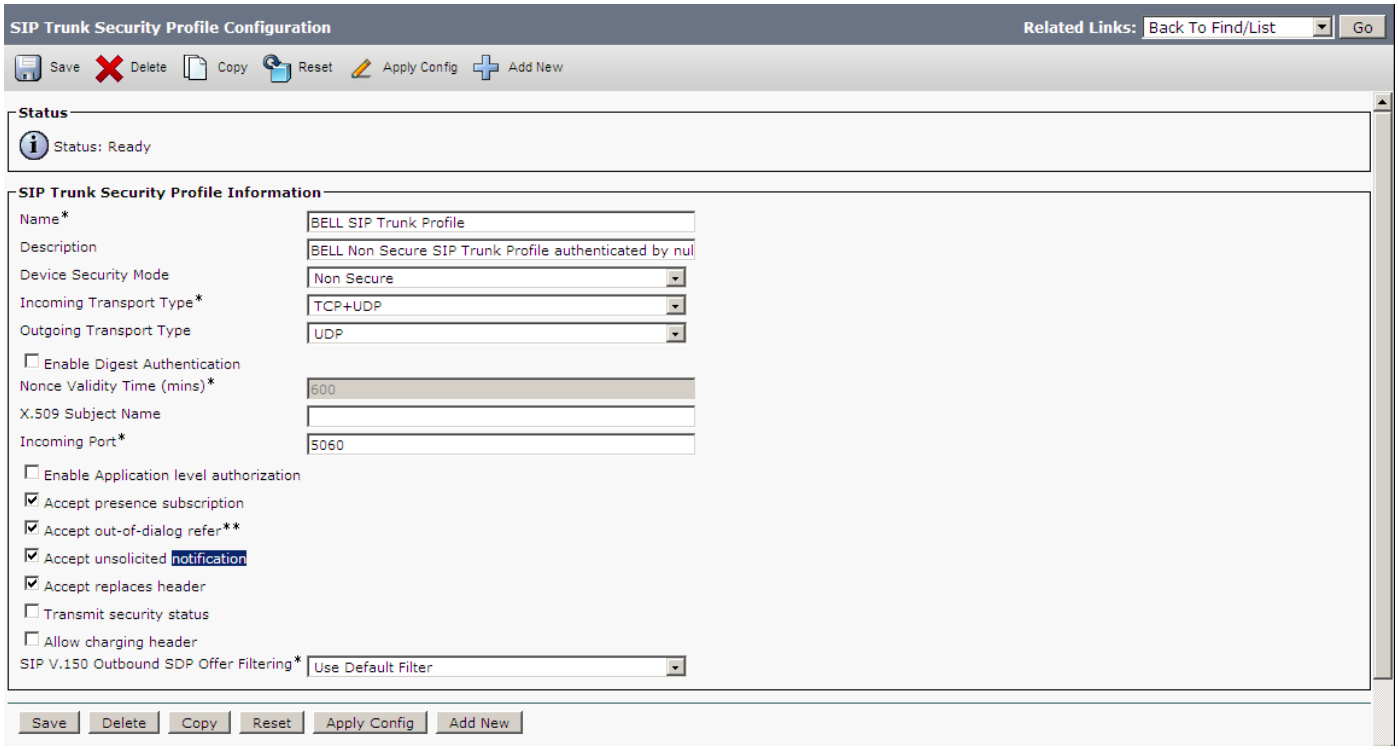
Caller Information

Caller ID DN
Caller Name
☐ Maintain Original Caller ID DN and Caller Name in Identity Headers



Configure a SIP Trunk Security Profile

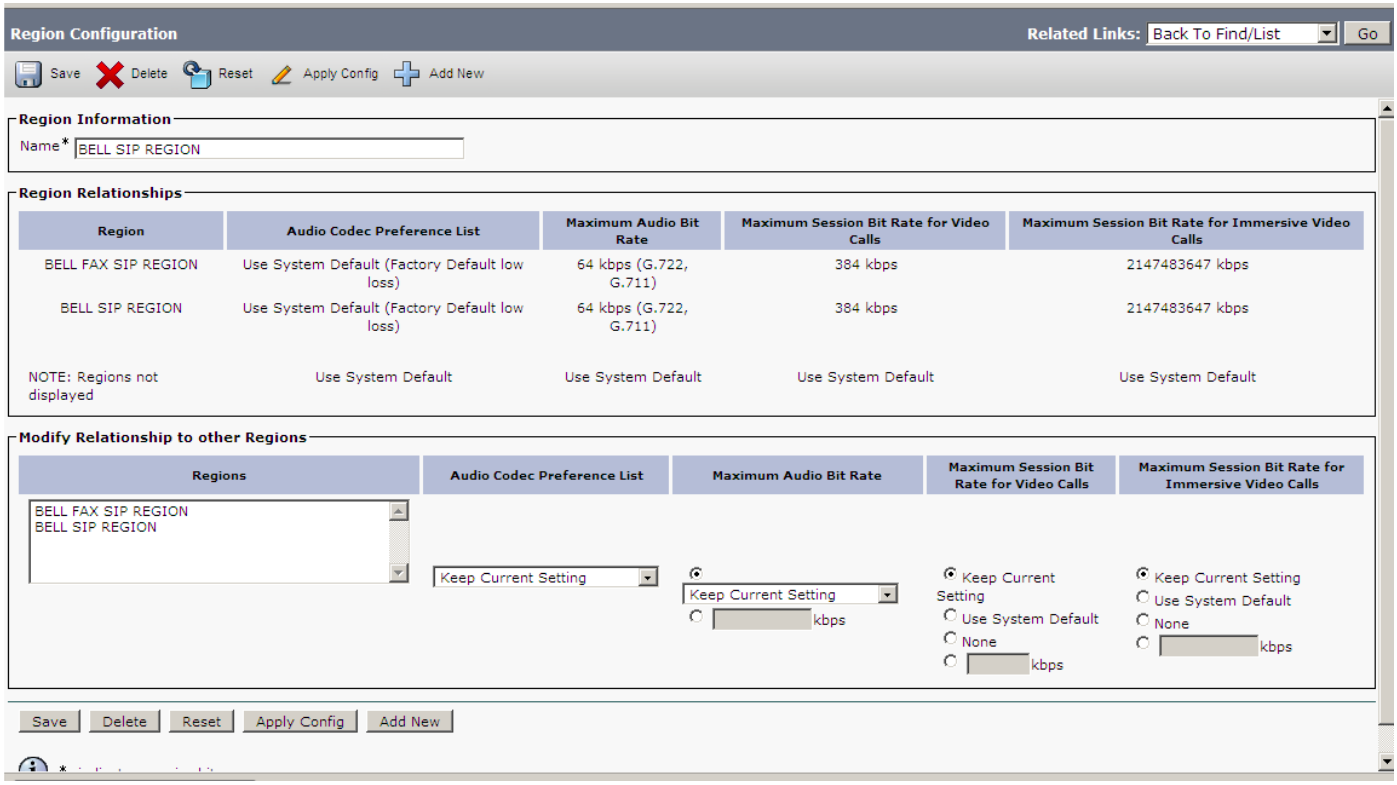
- Navigate to the System → Security Profile → SIP Trunk Security Profile page in the portal (see screen capture below):
- Copy the existing Non-Secure SIP Trunk Profile.
- Name the New SIP Trunk Profile.
- Save the configuration.

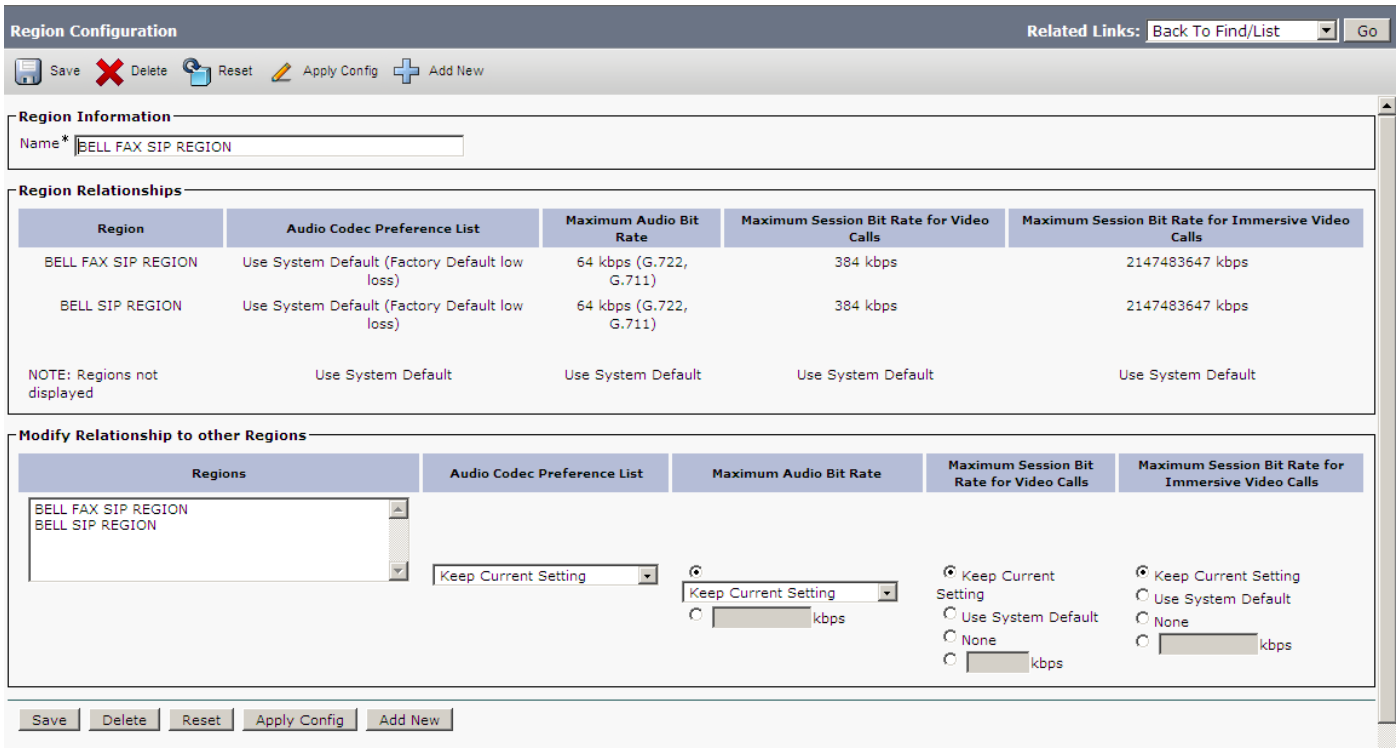


Configure a Region

Navigate to the System → Region Information → Region page in the CUCM portal.

- Click on the "Add New" button
- Name the new region.
- Click on the newly created region and choose Audio Codec Preference List and Maximum Audio Bit Rate to 64 kbps (G.722, G.711)
- Select other settings to be consistent with the customer‟s environment.
- Save the configuration.

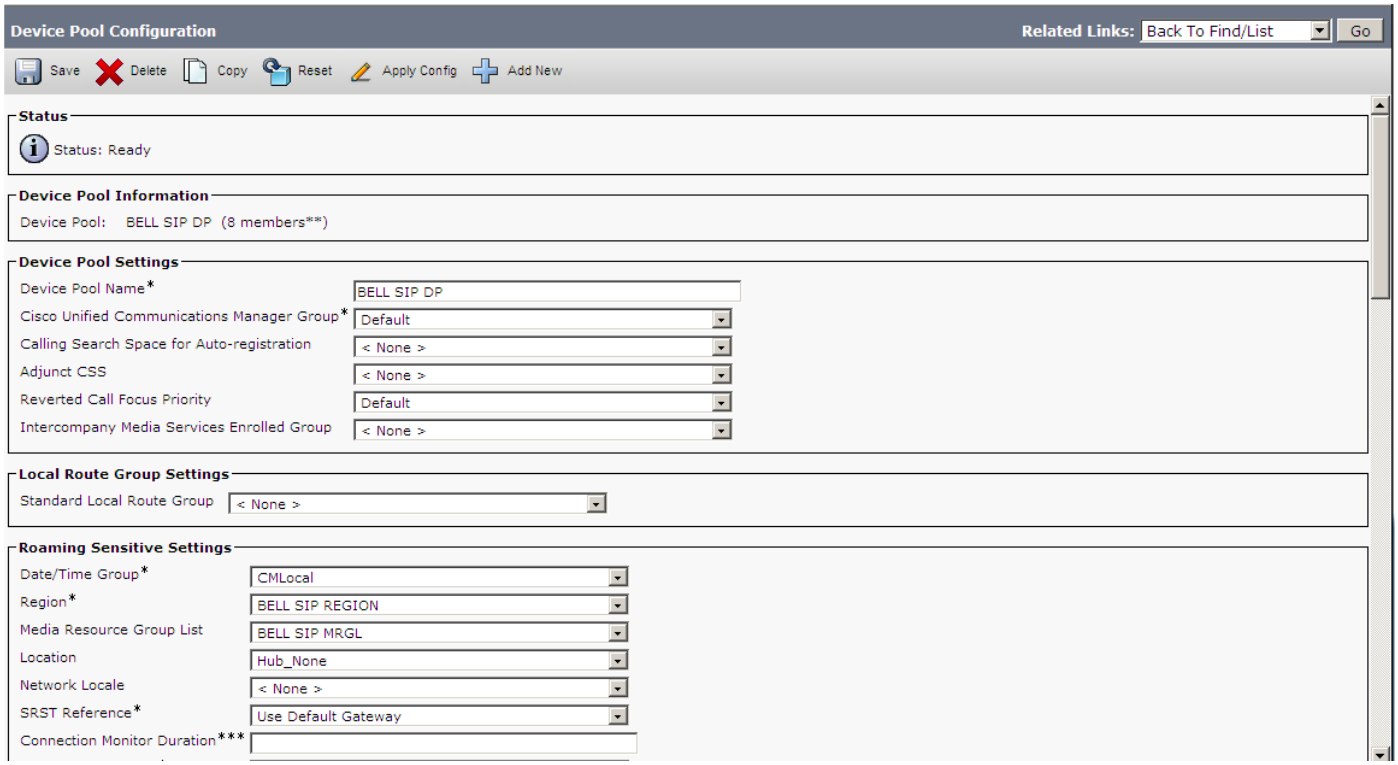


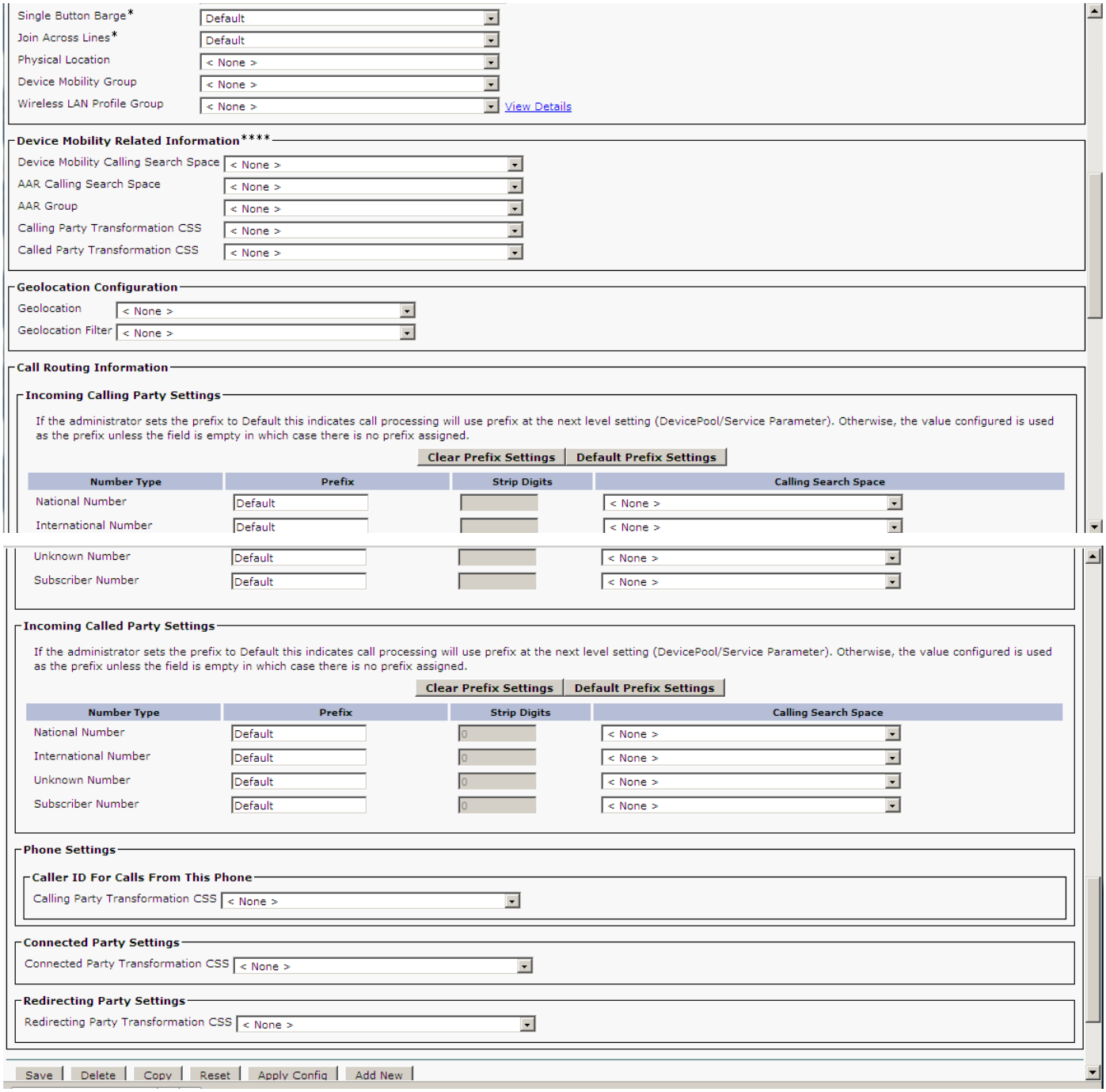


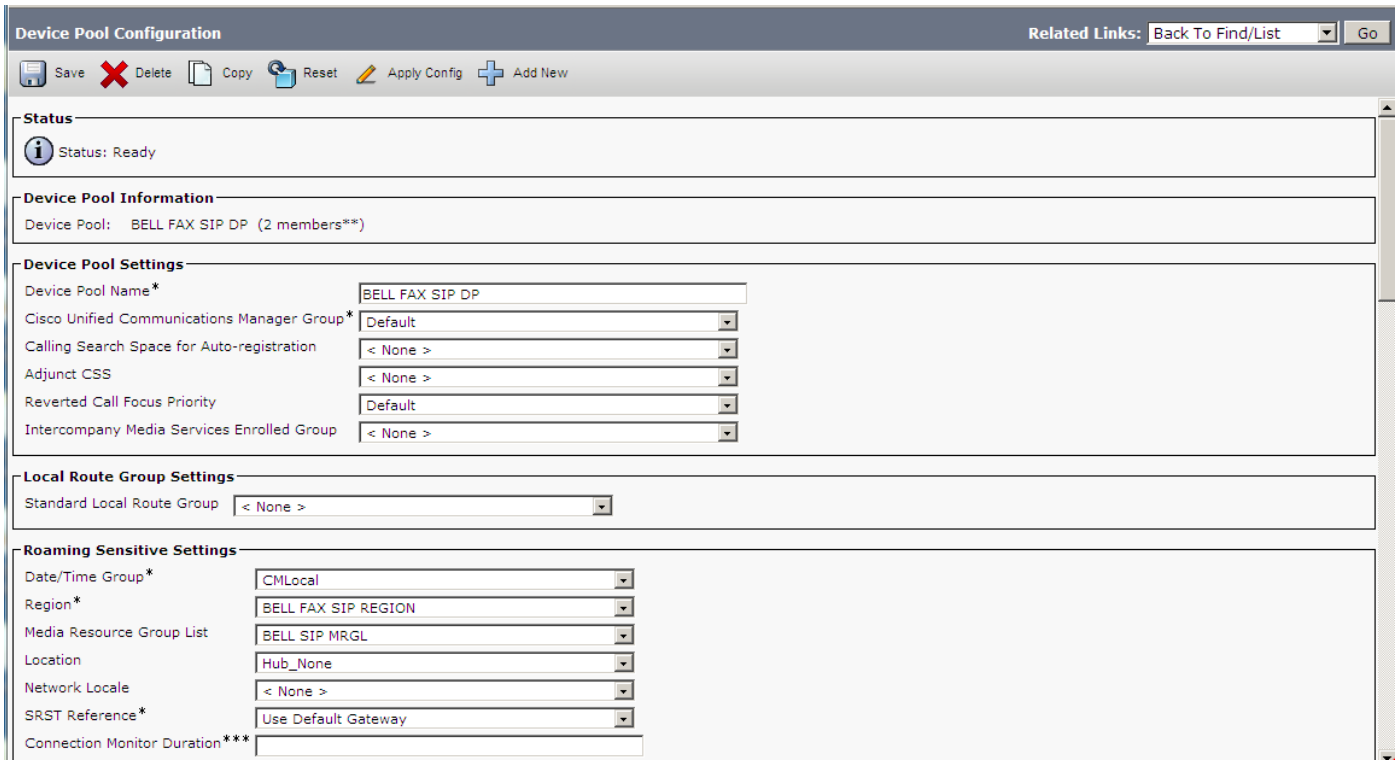
Configure a Device pool:

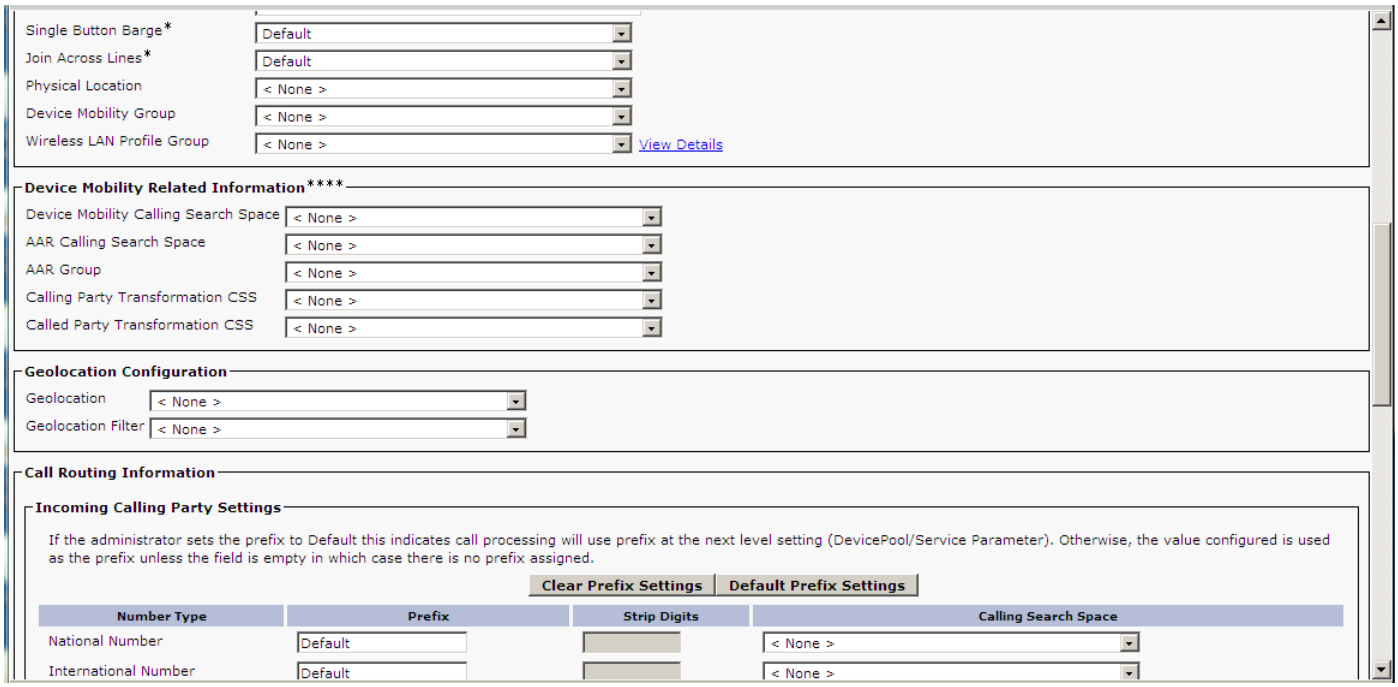
Navigate to the System → Device Pool page in the CUCM portal.

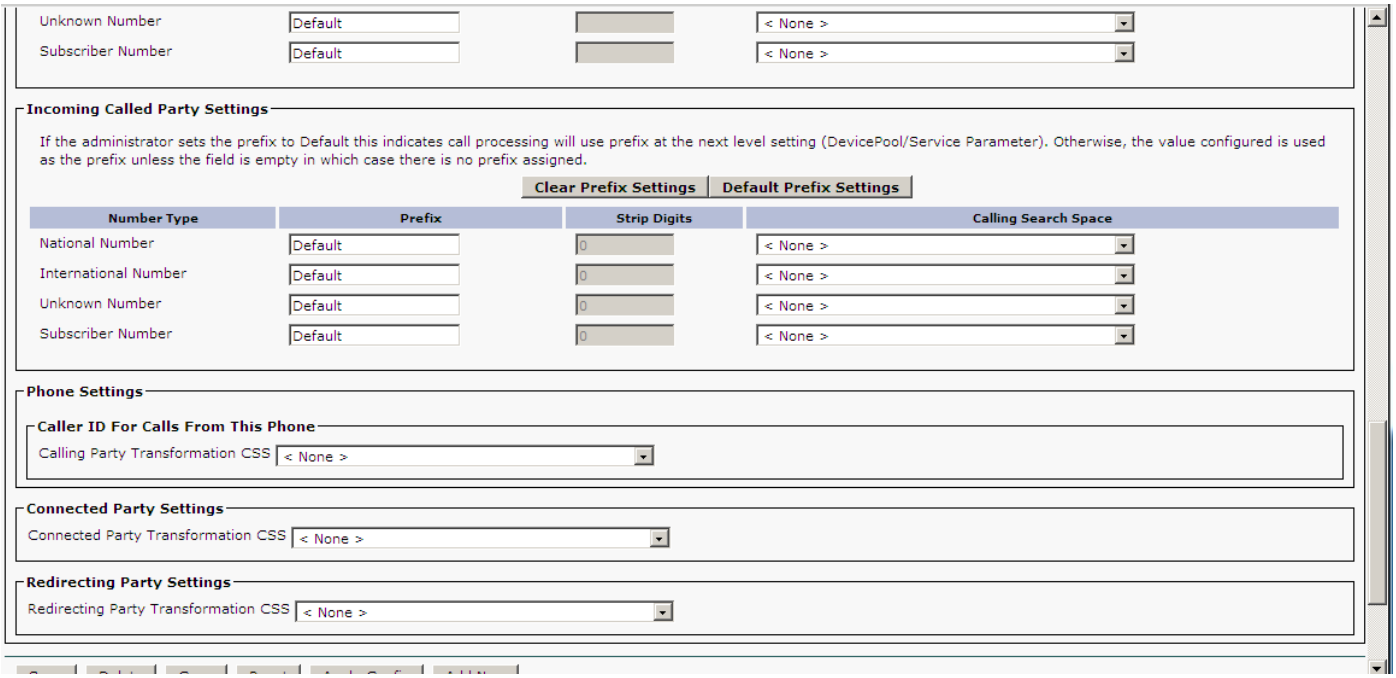
- Click on "Add New"
- Name the new Device Pool.
- Select desired Region created in the previous step)
- Fill in required fields with appropriate configurations for your customer environment.
- Save the configuration.









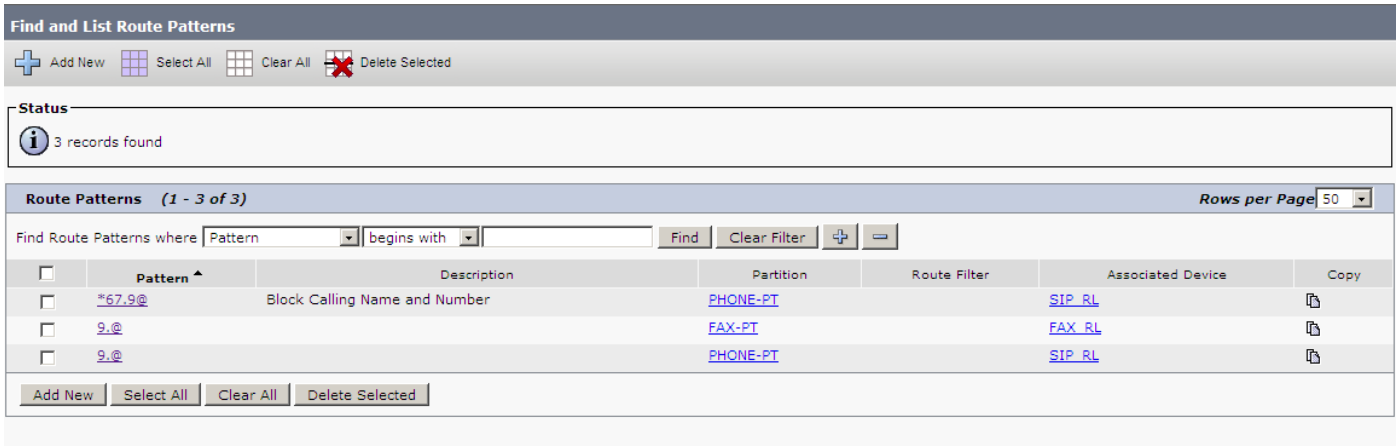


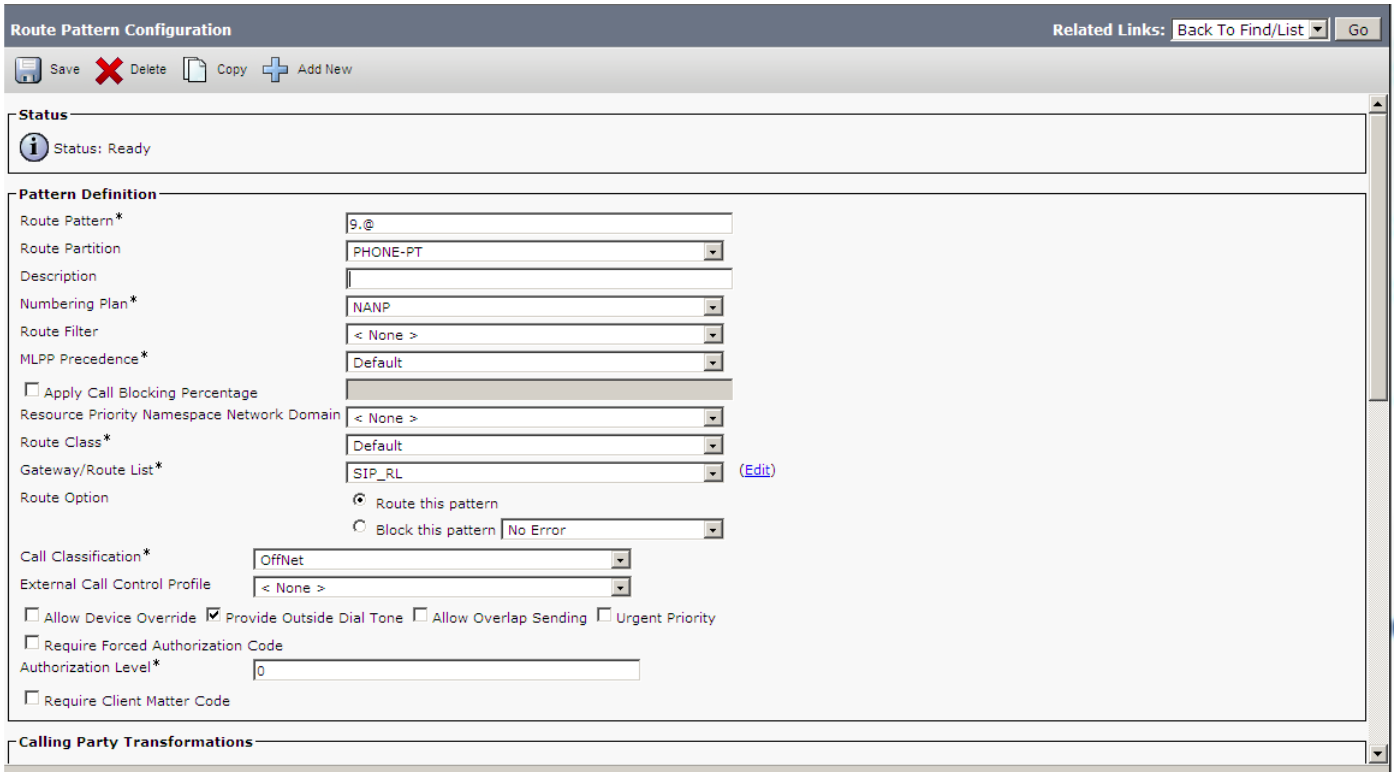
Configure Route groups, Route Lists, and Route Patterns

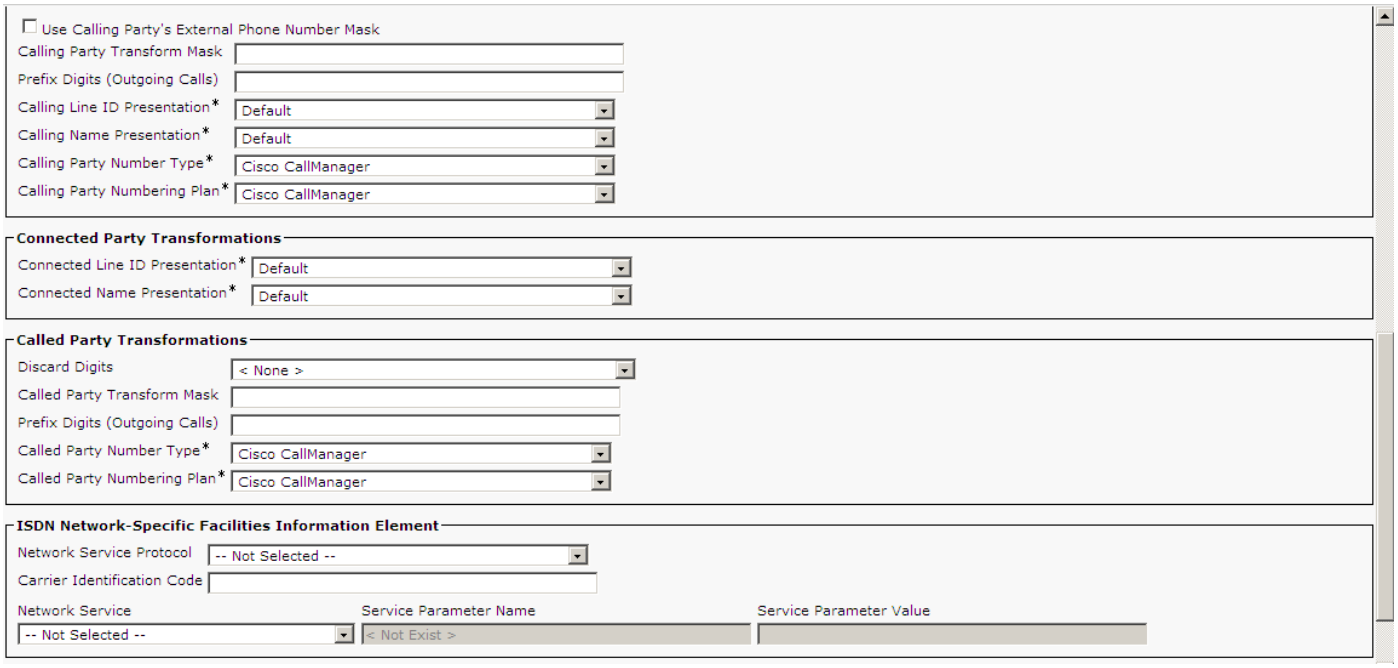
Due to the complexity and the wide variance in the individual needs of each enterprise, configuring enterprise route patterns requires detailed planning and consideration. For this reason, route patterns are not covered in any detail in this reference document. Please refer to the appropriate Cisco documentation for guidance in configuring Route Patterns suitable to your particular situation.

For the sake of a simple example, a simple Route Pattern that routes calls to a CUBE is defined here:

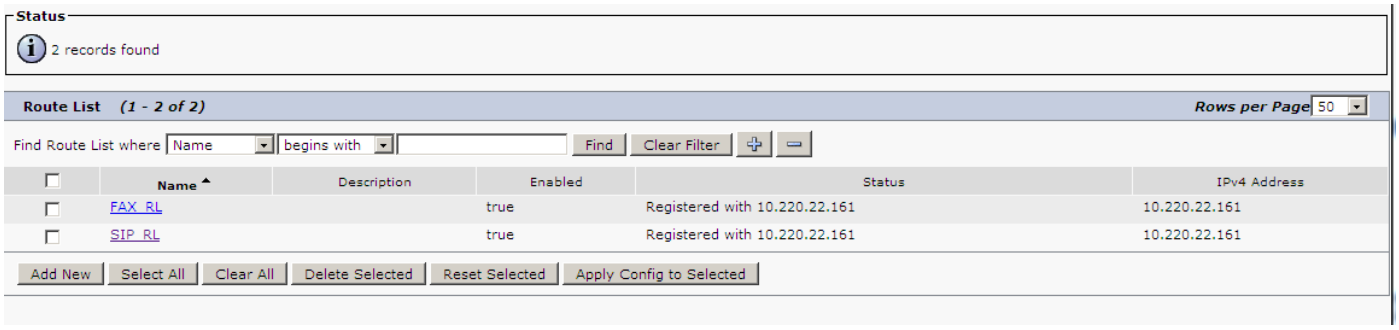
- Navigate to Call routing → Route/Hunt → Route Pattern
- Click on "Add New"
- Insert route-pattern expression
- Choose a route partition
- Give it a description describing the intention of this route pattern.
- Choose the Gateway/route list as per provisioned.
- Define any other trunk specific configurations.
- Save the configuration

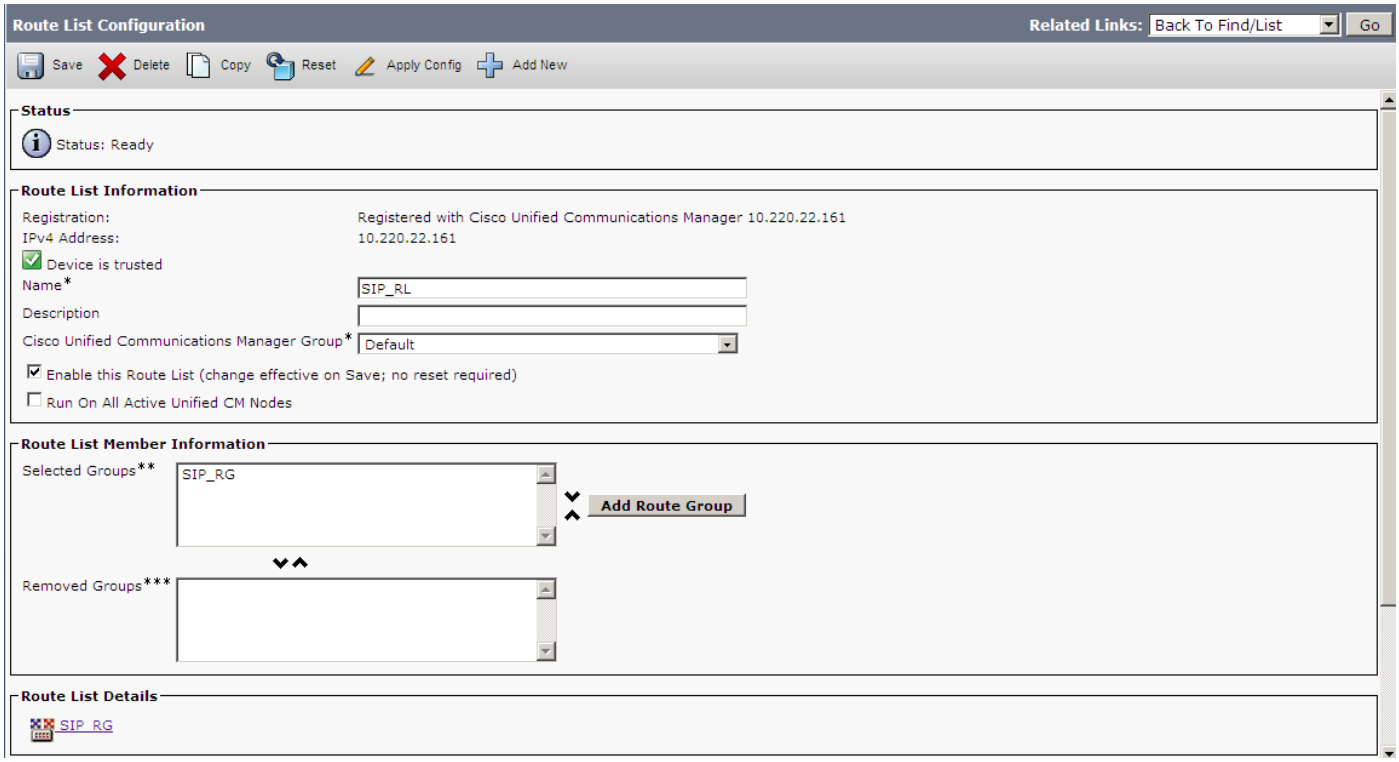


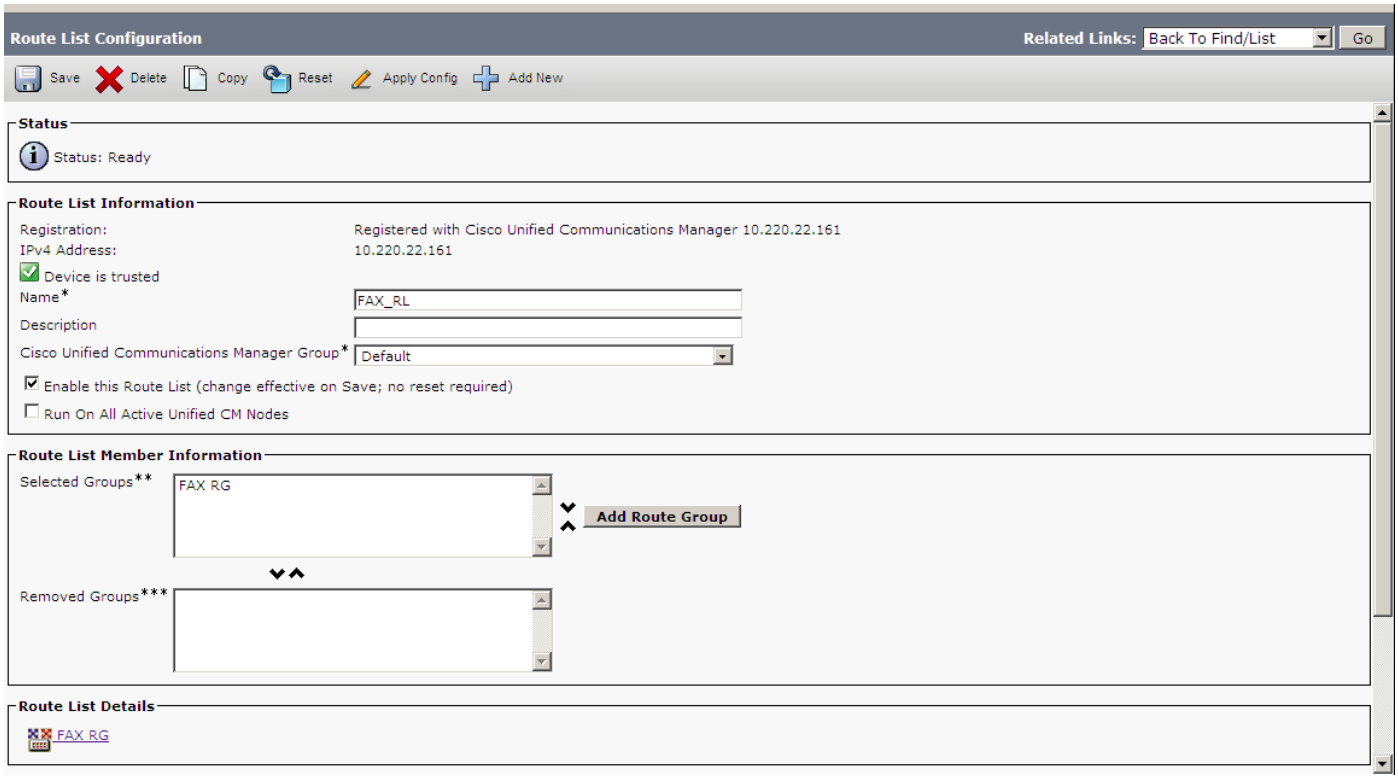




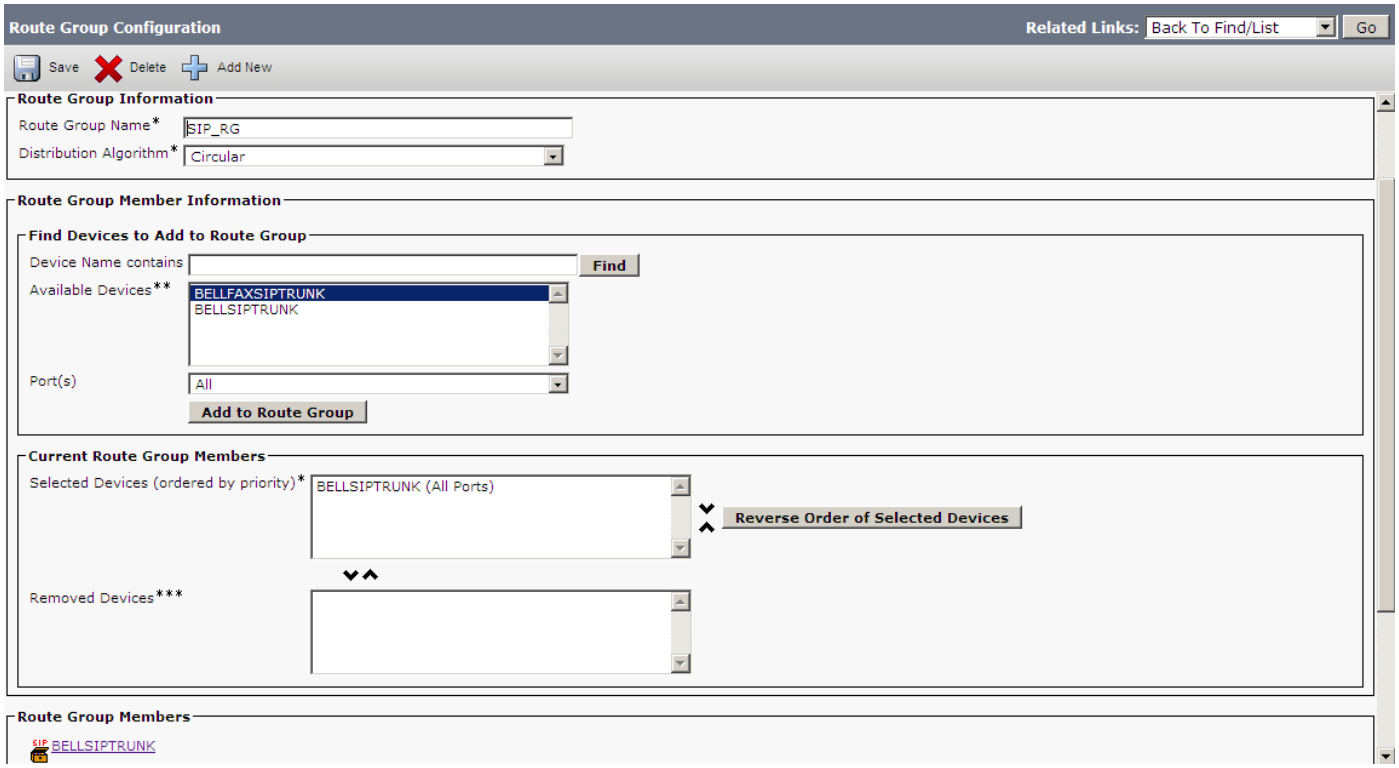
Route List Configuration

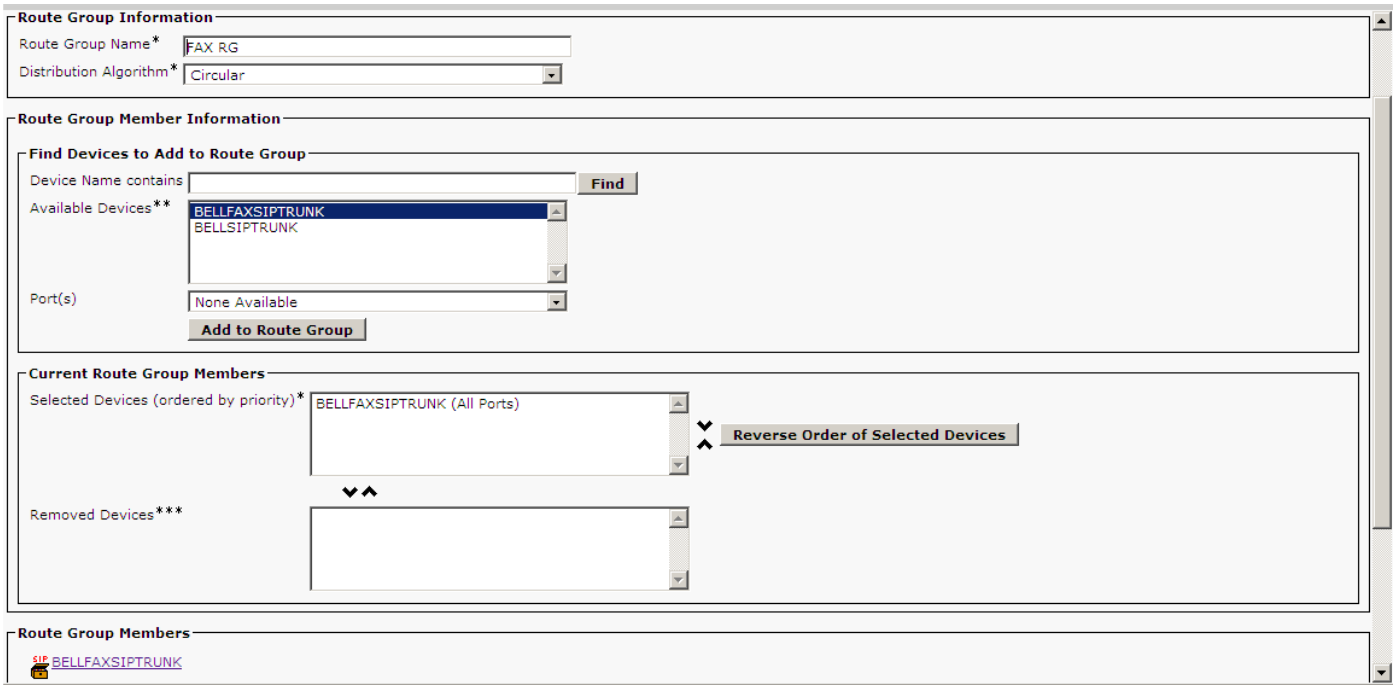




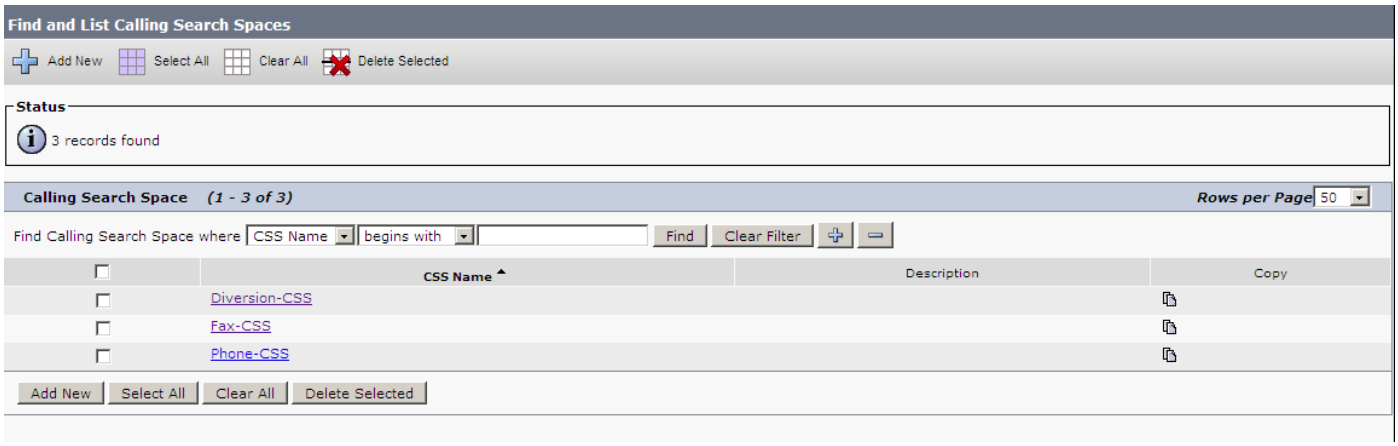


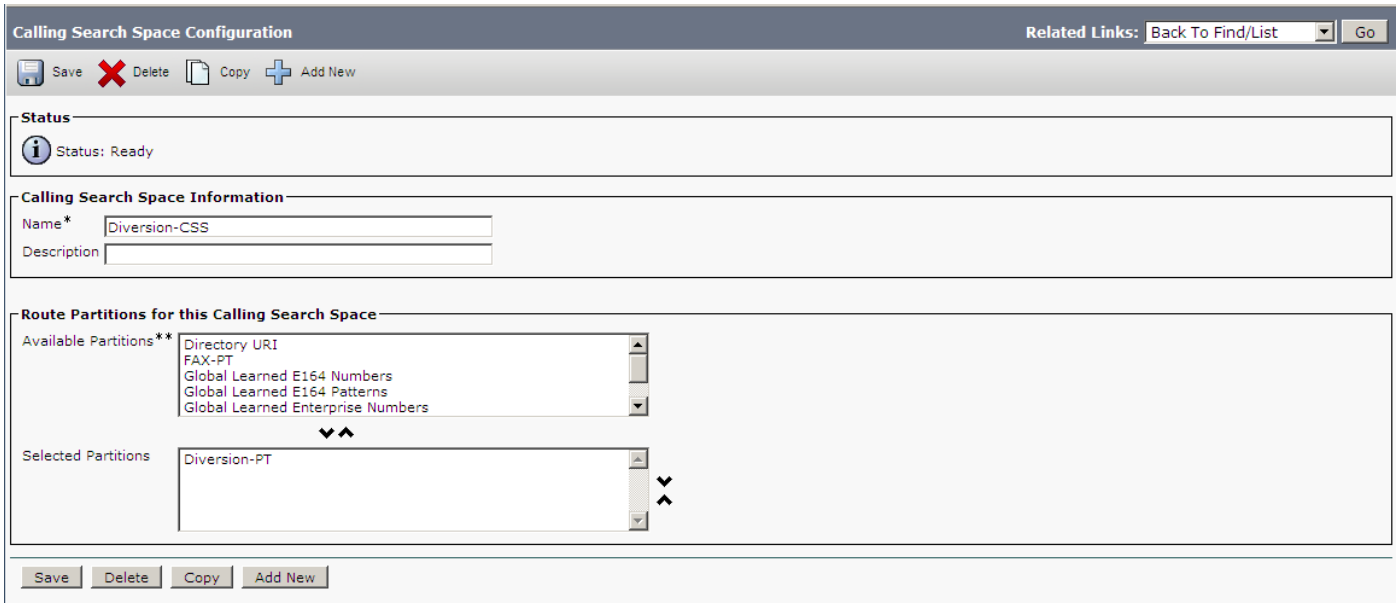
Route Group Configuration

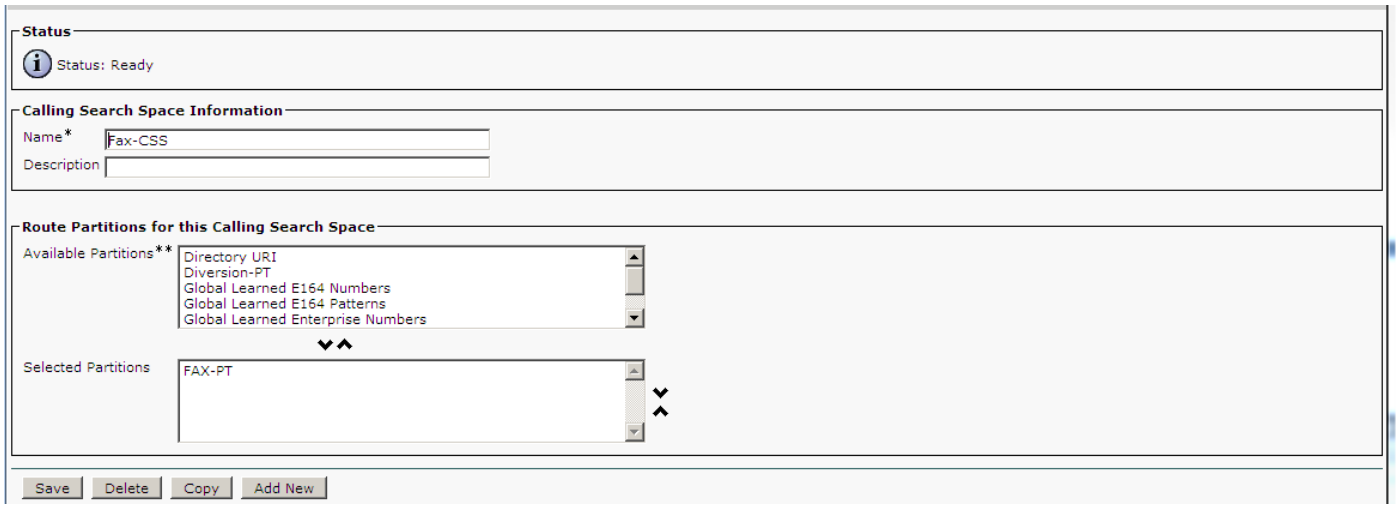


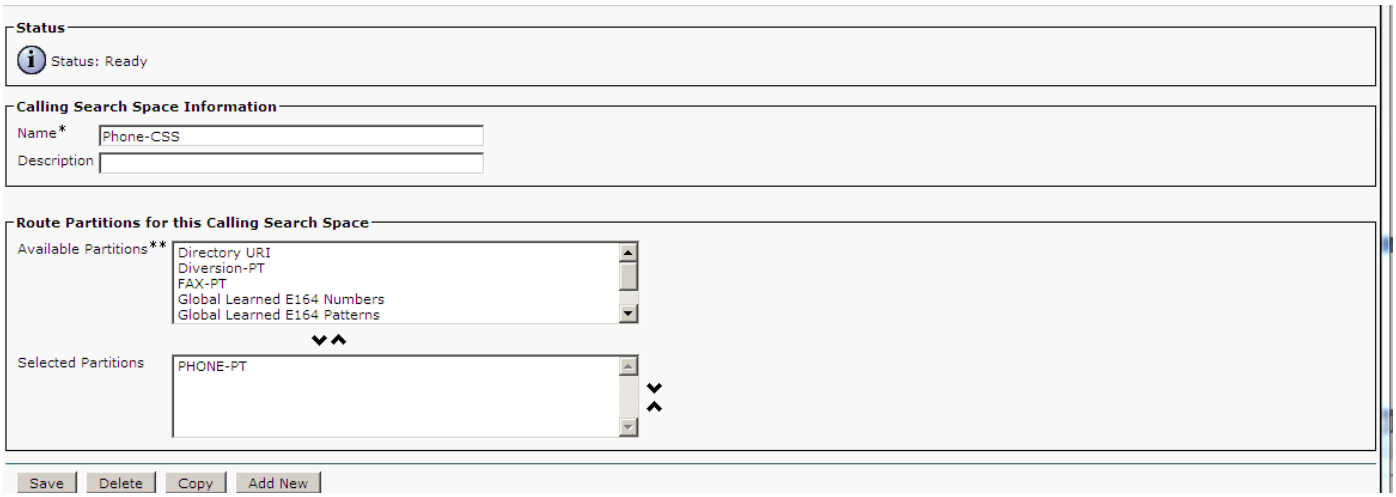


Calling Search Space Configuration

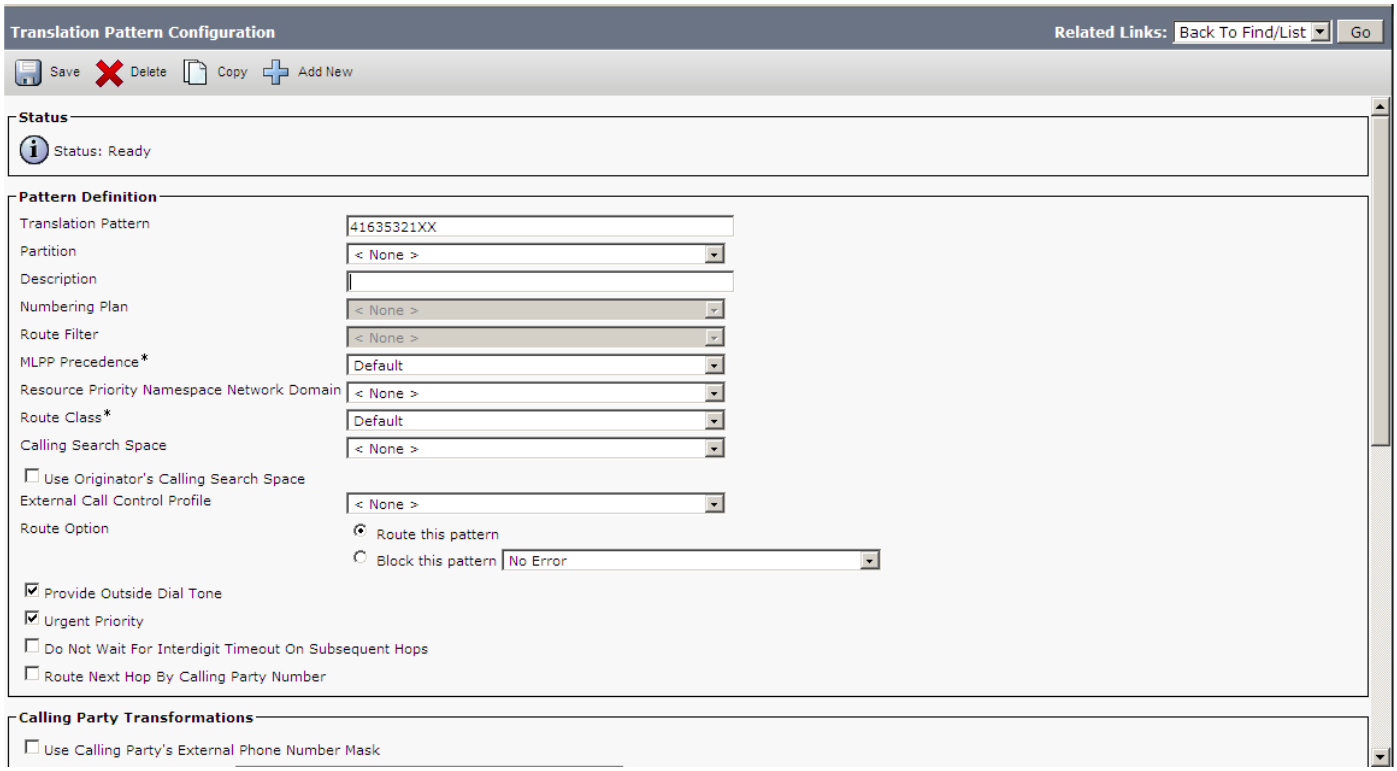


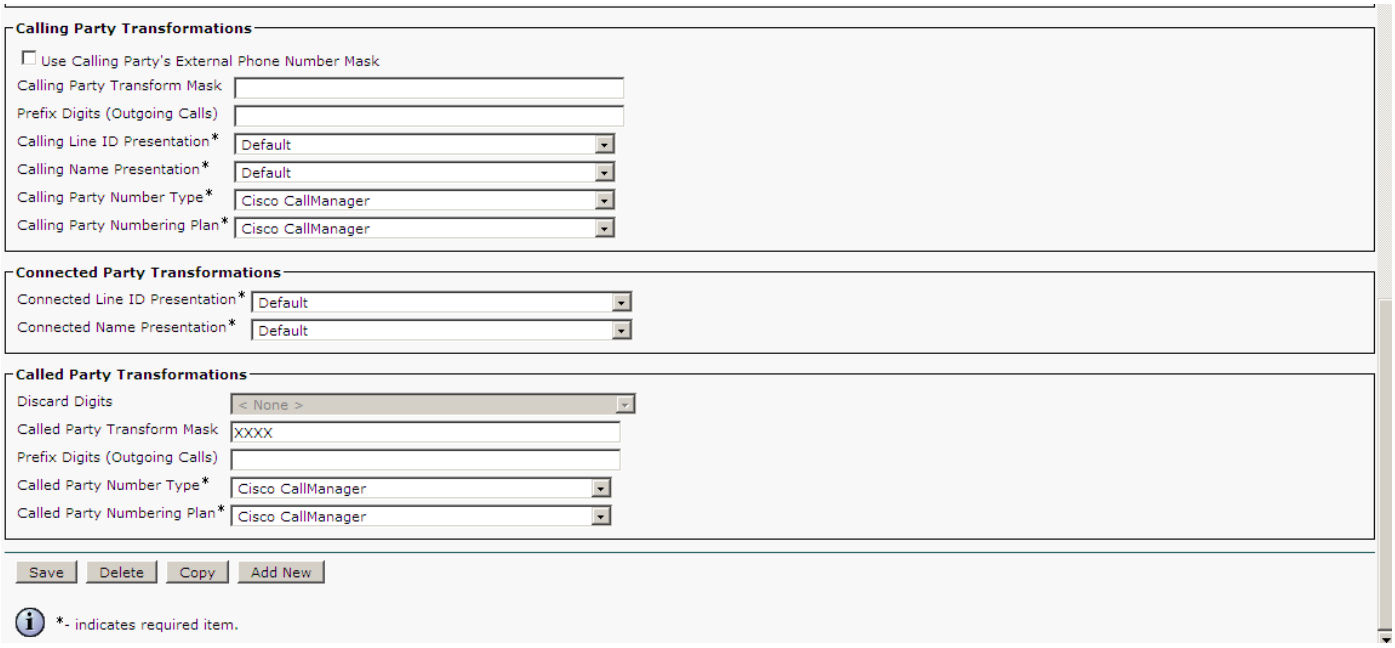




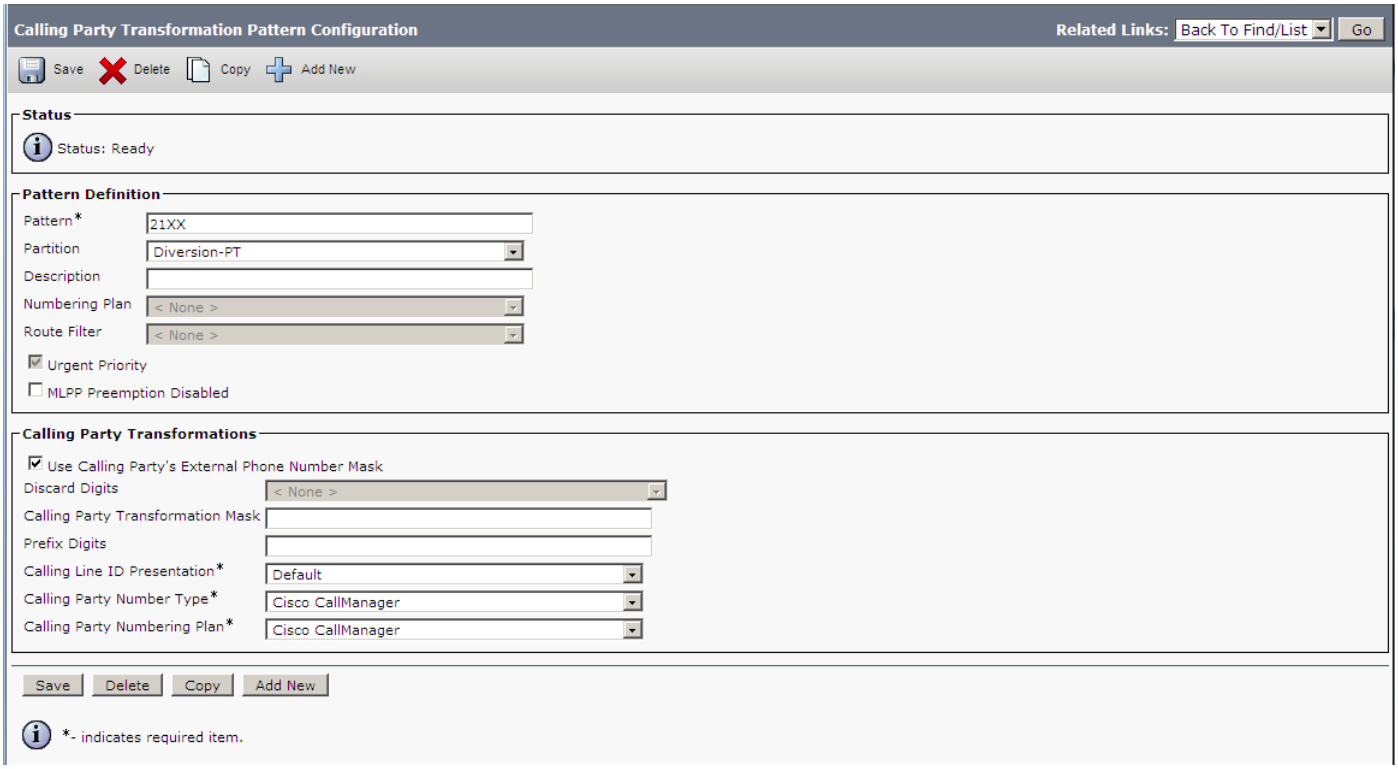


Translation Pattern Configuration

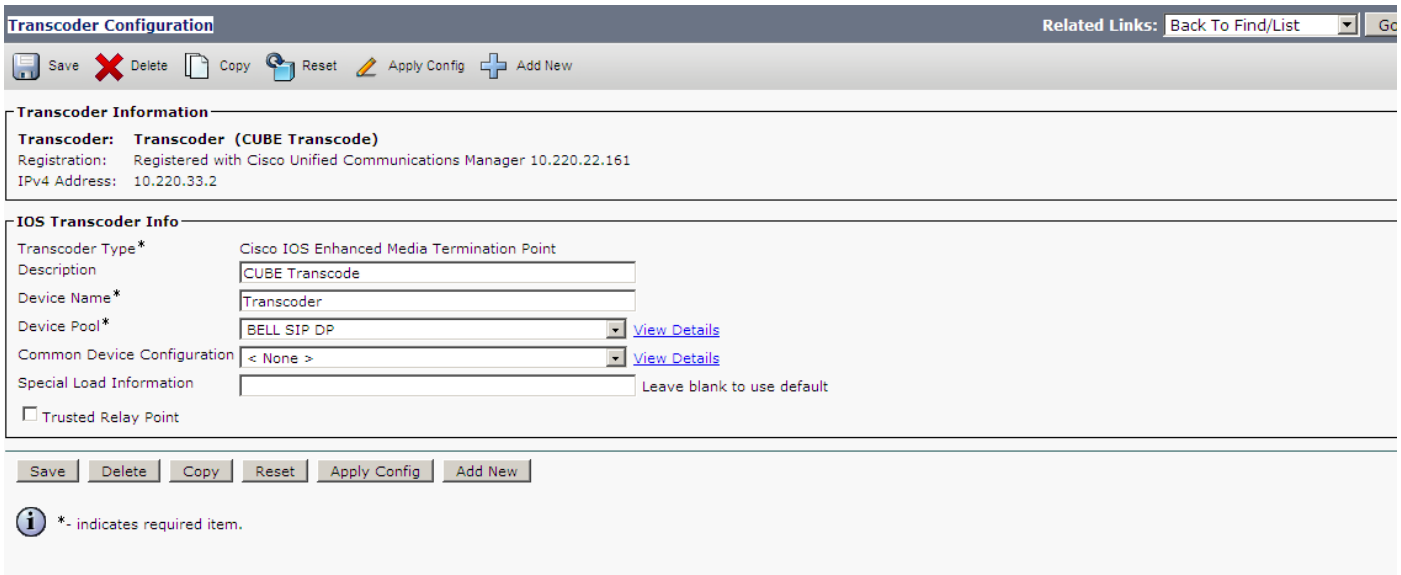




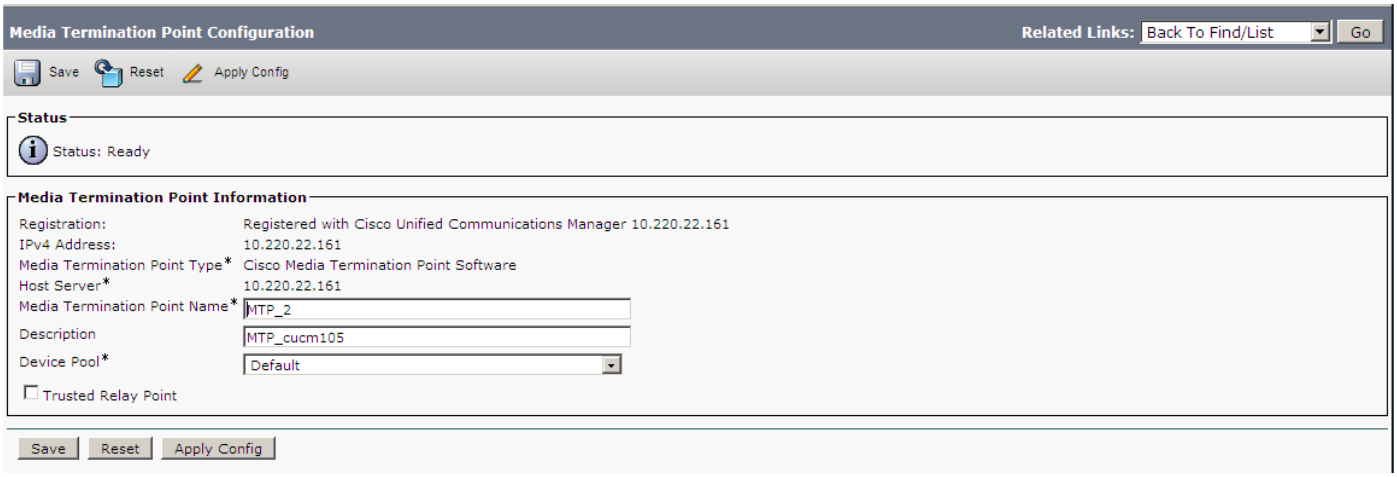
Calling Party Transformation Pattern Configuration



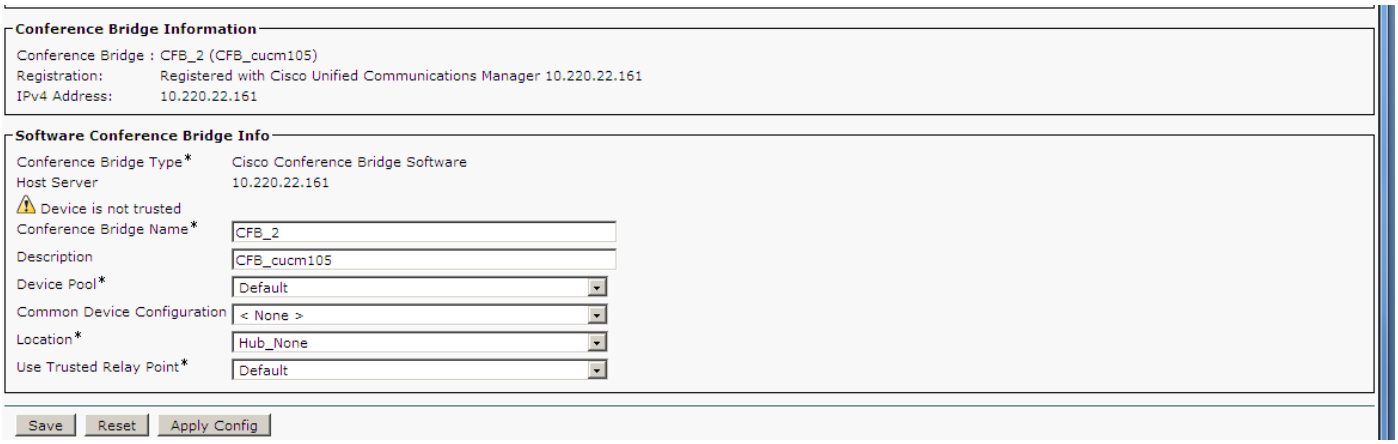
Transcoder Configuration



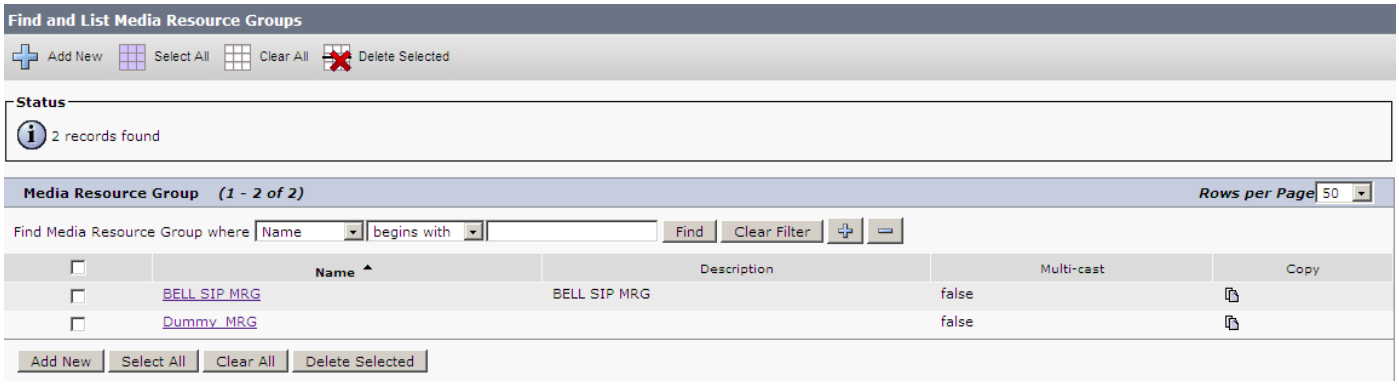
Media Termination Point

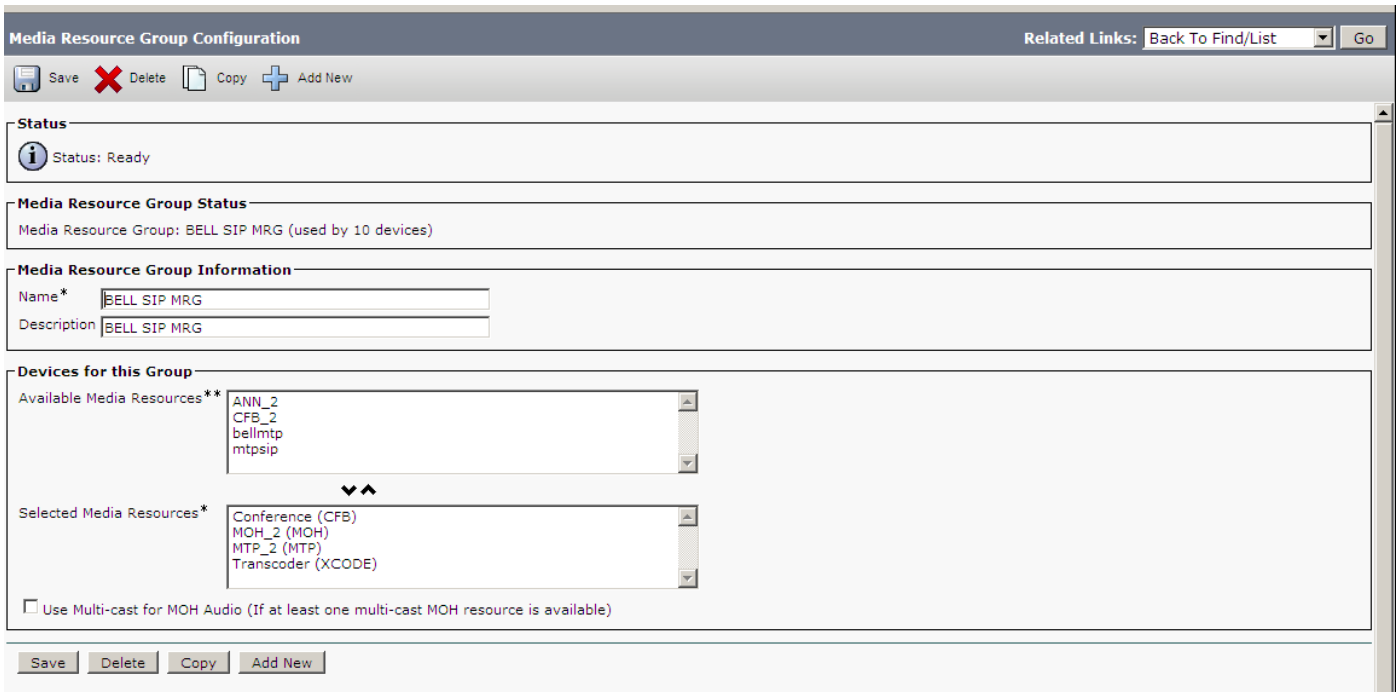


Conference Bridge Configuration

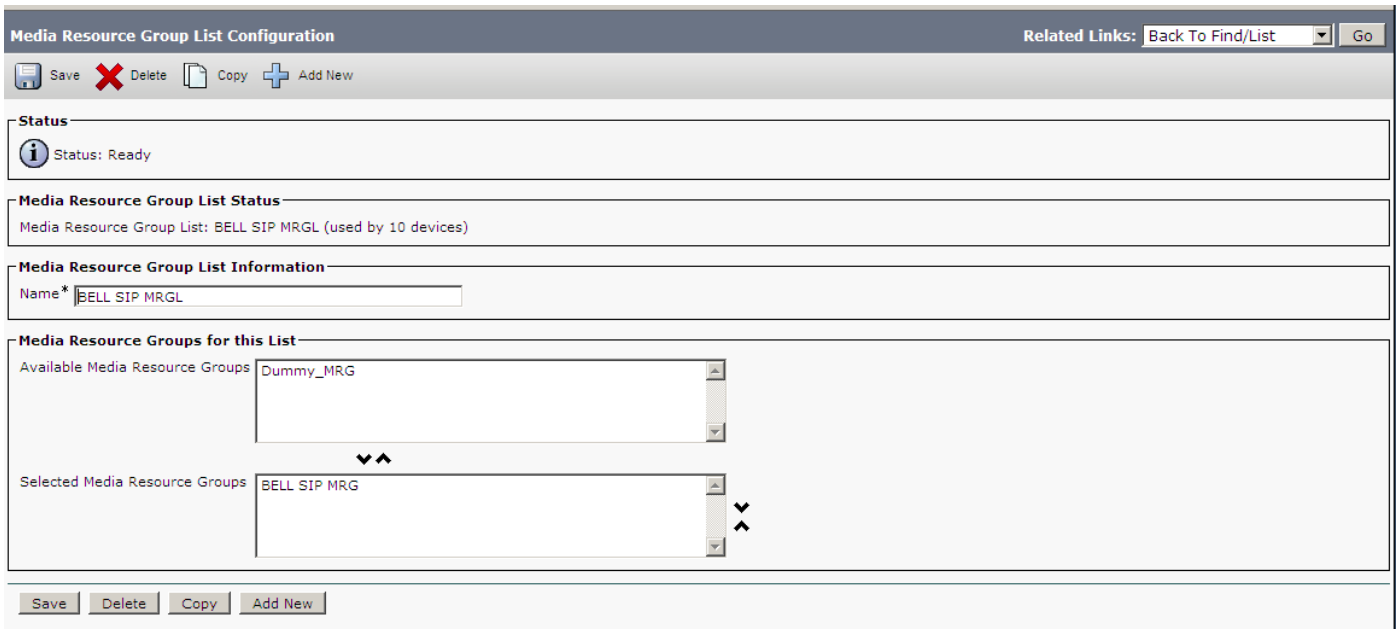


Media Resource Group Configuration



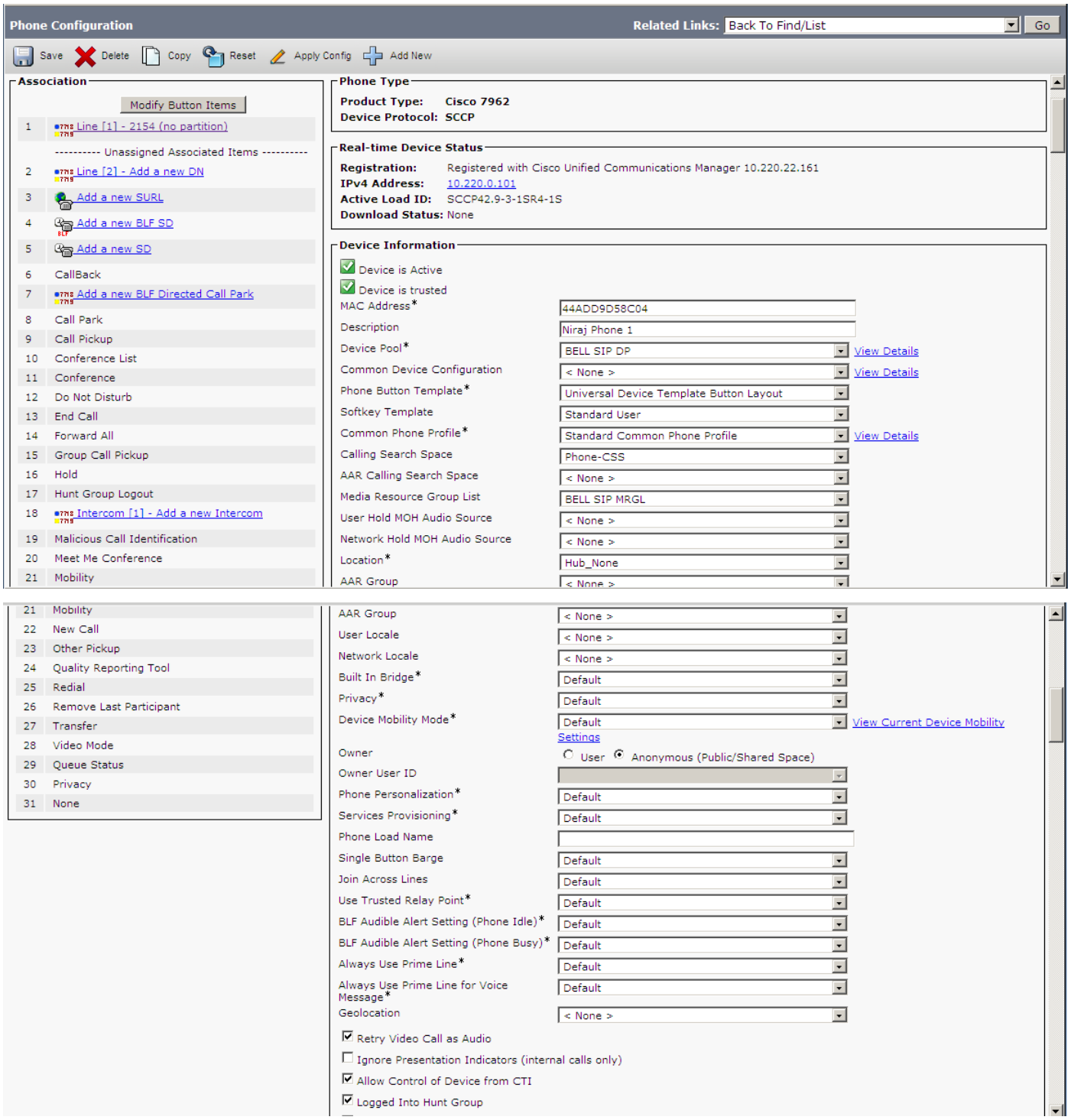


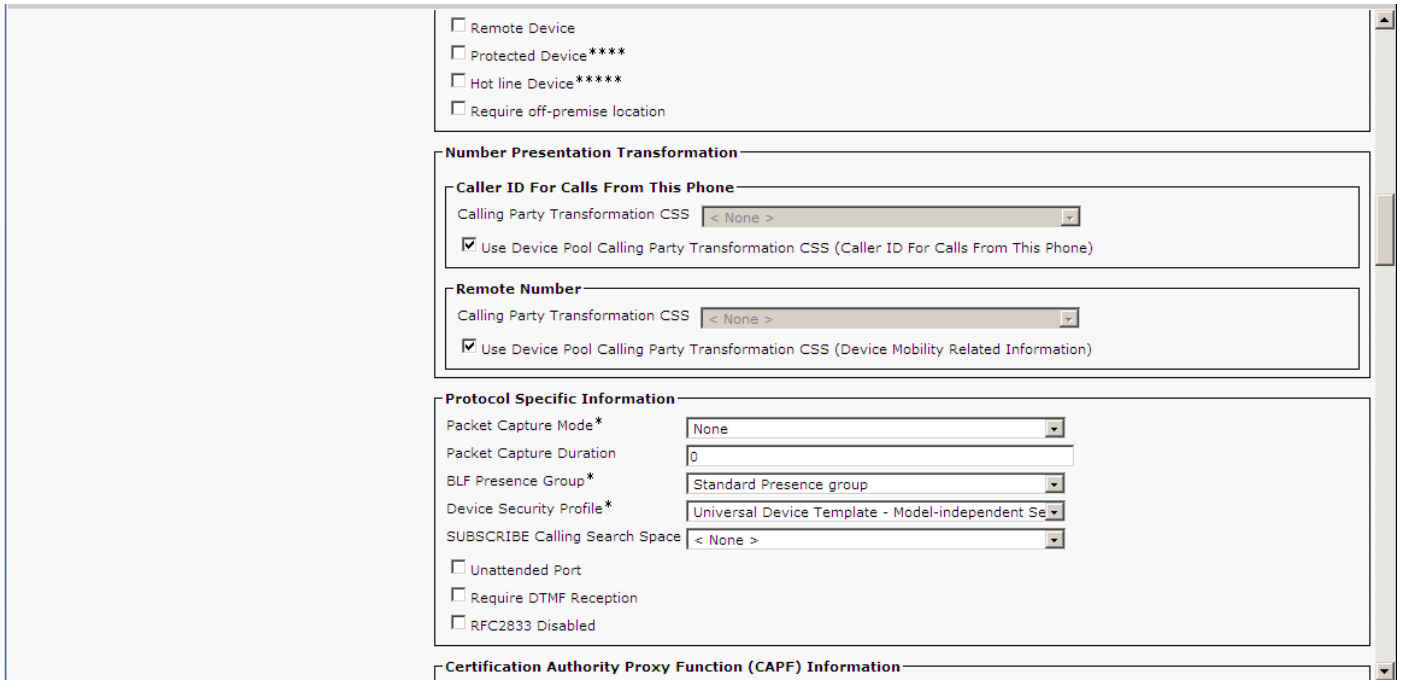
Media Resource Group List Configuration

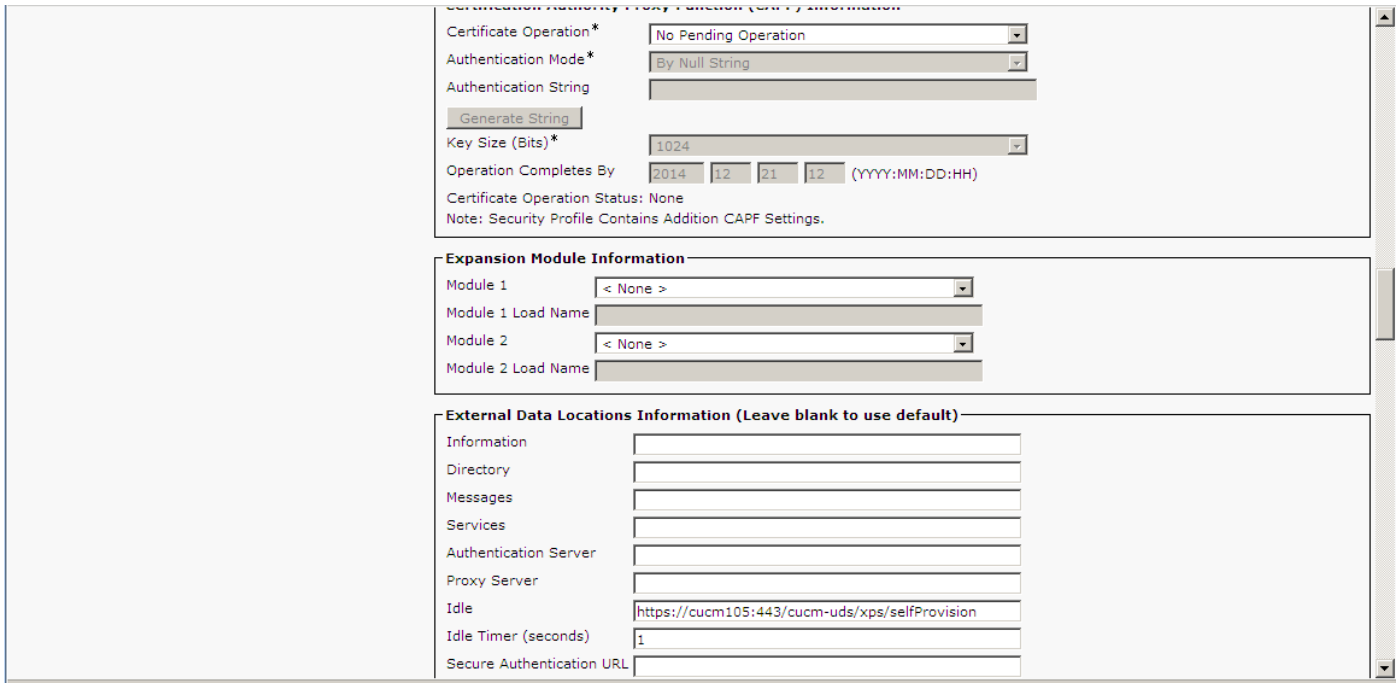


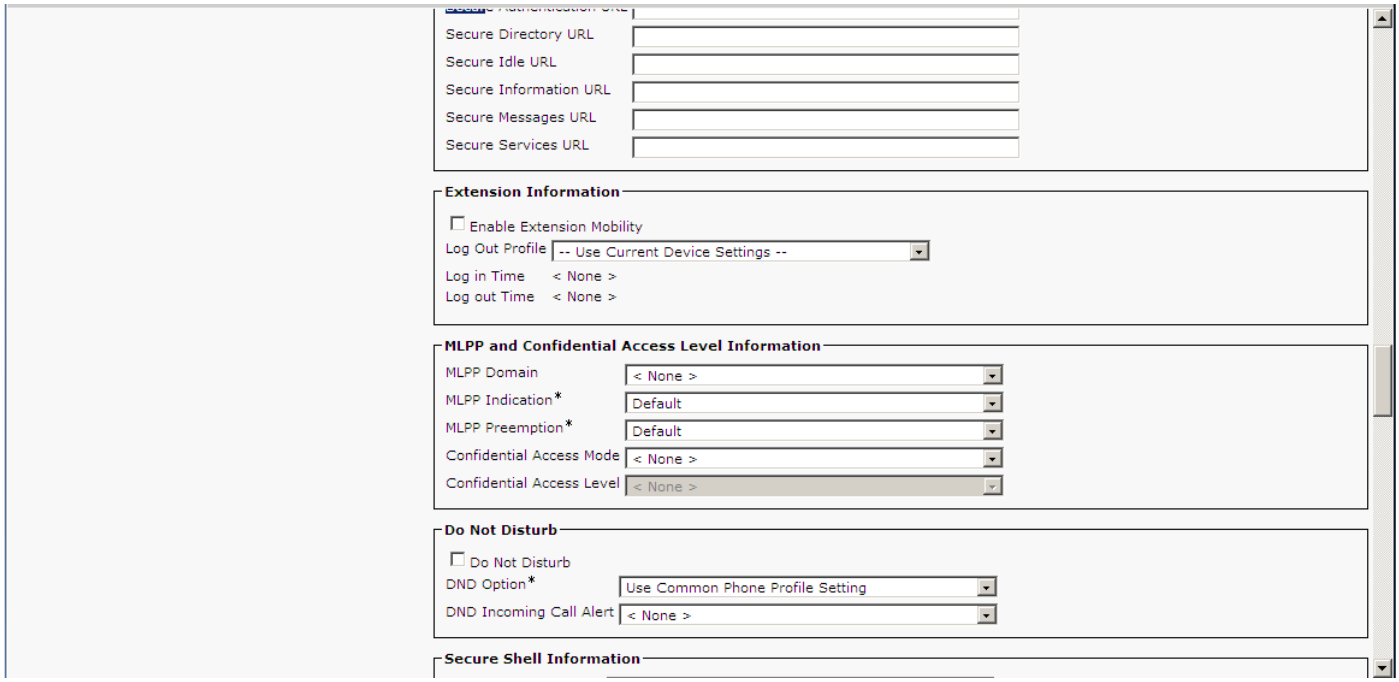
Note: The Media Resource Group (MRG) contains media resources, such as Conference Bridge, Transcoder, MoH server, and Annunciator. It will be assigned to a Media Resource Group List (MRGL), used to allocate media resources to groups of devices through Device Pools, or individually by configuring a valid MRGL at the device configuration page.

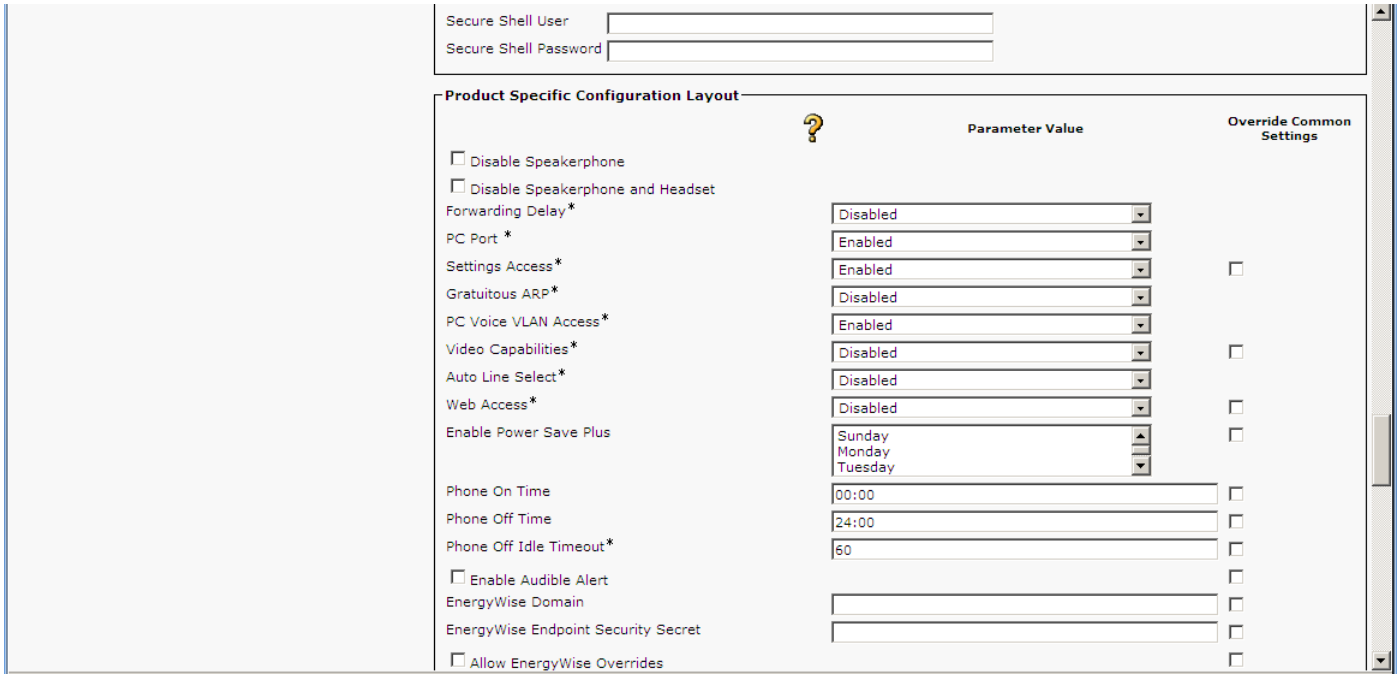
Phone Configuration

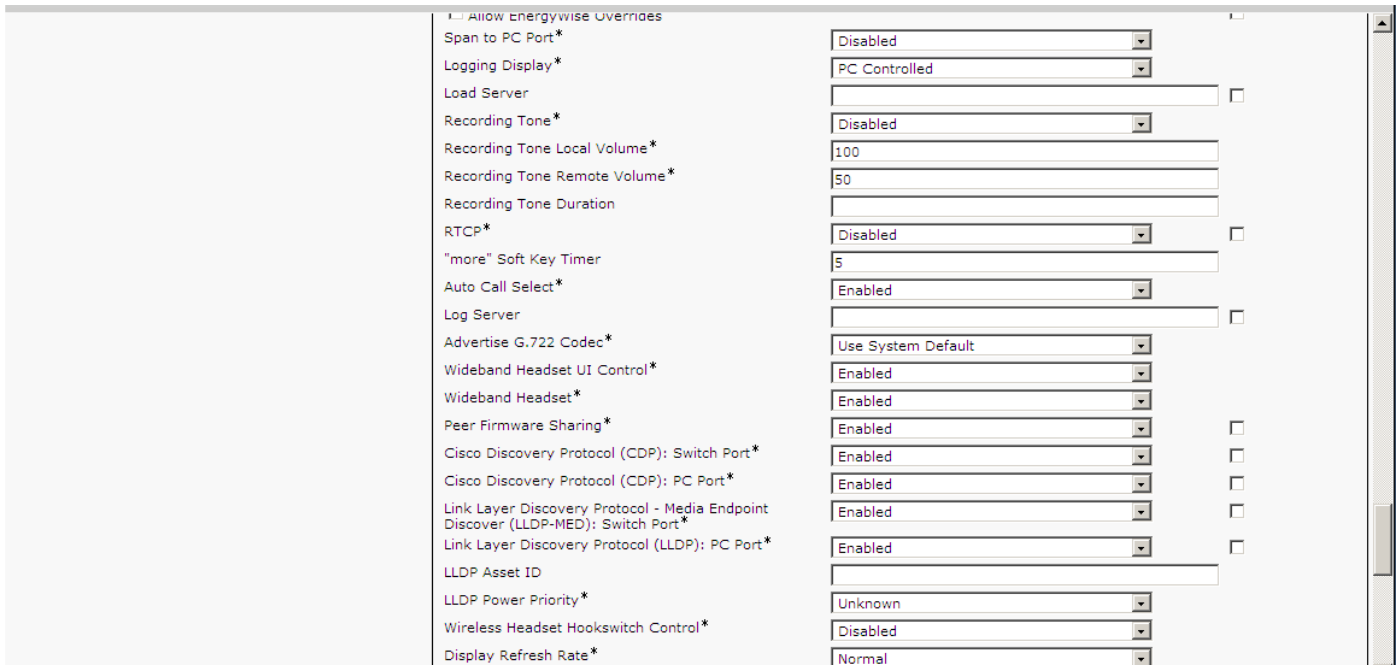




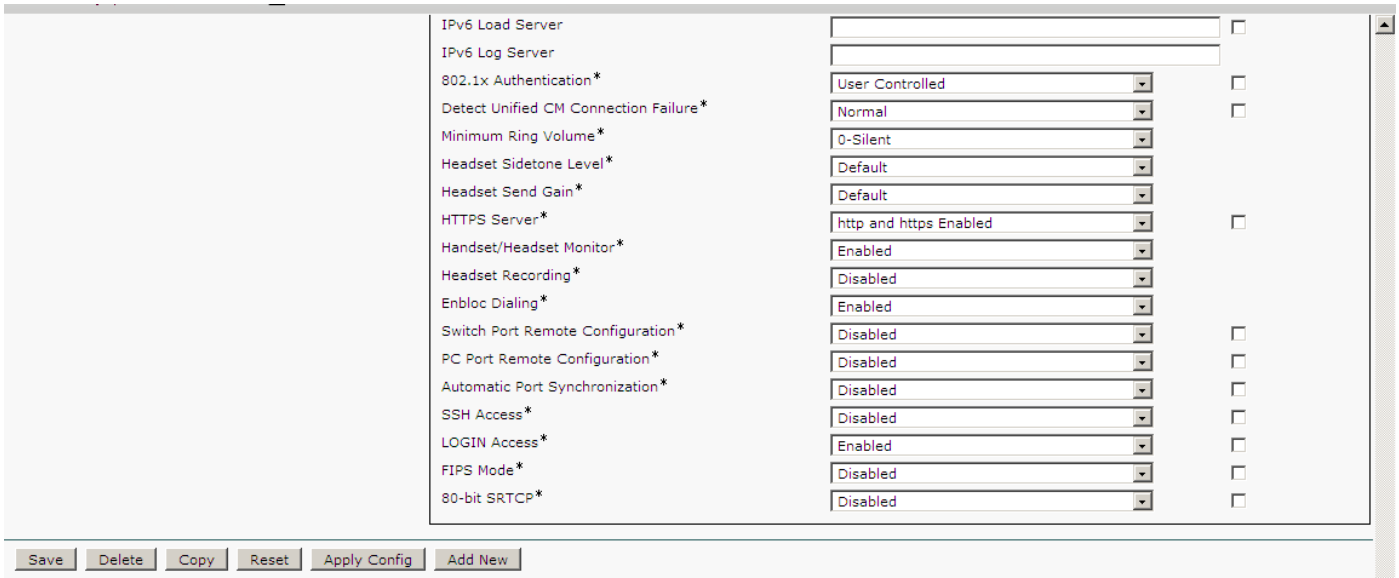






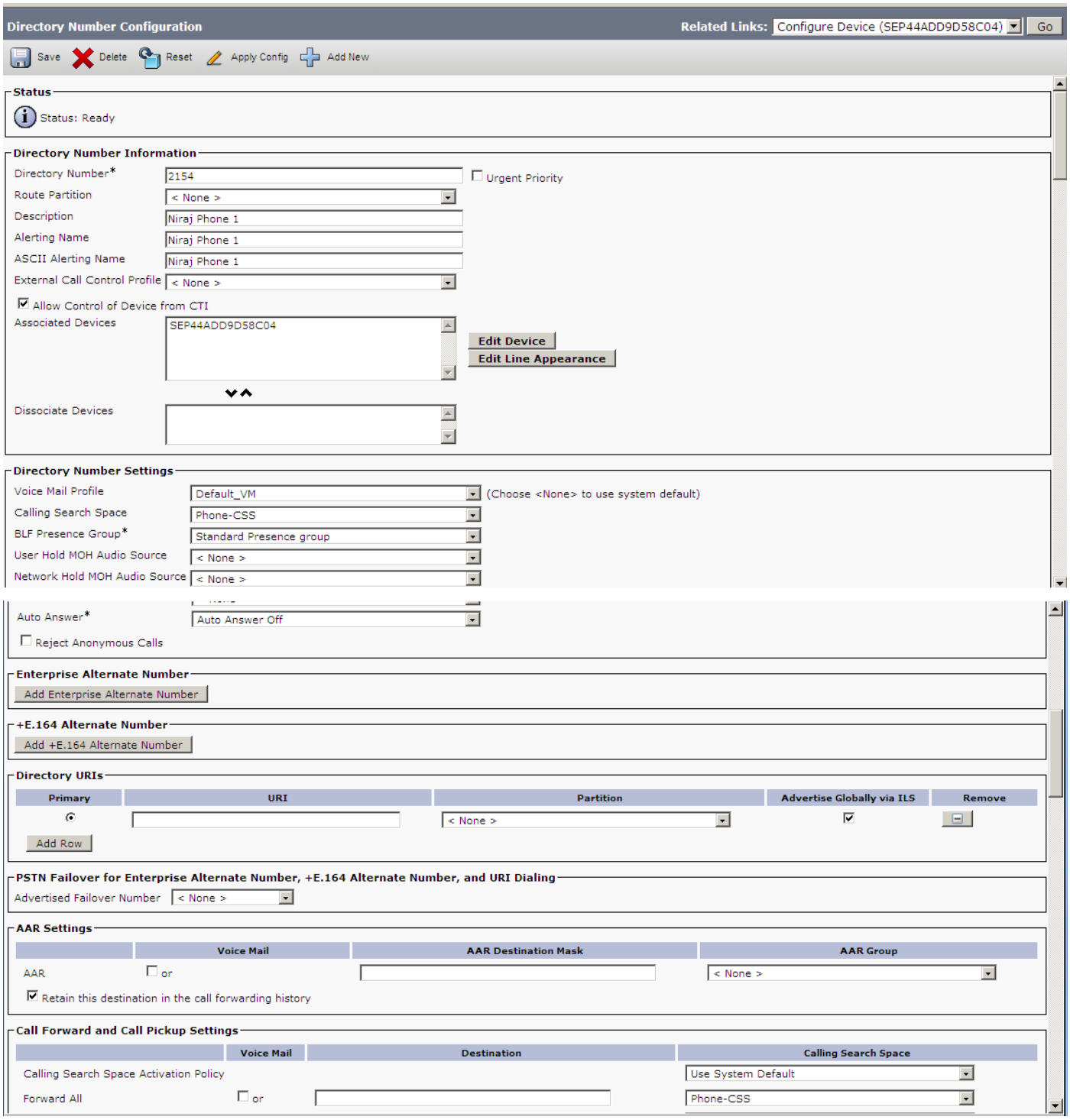


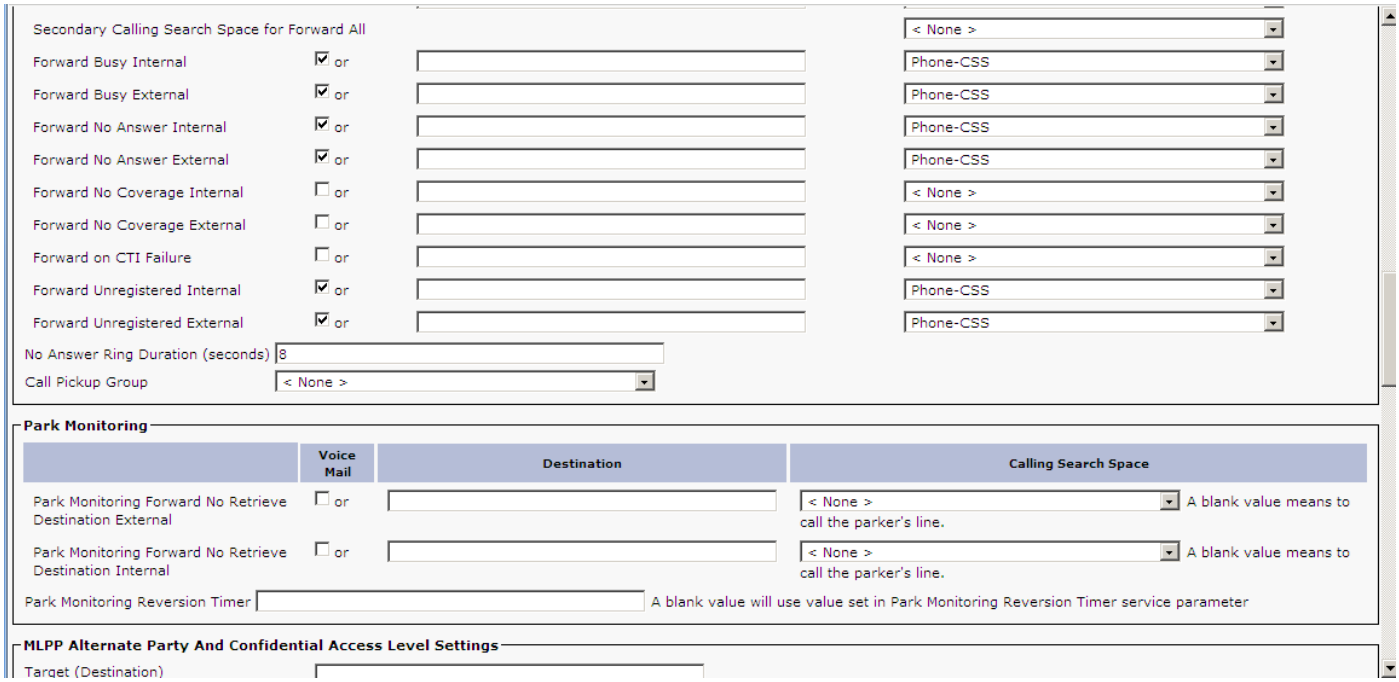
Allow Energywise Overrides	
Span to PC Port*	Disabled
Logging Display*	PC Controlled
Load Server	
Recording Tone*	Disabled
Recording Tone Local Volume*	100
Recording Tone Remote Volume*	50
Recording Tone Duration	
RTCP*	Disabled
"more" Soft Key Timer	5
Auto Call Select*	Enabled
Log Server	
Advertise 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

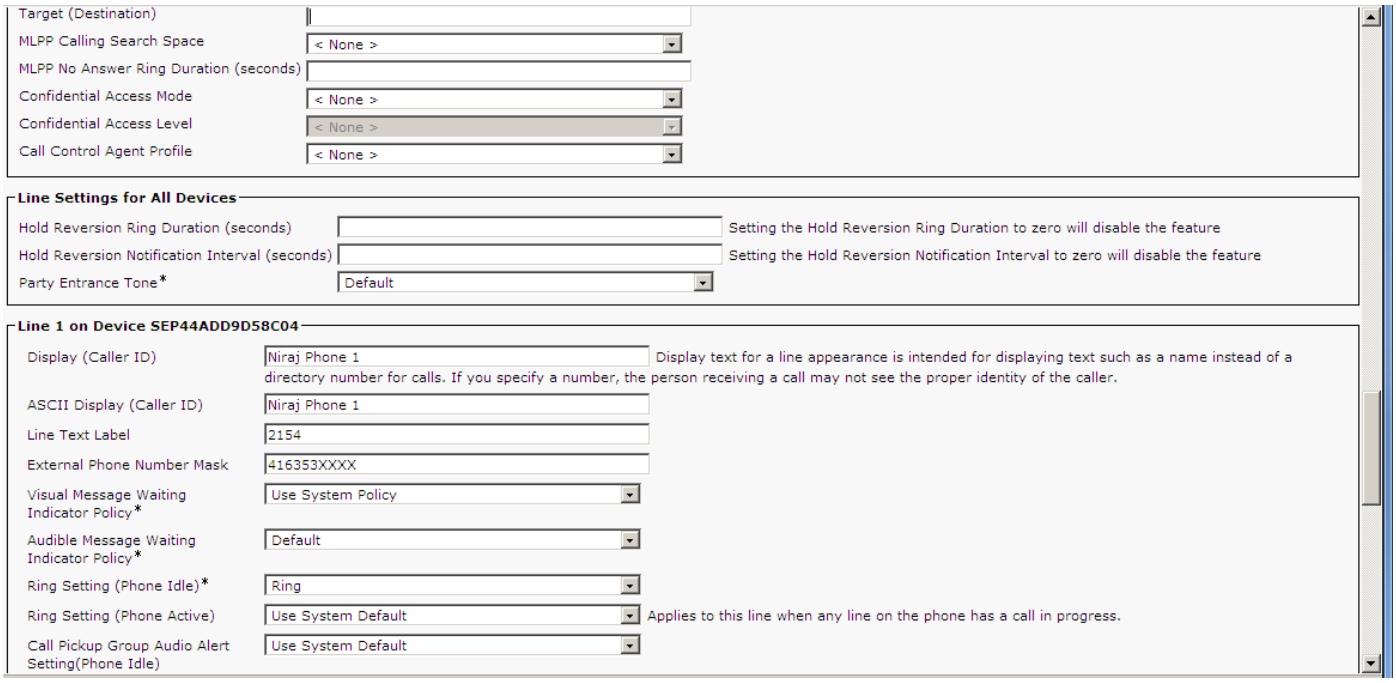


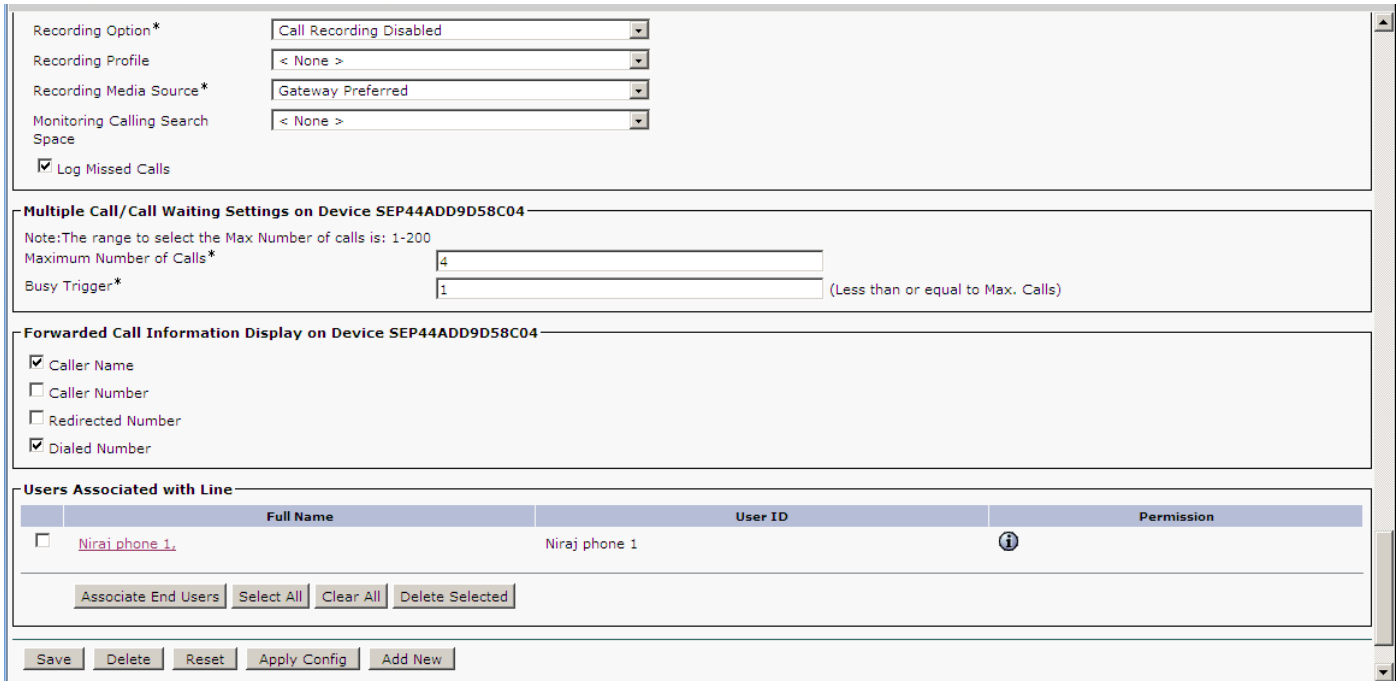
IPv6 Load Server	
IPv6 Log Server	
802.1x Authentication*	User Controlled
Detect Unified CM Connection Failure*	Normal
Minimum Ring Volume*	0-Silent
Headset Sidetone Level*	Default
Headset Send Gain*	Default
HTTPS Server*	http and https Enabled
Handset/Headset Monitor*	Enabled
Headset Recording*	Disabled
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

Save | Delete | Copy | Reset | Apply Config | Add New

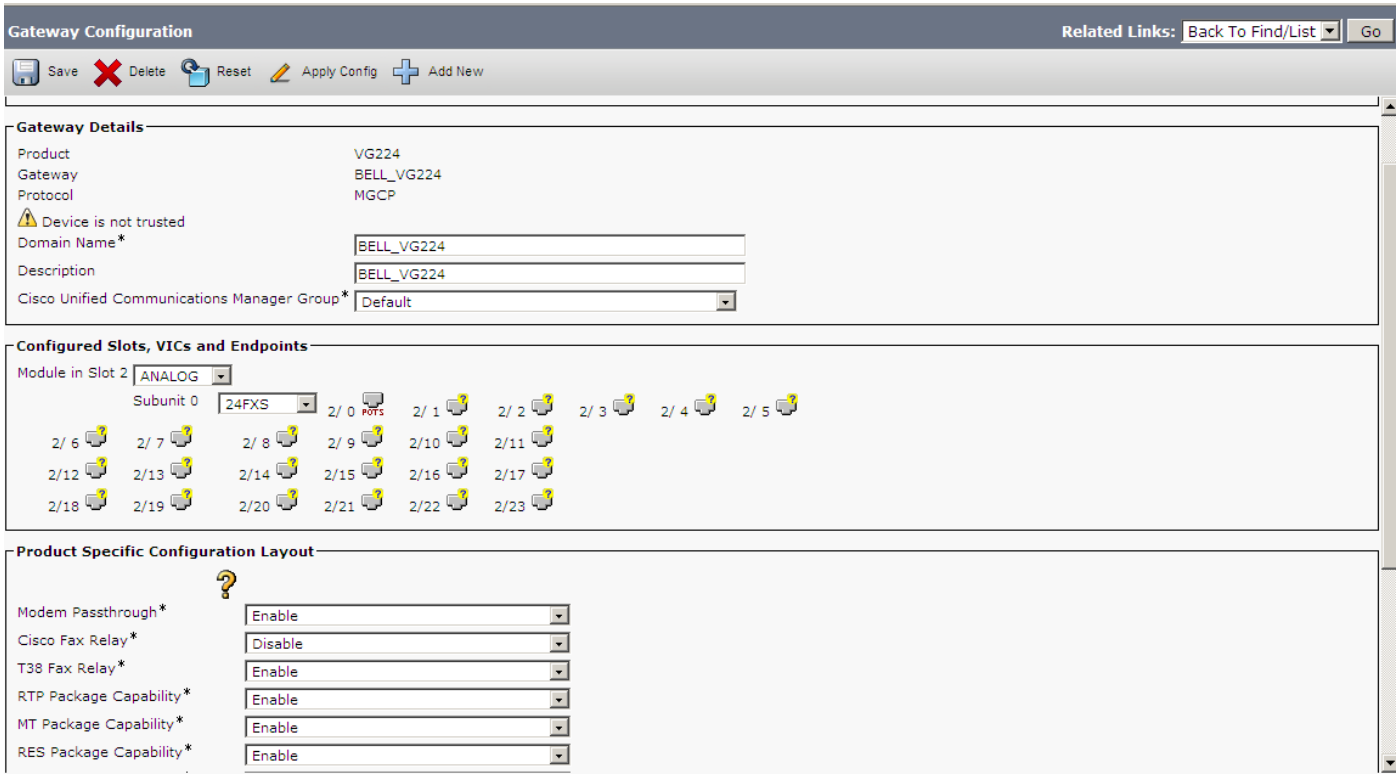


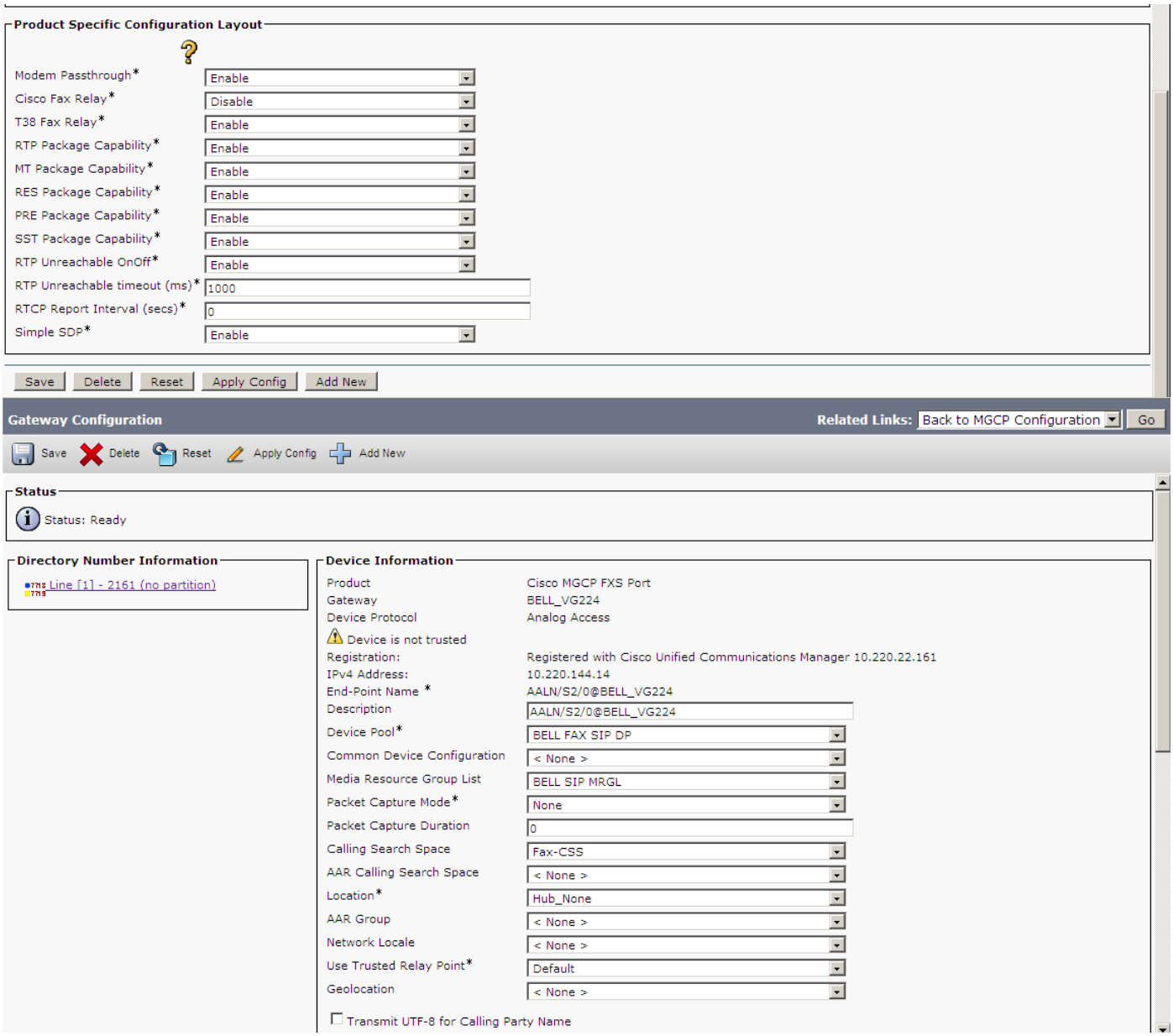


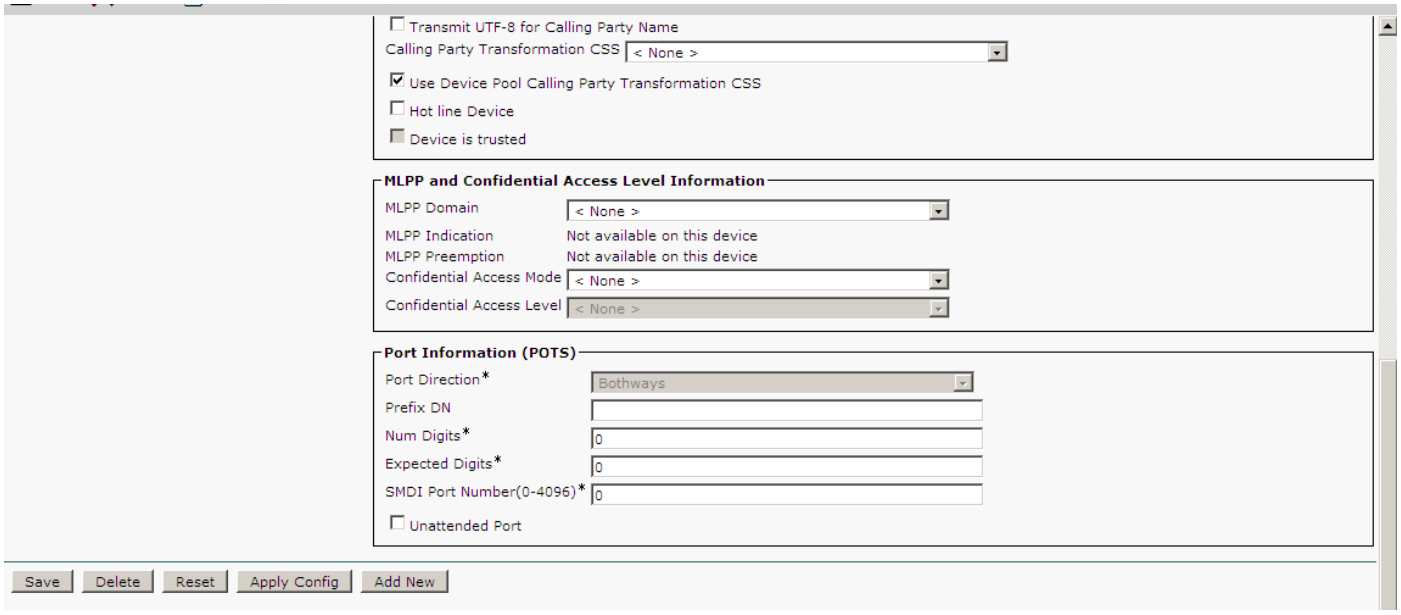




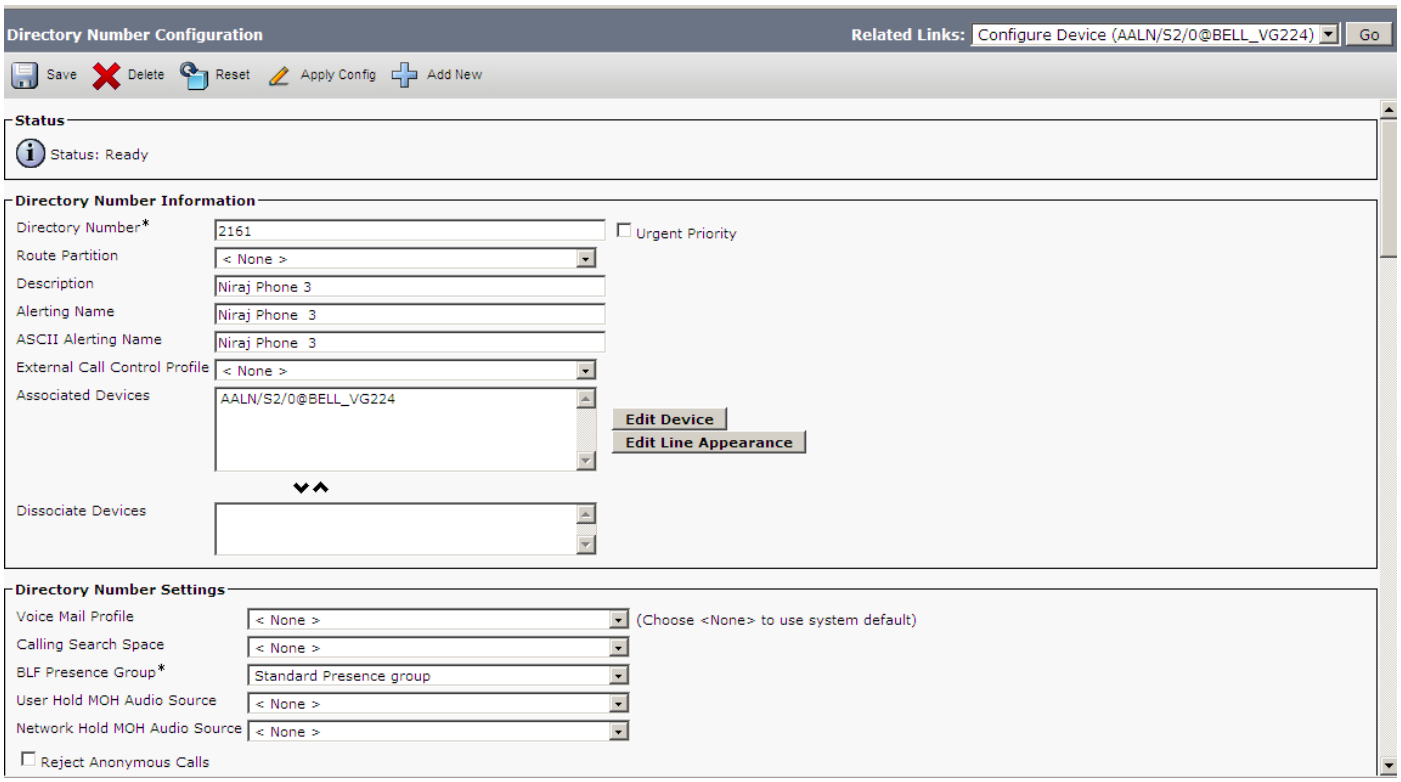
VG224 Analog Gateway (FAX application)







VG224 Analog Gateway Directory Number Configuration



Network Hold MOH Audio Source < None >

Reject Anonymous Calls

Enterprise Alternate Number

Add Enterprise Alternate Number

+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	☐ 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 >

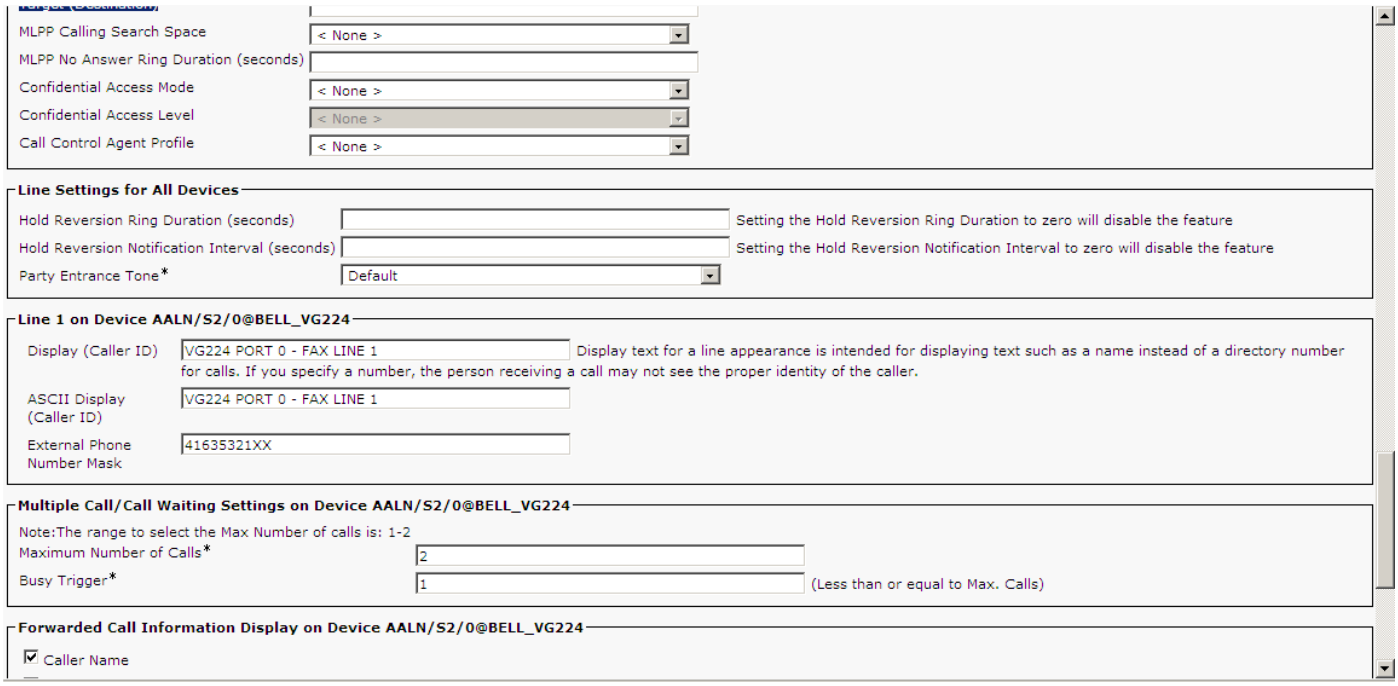
Park Monitoring

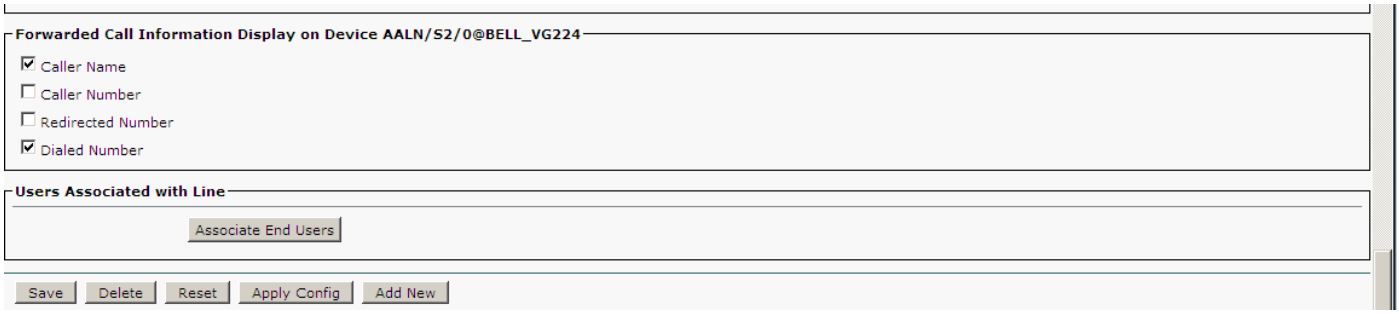
	Voice Mail	Destination	Calling Search Space	
Park Monitoring Forward No Retrieve Destination External	☐ or		< None >	A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	☐ or		< None >	A blank value means to call the parker's line.

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)





IOS Configuration of VG224 Analog Gateway

```
Bell_VG224#show run
Building configuration...

Current configuration : 3403 bytes
!
version 12.4
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Bell_VG224
!
boot-start-marker
boot-end-marker
!
no aaa new-model
!
resource policy
!
ip subnet-zero
no ip domain lookup
no ip dhcp use vrf connected
!
voice-card 0
!
voice service voip
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 supplementary-service h450.12
!
interface FastEthernet0/0
 ip address 10.220.144.14 255.255.255.240
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.220.144.1
!
ip http server
!
control-plane
!
 voice-port 2/0
 timeouts initial 60
 timeouts interdigit 60
 timeouts ringing infinity
!
voice-port 2/1
```

```
 timeouts initial 60
 timeouts interdigit 60
 timeouts ringing infinity
!
voice-port 2/2
 output attenuation -6
!
voice-port 2/3
!
voice-port 2/4
!
voice-port 2/5
!
voice-port 2/6
!
voice-port 2/7
!
voice-port 2/8
!
voice-port 2/9
!
voice-port 2/10
!
voice-port 2/11
!
voice-port 2/12
!
voice-port 2/13
!
voice-port 2/14
!
voice-port 2/15
!
voice-port 2/16
!
voice-port 2/17
!
voice-port 2/18
!
voice-port 2/19
!
voice-port 2/20
!
voice-port 2/21
!
voice-port 2/22
!
voice-port 2/23
!
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config
ccm-manager sccp local FastEthernet0/0
!
mgcp
mgcp call-agent 10.220.22.161 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
```

```
mgcp modem passthrough voip mode nse
mgcp ip qos dscp cs3 signaling
mgcp package-capability rtp-package
no mgcp package-capability res-package
mgcp package-capability sst-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
mgcp bind control source-interface FastEthernet0/0
mgcp bind media source-interface FastEthernet0/0
!
mgcp profile default
!
dial-peer voice 99920 pots
 service mgcpapp
 port 2/0
!
dial-peer voice 99921 pots
 service mgcpapp
 port 2/1
!
dial-peer voice 99922 pots
 service mgcpapp
 port 2/22
!
dial-peer voice 99923 pots
 service mgcpapp
 port 2/23
!
dial-peer voice 99924 pots
 service mgcpapp
 port 2/4
!
dial-peer voice 99925 pots
 service mgcpapp
 port 2/5
!
dial-peer voice 99926 pots
 service mgcpapp
 port 2/6
!
dial-peer voice 99927 pots
 service mgcpapp
 port 2/7
!
dial-peer voice 99928 pots
 service mgcpapp
 port 2/8
!
dial-peer voice 99929 pots
 service mgcpapp
 port 2/9
!
dial-peer voice 999210 pots
 service mgcpapp
 port 2/10
```

```
!
dial-peer voice 999211 pots
 service mgcpapp
 port 2/11
!
dial-peer voice 999212 pots
 service mgcpapp
 port 2/12
!
dial-peer voice 999213 pots
 service mgcpapp
 port 2/13
!
dial-peer voice 999214 pots
 service mgcpapp
 port 2/14
!
dial-peer voice 999215 pots
 service mgcpapp
 port 2/15
!
dial-peer voice 999216 pots
 service mgcpapp
 port 2/16
!
dial-peer voice 999217 pots
 service mgcpapp
 port 2/17
!
dial-peer voice 999218 pots
 service mgcpapp
 port 2/18
!
dial-peer voice 999219 pots
 service mgcpapp
 port 2/19
!
dial-peer voice 999220 pots
 service mgcpapp
 port 2/20
!
dial-peer voice 999221 pots
 service mgcpapp
 port 2/21
!
!
line con 0
line aux 0
line vty 0 4
 login
!
end
```

Acronyms

Acronym	Definitions
SIP	Session Initiation Protocol
MGCP	Media Gateway Control Protocol
SCCP	Skinny Client Control Protocol
CUCM	Cisco Unified Communications Manager
CUBE	Cisco Unified Border Element
DNS	Domain Name Resolution
G.711	Voice Codec (Uncompressed)
G729	Voice Codec (Compressed)
MGCP	Media Gateway Control Protocol
PAI	P-Asserted Identity
SCCP	Skinny Client Control Protocol
URI	Uniform Resource Indicator
RTP	Real Time Protocol

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Corporate Headquarters

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters

Cisco Systems International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

Asia Pacific Headquarters

Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

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Note: Testing was conducted in (*Bell Canada*) labs