Microsoft Lync 2010 using SIP via Cisco Unified Communications Manager- Session Manager Edition 8.5 to Cisco Unified Communications Manager 8.5 and Cisco Unified Border Element (Enterprise Edition) 1.4 on ASR to Service Provider

September 28, 2011 - Initial Version

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Introduction

- This application note describes the necessary steps and configurations for connectivity between Cisco Unified Communications Manager-Session Manager Edition (Cisco UCM-SME) 8.5(1) via Direct SIP to Microsoft Lync 2010 server using Microsoft Mediation Server, and via SIP connection to a Cisco Unified Communications Manager (Cisco UCM) 8.5(1), and SIP connection to simulated Service Provider (SP) using Cisco Unified Border Element (Cisco UBE) 1.4. The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability between the different leaf nodes (Microsoft Lync 2010, Cisco UCM and Cisco UBE/SP) via the Cisco UCM-SME.

- Features tested are basic call, 3-way (ad-hoc) conference, call transfer (attended and unattended), call forward (all, busy and no answer), hold/resume, DTMF inter-working and voice mail. This test setup also includes a connection to a simulated service provider using SIP trunks. Cisco UBE is used as a session border controller (SBC), providing demarcation, security, and inter-working services between the customer's private network and the service provider's SIP network. Voice mail was tested for Cisco UCM users with mailboxes registered to Unity Connection via a Direct SIP trunk from Cisco UCM-SME. Fax could not be tested (see limitations section).

- During testing, a Cisco ASR 1002 router was used as Cisco UBE, however other Cisco voice gateways, such as the 3900 series or AS5400 XM can be used and the decision to choose what Cisco gateway model to use is left to the customer. The customer should choose a Cisco IOS gateway model based on the capabilities and the capacity that will be required based on the planned network deployment. Here is a list of Cisco IOS products capable of running Cisco UBE.

  Cisco ASR 1000 Series Aggregation Services Router
  Cisco 3900 Series Integrated Services Routers
  Cisco 2900 Series Integrated Services Routers
  Cisco 3800 Series Integrated Services Routers
  Cisco 2800 Series Integrated Services Routers
  Cisco AS5400XM Universal Gateway
  Cisco AS5350XM Universal Gateway

- If additional guidance on the Cisco UBE is needed, please refer to the Cisco UBE section on the Cisco Interoperability Portal (www.cisco.com/go/interoperability).

- This configuration was tested to simulated service provider. Results may vary based on service provider being used.

Network Topology

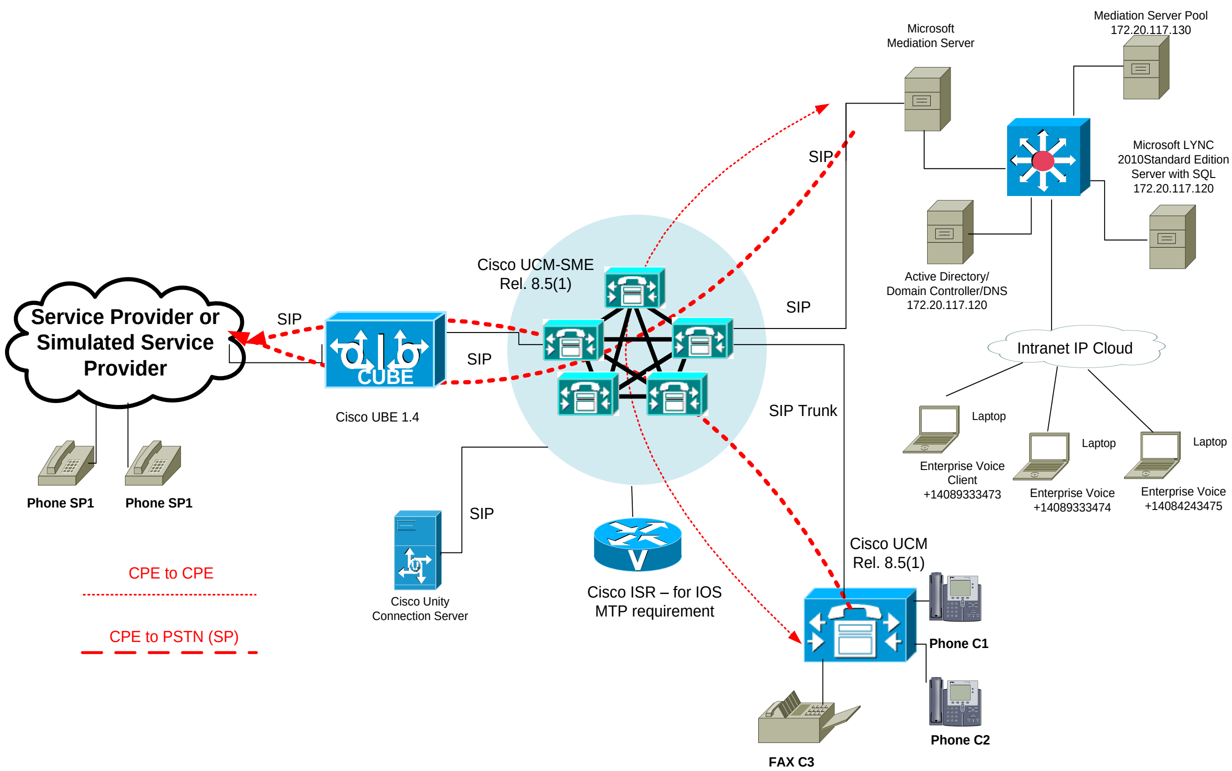


Figure 1. Basic Call Setup

Note: As of Cisco UCM-SME Rel. 8.5, for SIP trunk connection requiring Early Offer function, configuration is made under the SIP Profile configuration associated with the SIP trunk. Prior to Release 8.5, network administrators were forced to configure SIP trunks on Cisco UCM-SME to send early offer by using/allocating Media Termination Point. In Cisco UCM-SME Rel. 8.5, the Media Termination Point (MTP) checkbox under the SIP Trunk configuration page no longer needs to be checked to implement early offer function.

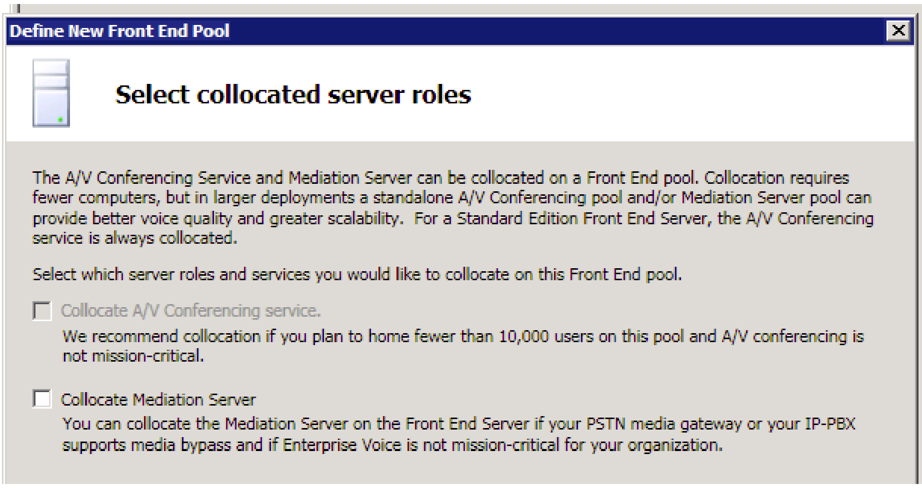
Capabilities

- Voice calls including supplementary services can be successfully established between endpoints controlled by the Microsoft Lync 2010 and endpoints controlled by the Cisco UCM.
- Voice calls including supplementary services can be successfully established between endpoints controlled by the Microsoft Lync 2010 and the PSTN, using Cisco UBE as a session border controller.
- IOS MTPs and transcoders registered to the Cisco UCM-SME can be employed to offer G729 as originating codec option.

Limitations

These are the known limitations, caveats, or integration issues.

- Microsoft Lync does not support sending or receiving calling and connected name in certain scenarios (like call forward and call-transfer). Cisco UCM sends its calling name and number to Microsoft Lync Mediation Server, but the Lync only displays the calling number and no name is displayed.
- Microsoft Lync 2010 does not provide music on hold (MoH) capability by default except on Microsoft Attendant Console. MoH needs to be enabled using CLI through the Lync Management Shell.
- Microsoft Lync 2010 displays presence information in the Enterprise Voice client only for other Enterprise voice users configured in the Active Directory and Enterprise voice enabled.
- For Microsoft Lync 2010, the Enterprise Voice client cannot be configured for Call Forward Busy. As an option the Enterprise Voice client allows the user to receive a notification of an incoming call during an active call and "redirect" the incoming call to a destination of the user's choice.
- Calling and connected number updates during call-forward and call-transfer scenarios are not fully supported due to SIP UPDATE messages not being interoperable between the systems. For e.g. when a Lync user transfers a PBX call to another PBX user, the local PBX phone will still show connected to the first Lync user.
- Cisco SIP Phones do not send out Comfort Noise payload 13 in the m line of the SDP to indicate support for Comfort Noise as per RFC 3389 and as a result Microsoft doesn't send out SID frames when call is muted on Microsoft side.
- Microsoft Lync 2010 sends its SIP URI @domain in the INVITE, while setting up a conference call, as a result the caller ID will not be displayed properly on the far end.
- When the Enterprise Voice client, initiates a conference call, the conference call does not end when the initiator hangs up, but only after all parties hang up. Additionally for the Enterprise Voice users on the conference call, the session is not dropped and they can rejoin the conference, unless they close the window pop-up for the call, referred to as the toast.
- The Media Bypass capability introduced in Microsoft Lync 2010 requires all media through a SIP trunk integration to originate on a single IP address. For this Microsoft requirement in a Lync to Cisco UCM inter-working, media termination point is required on the SIP trunk for Media Bypass to work.
- Media Bypass is a mandatory requirement if customers want to collocate the Mediation Server role with the Front-End server role. **Important:** According to the Microsoft Lync Planning Tool collocation should only be used if the IP-PBX and/or the voice gateways support Media Bypass and Enterprise voice is not considered mission critical to the organization.



- Media Bypass works as long as an MTP is required on the SIP trunk. The need for an MTP is due to the following:

    1. Lync supports Media Bypass with early offer, but does not support re-invite without SDP
    2. Lync doesn't support Media Bypass with delayed offer
    3. Because of #1, hold/resume, transfer, forward, etc won't work without an MTP.

    RFC 3261 sections 13.2.1 and 14.1 make a strong case that Lync's behavior (#1 above) isn't compliant with MUST aspects of the SIP standard. Including the MTP enables standards-compliant interoperability in light of the Lync implementation.

- Microsoft Lync 2010 only supports TCP transport to carry SIP messages. If UDP is required CUBE can be used to perform the conversion.

- Microsoft Lync 2010 only supports G711 ulaw or alaw on the outside interface. If G729 is required CUBE or transcoder registered to SME can perform the transcoding function.

- Microsoft Lync 2010 does not support G711 Fax Passthrough.

- Fax T.38 to Service provider is not tested. Service Provider used for this test does not support Fax T.38 switchover.

- On Lync Mediation Server, RTCP is set to disabled, as RTCP is an end point specific configuration and not all Cisco phones support RTCP. Phones like 7970, 7962, and 9971 have this support and this can be enabled under Phone Configuration – Product Specific Configuration Layout.

Features Supported

- Basic calls
- Digits Translation
- Intra-site call conferencing
- Call Transfer-Attended and Unattended (See Limitations section for details)
- Hold and resume
- Call forward – Unconditional, Busy and No reply (See Limitations section for details.)
- Simultaneous calls
- Inbound/Outbound Basic calls (G711 u-law and a-law) (RFC 3261)
- Call Forward (RFC 3261)
- Conference calls
- DTMF (RFC 2833)
- Options Testing
- Reliable 1XX Provisional Response or PRACK (RFC 3262)
- Early Media (RFC 3261)
- Call hold and resume using Offer/Answer Model (RFC 3264)
- ISDN Mapping

Features Not Supported and Not Tested

- T.38 FAX Relay was not tested. - The Service provider used for this test does not support T.38 FAX.
- Blind call transfers
- Comfort noise is not supported
- Decline a call (CUCM phones divert calls to voicemail, instead of declining the call)

System Components

Hardware Requirements

- Cisco MCS 7835 Unified Communications Manager
- Cisco MCS 7835 Session Manager Edition
- Cisco 3825 voice gateway
- Cisco ASR1K router
- Two Cisco Unified IP phone 7961 configured as SCCP phones
- One Cisco Unified IP phone 7960 configured as SIP phones
- Three DELL notebooks running LYNC clients

Software Requirements

The following software is required:

- Cisco Unified Communications Manager Release 8.5
- Cisco UCM-Session Manager Edition Release 8.5
- Microsoft Lync Server 2010 Standard Edition, Windows Server 2008 R2 x64 Enterprise Edition OS
- Windows Active Directory/DNS/Cert Server, Windows Server 2008 R2 x64 Enterprise Edition OS
- Backend Database: Windows SQL Server 2008 Enterprise Edition, Windows Server 2008 x64 Enterprise Edition OS
- Microsoft Lync 2010 (build 4.0.7577.4)
- Cisco ASR1K router Release 3.3.1.S

IMPORTANT:

Microsoft Lync Server, Mediation Server and Lync Communicator all need to run through the Microsoft Update website to have the latest updates installed for this integration to work correctly.

System Configuration

This section contains configuration menus and commands and describes configuration sequences and tasks.

Configuring the Microsoft LYNC 2010 Server Standard Edition

The look and feel between Lync and the previous versions are different (refer to the captured screenshots) and tools like Lync planning tool, Topology builder, Lync Server Control Panel, Lync Management Shell are added. The main differences were seen during the deployment of the Lync environment itself and how its own components connected to each other. Those differences include, the use of 64-bit servers for each component (refer to Software Requirements section), the need for an interface module (automatically downloaded during installation) on the Front End Servers to enable communication with SQL 2008 (refer to the Microsoft deployment and installation guides for more information, links included below), and finally the configuration of the certificate authority server (refer to the Microsoft deployment and installation guides for more information, links included below).

For Cisco documentation guides including release notes, compatibility matrix, deployment, installation guides etc, go to:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/8x/uc8x.html

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/docguide/8_5_1/dg851.html

For Microsoft deployment and installation guides, go to:
http://technet.microsoft.com/en-us/library/gg293124.aspx
http://technet.microsoft.com/en-us/library/gg398616.aspx

Configuration Steps:

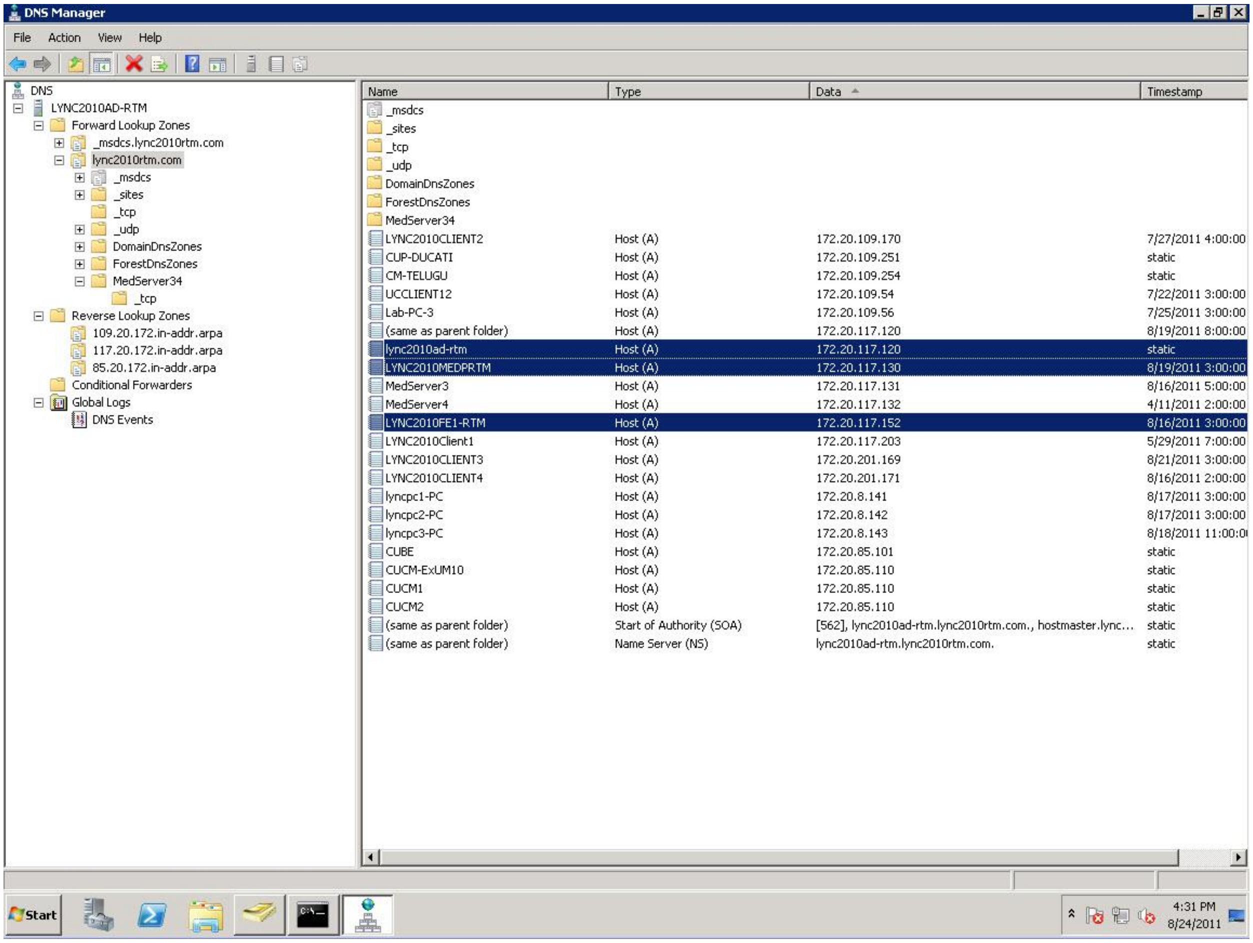
1. Domain Name System Configuration
2. Front End Server/Pool Configuration
3. Mediation Server Configuration
4. User Configuration
5. Microsoft Lync 2010 Configuration

Please refer to the Microsoft Lync Standard Edition Server deployment guide for setup details. Only interoperability related configurations are included in this document.

Domain Name System Configuration

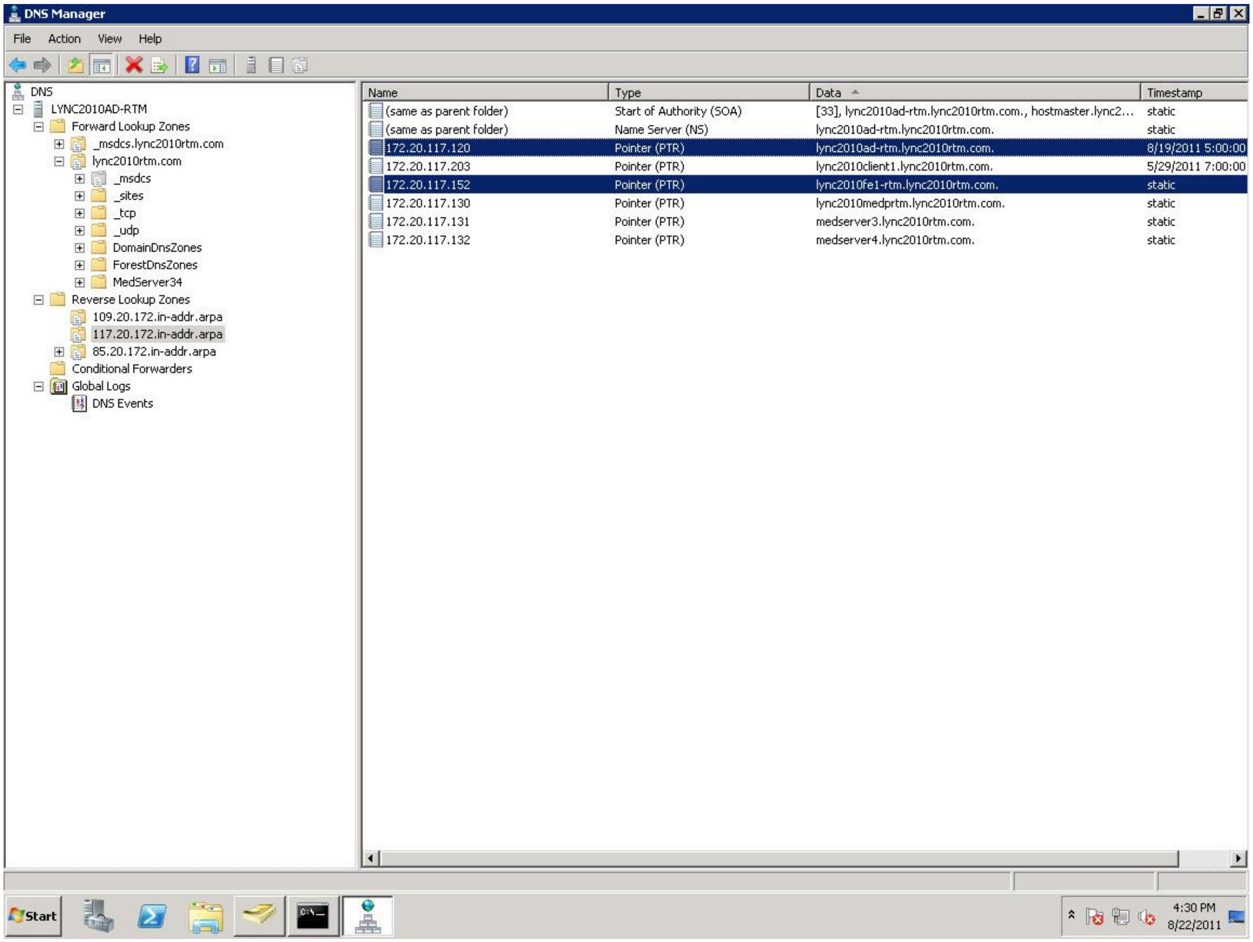
Start > Administrative Tools > DNS

Forw ard Lookup Z one



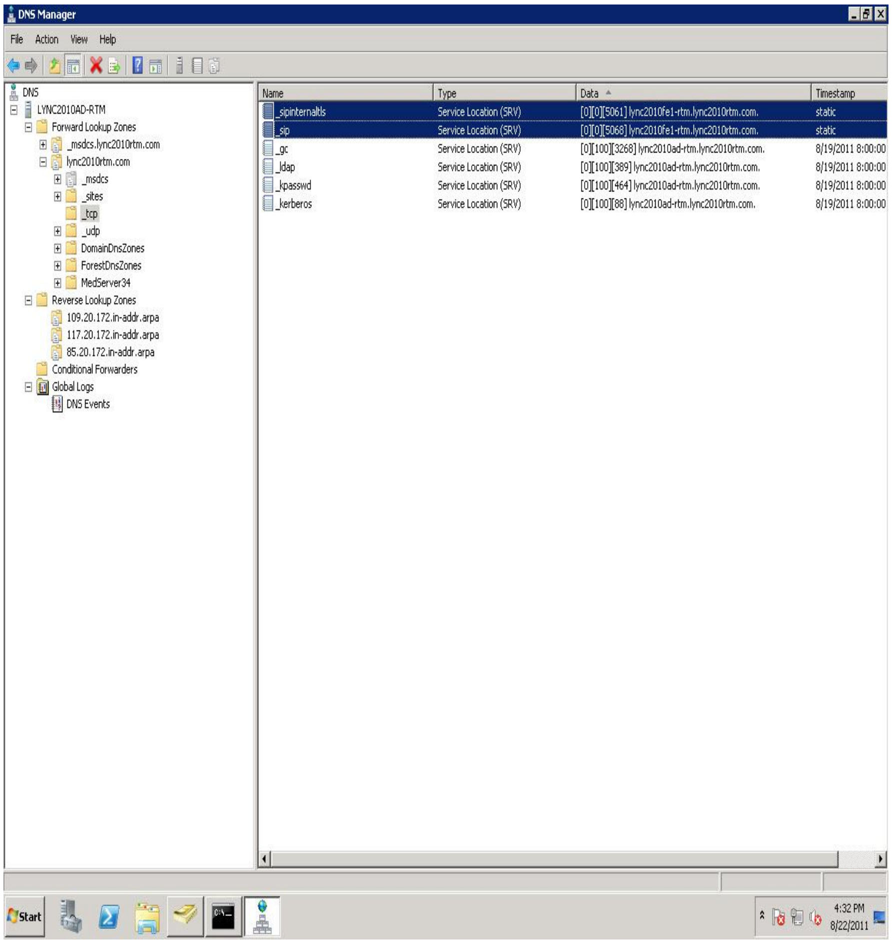
Host A records adde d for Lync Front End server, Active Directory/DNS and Mediation Server.

Reverse Lookup Zones Configuration

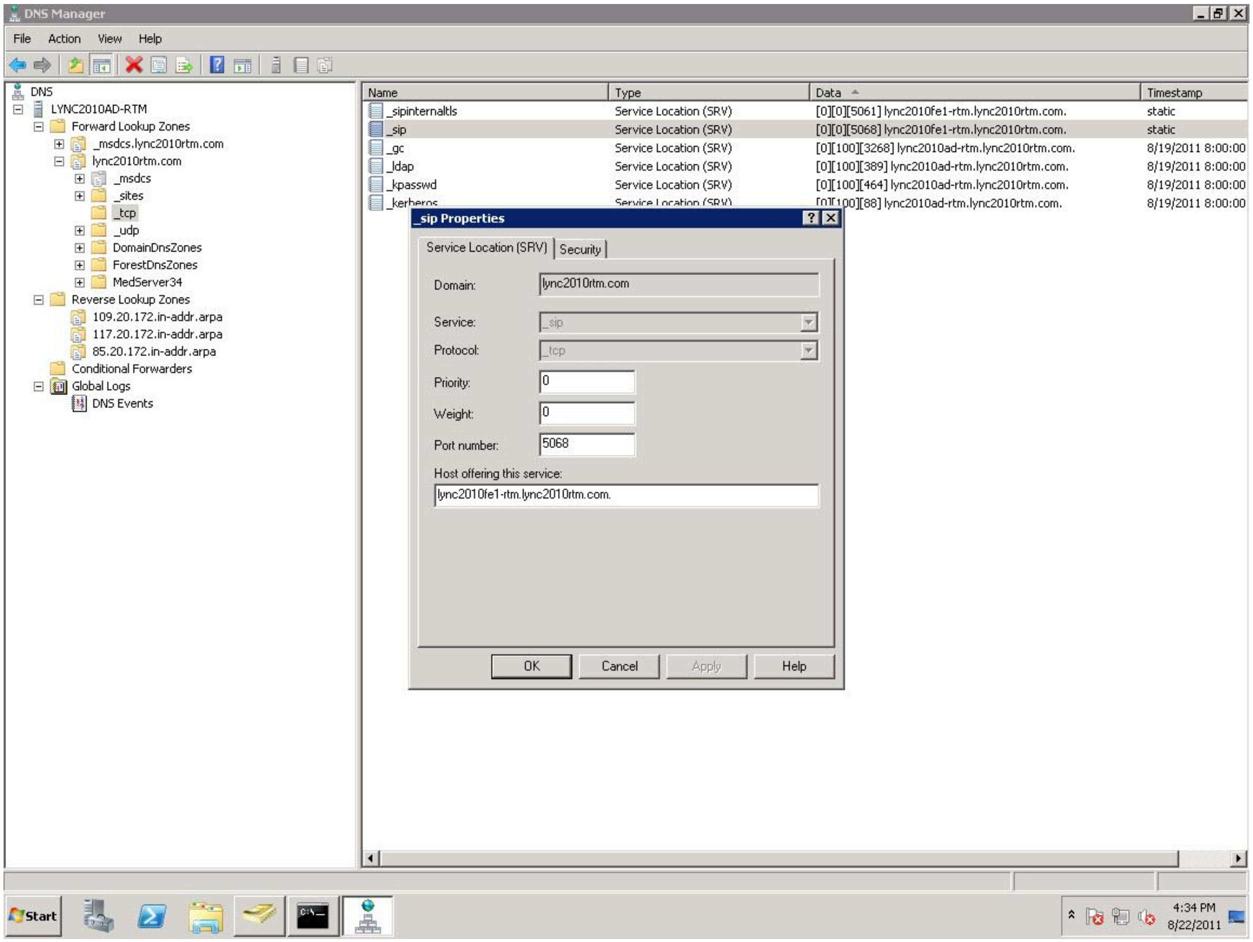


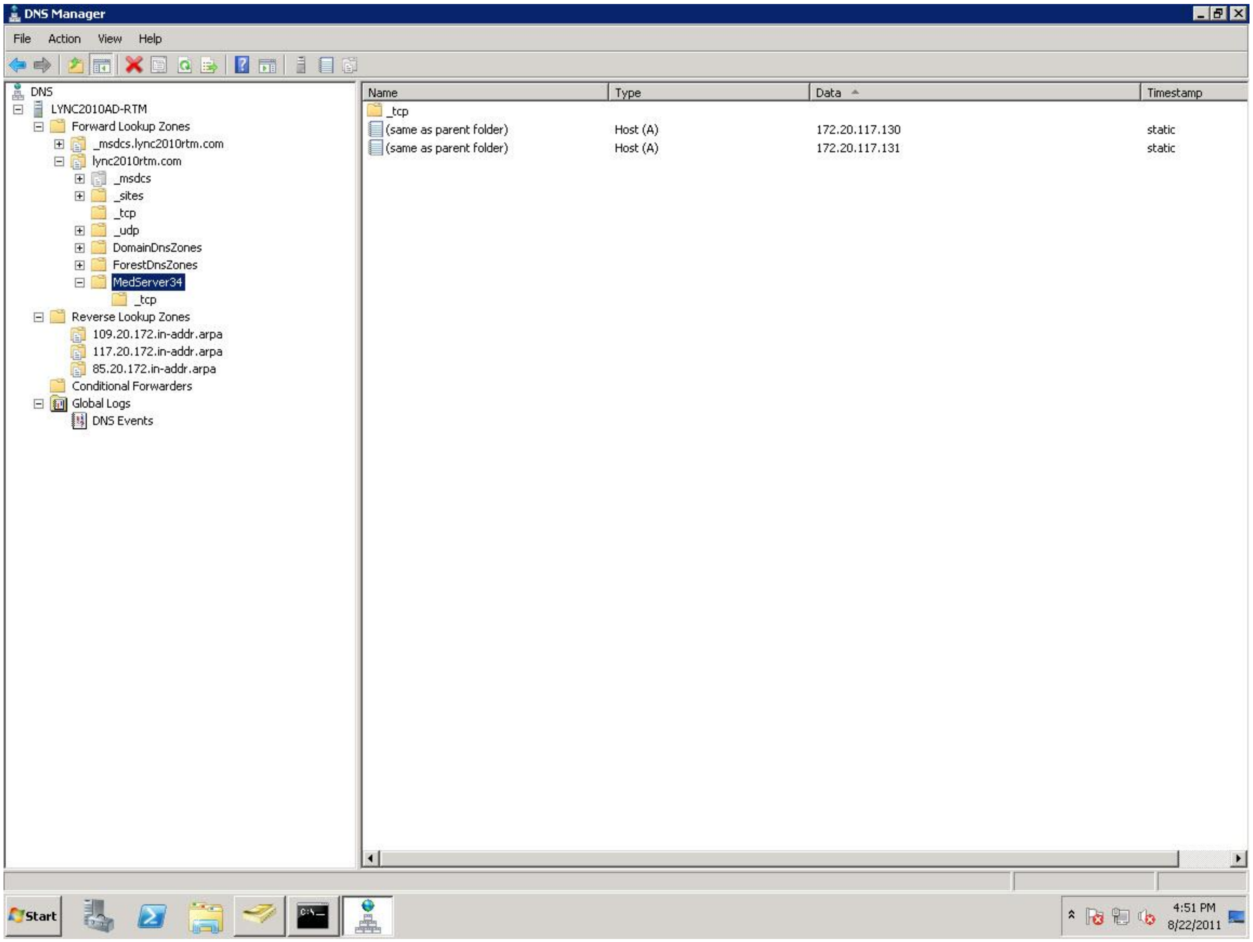
PTR Records added for the Lync server and Active Directory.

SRV Records Configuration

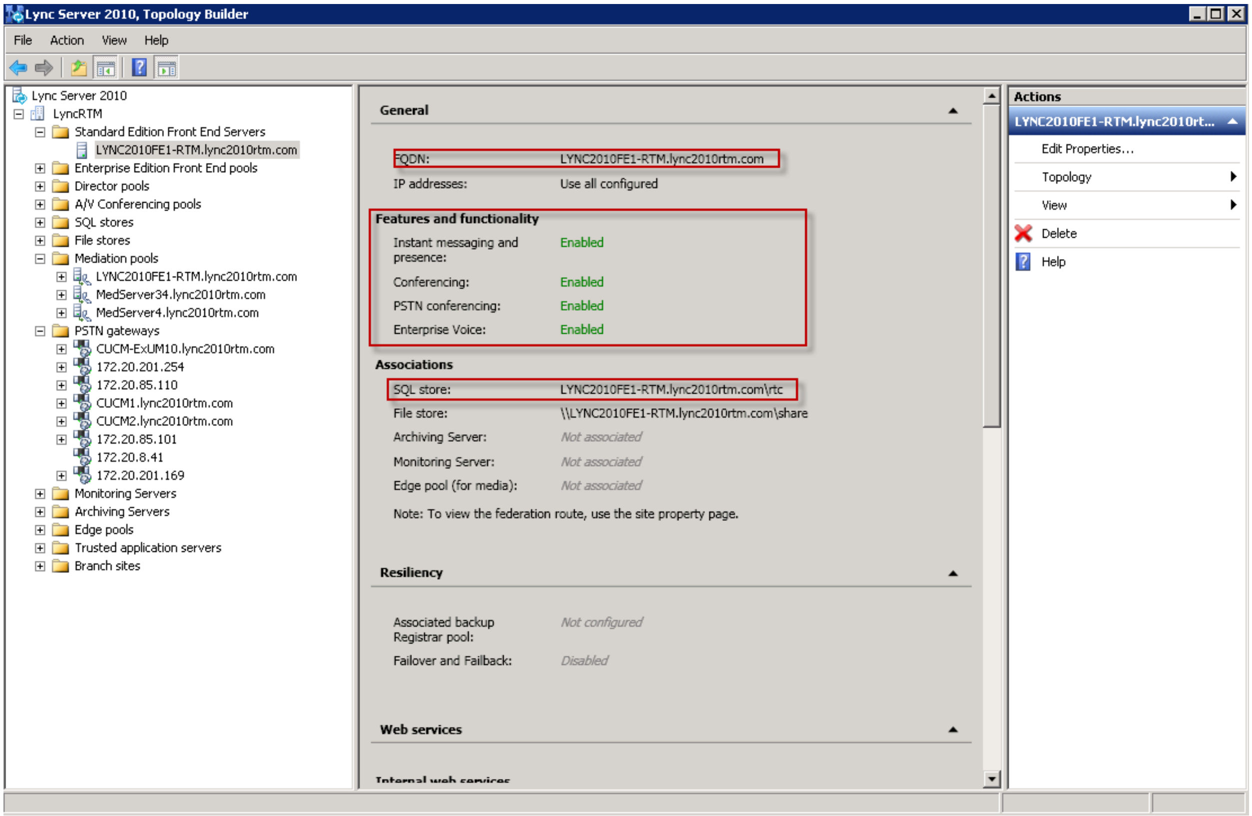


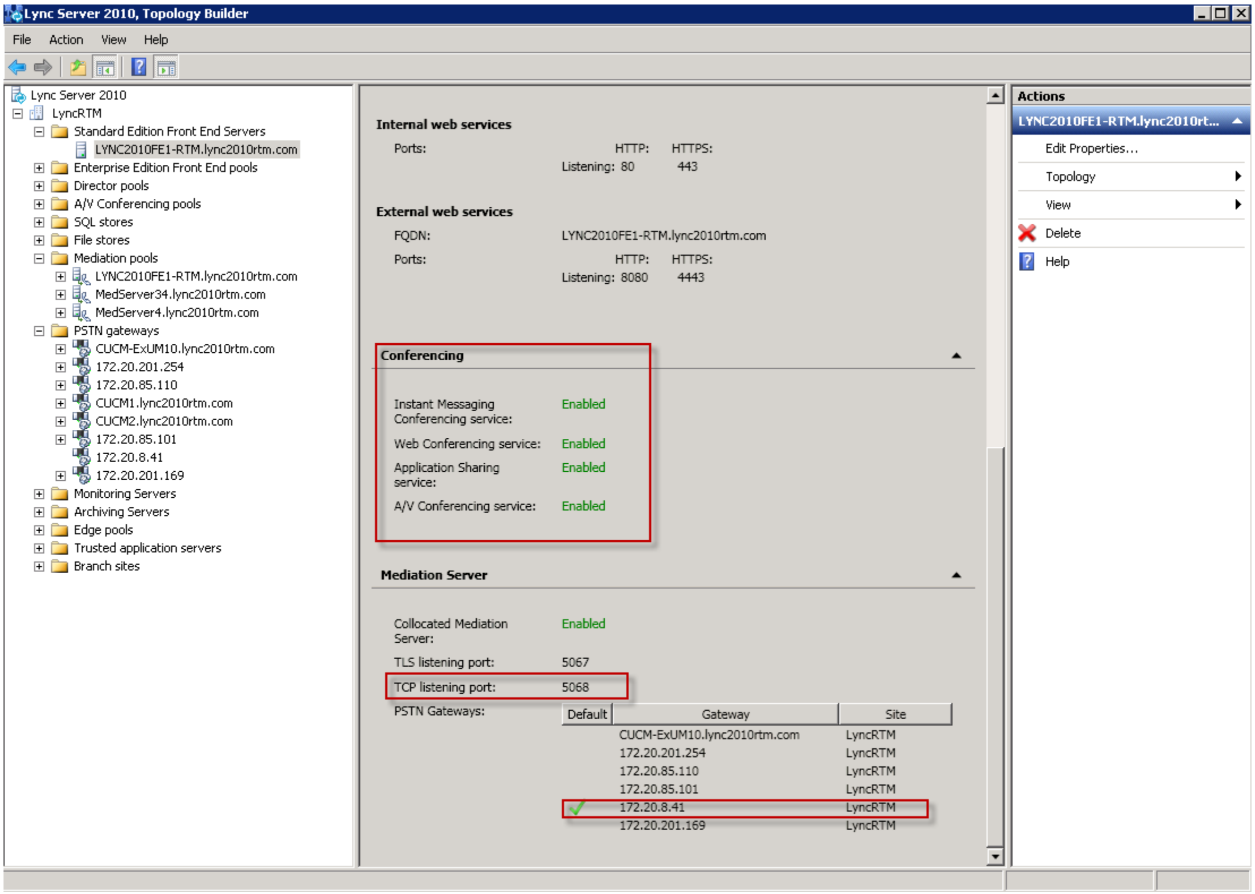
SRV Record added for SIP domain service offered by the Lync pool (for Automatic Client Sign-in).


DNS Manager
File Action View Help
DNS
LYNC2010AD-RTM
Forward Lookup Zones
_msdcs.lync2010rtm.com
lync2010rtm.com
_msdcs
_sites
_tcp
_udp
DomainDnsZones
ForestDnsZones
MedServer34
Reverse Lookup Zones
109.20.172.in-addr.arpa
117.20.172.in-addr.arpa
85.20.172.in-addr.arpa
Conditional Forwarders
Global Logs
DNS Events
Name
Type
Data
Timestamp
_sipinternaltls
Service Location (SRV)
[0][0][5061] lync2010fe1-rtm.lync2010rtm.com.
static
_sip
Service Location (SRV)
[0][0][5068] lync2010fe1-rtm.lync2010rtm.com.
static
_gc
Service Location (SRV)
[0][100][3268] lync2010ad-rtm.lync2010rtm.com.
8/19/2011 8:00:00
_ldap
Service Location (SRV)
[0][100][389] lync2010ad-rtm.lync2010rtm.com.
8/19/2011 8:00:00
_kpasswd
Service Location (SRV)
[0][100][464] lync2010ad-rtm.lync2010rtm.com.
8/19/2011 8:00:00
[0][100][88] lync2010ad-rtm.lync2010rtm.com.
8/19/2011 8:00:00
_sip Properties
Service Location (SRV)
Security
Domain:
lync2010rtm.com
Service:
_sip
Protocol:
_tcp
Priority:
0
Weight:
0
Port number:
5068
Host offering this service:
lync2010fe1-rtm.lync2010rtm.com.
OK
Cancel
Apply
Help
Start
4:34 PM
8/22/2011

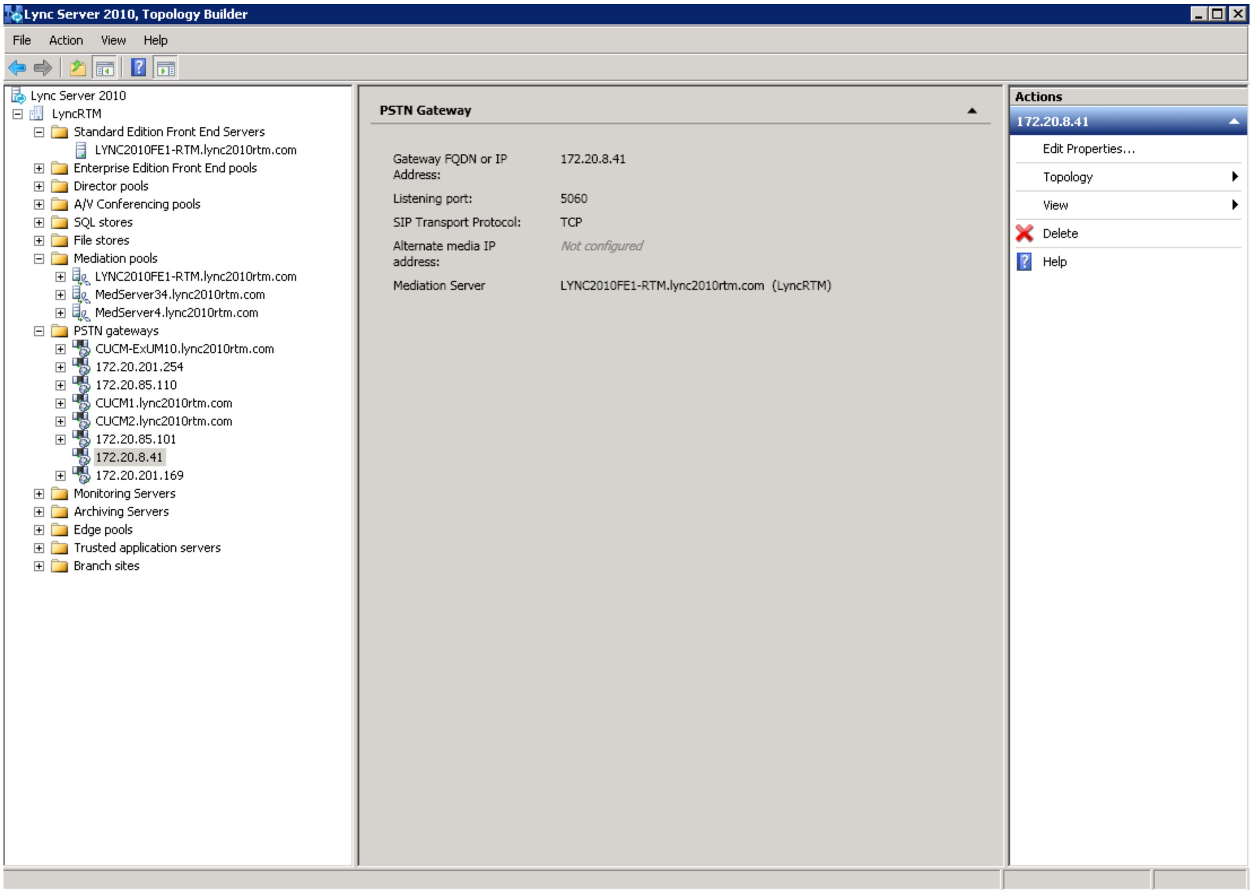

DNS Manager
File Action View Help
DNS
LYNC2010AD-RTM
Forward Lookup Zones
_msdcs.lync2010rtm.com
lync2010rtm.com
_msdcs
_sites
_tcp
_udp
DomainDnsZones
ForestDnsZones
MedServer34
_tcp
Reverse Lookup Zones
109.20.172.in-addr.arpa
117.20.172.in-addr.arpa
85.20.172.in-addr.arpa
Conditional Forwarders
Global Logs
DNS Events
Name
Type
Data
Timestamp
_tcp
(same as parent folder) Host (A) 172.20.117.130 static
(same as parent folder) Host (A) 172.20.117.131 static
Start
4:51 PM
8/22/2011

Lync Server 2010 Site Topology Overview



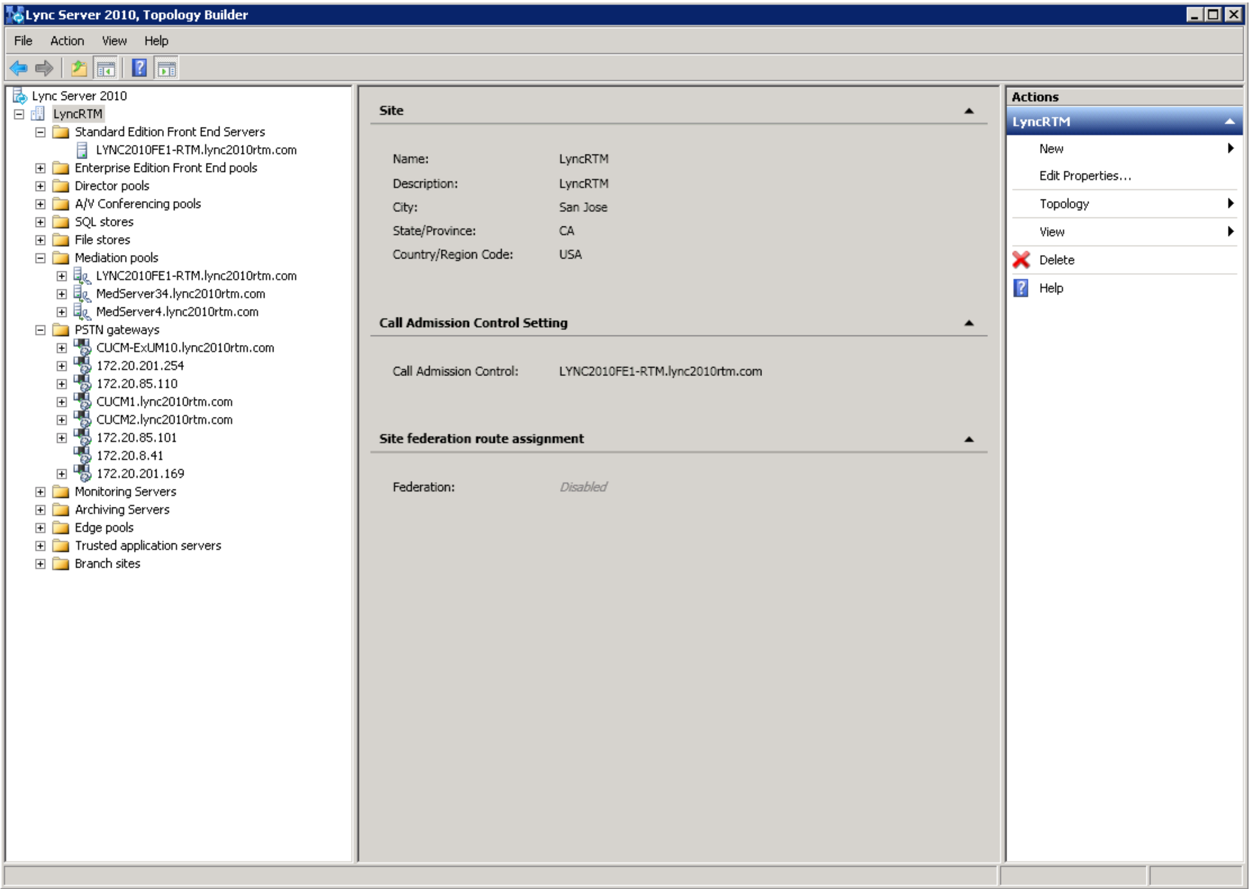


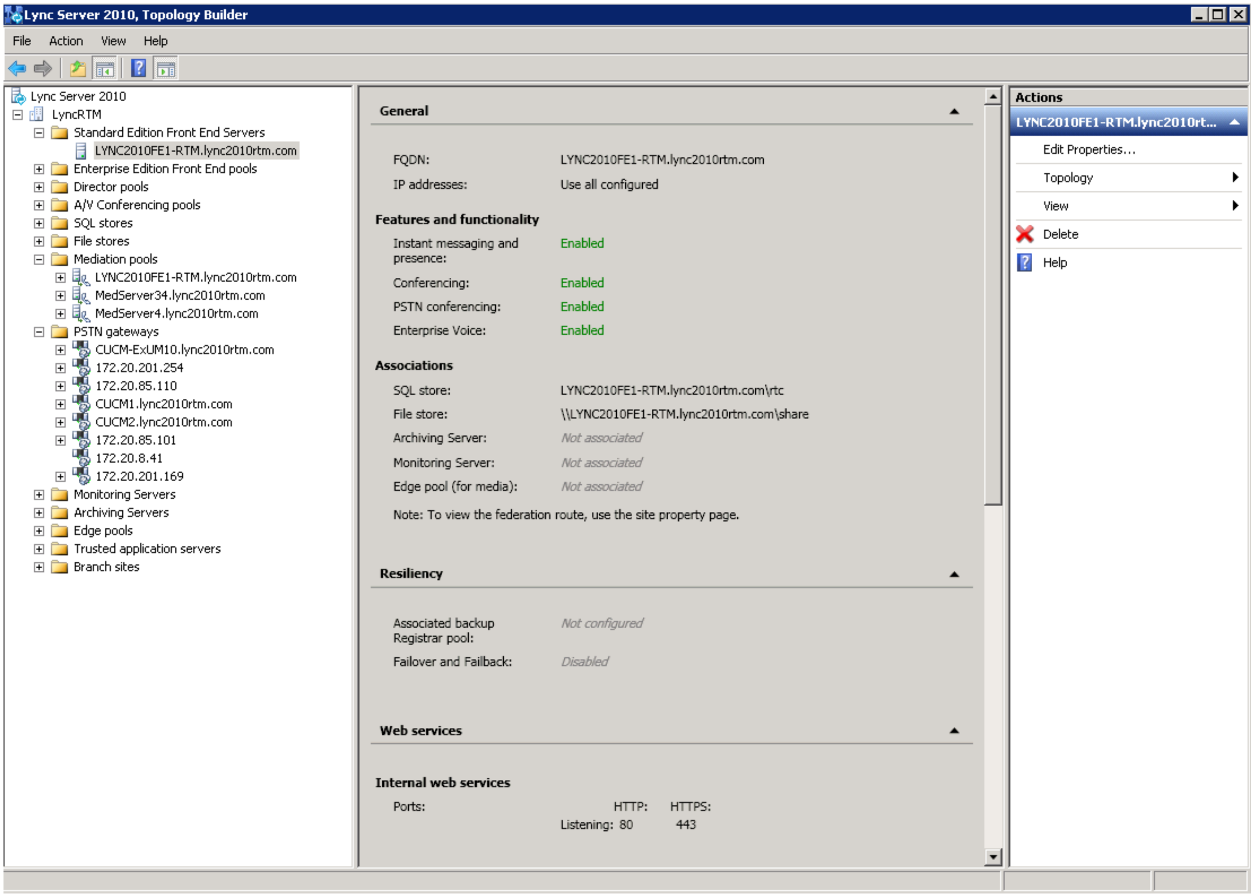
SME must be added in Topology Builder as a PSTN gateway. PSTN gateway is the generic term to refer to IP-PBX and other gateway devices. The topology must be published to Central Management store. After the PSTN gateway (that is, SME) is added to the topology, it appears in the Lync Server Control Panel. The first thing to do is to add a trunk to the IP - PBX (that is, SME)



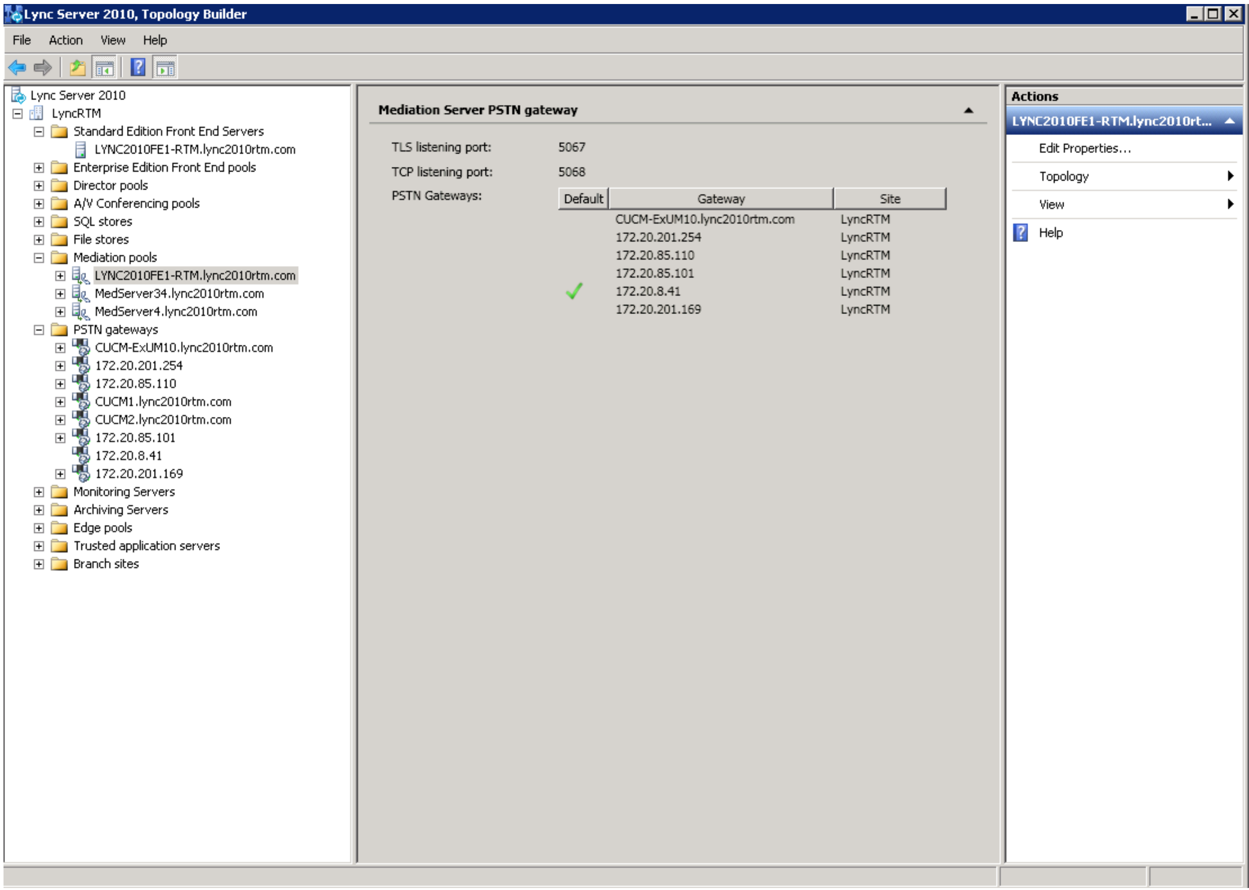
Listening port of the SME is left to default 5060.

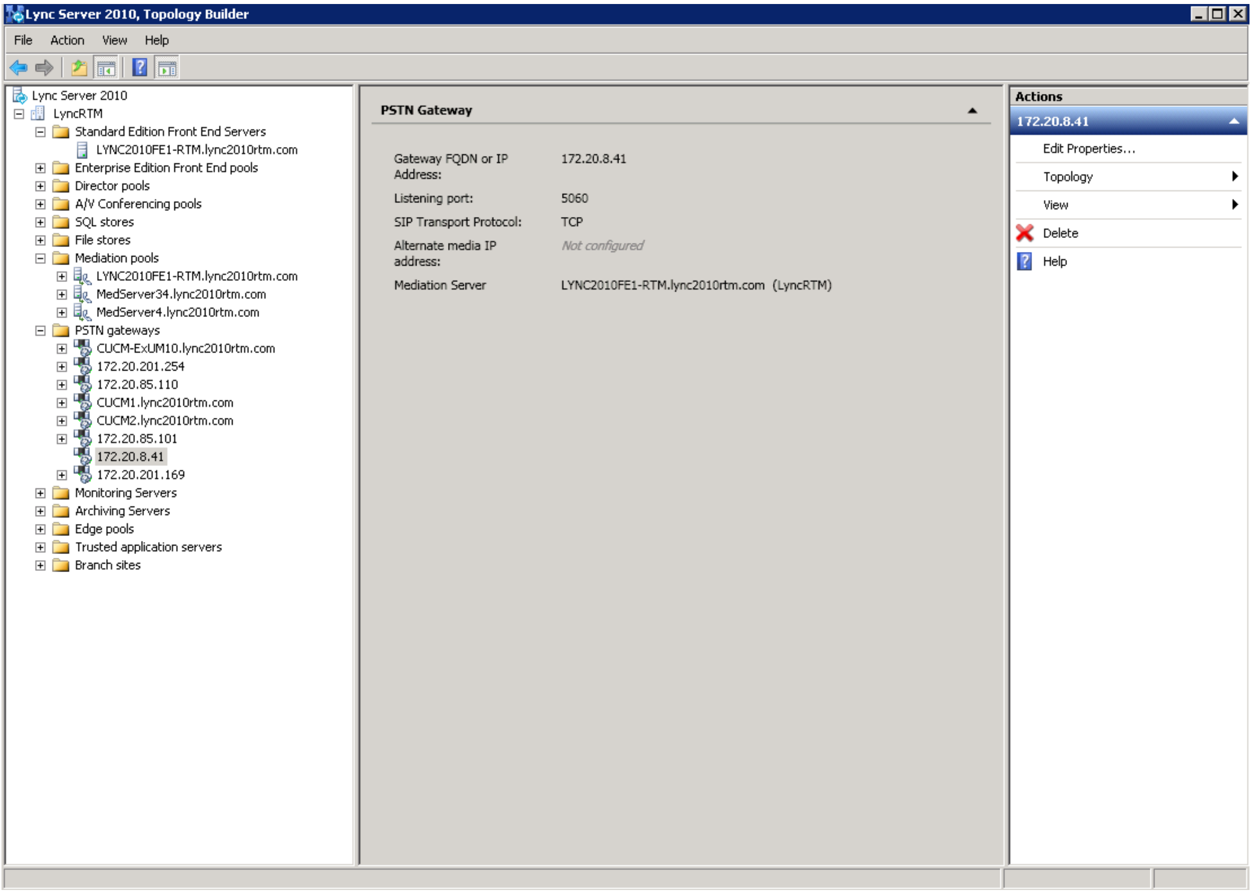
Mediation Server Overview





LYNC Front End server configuration with FQDN

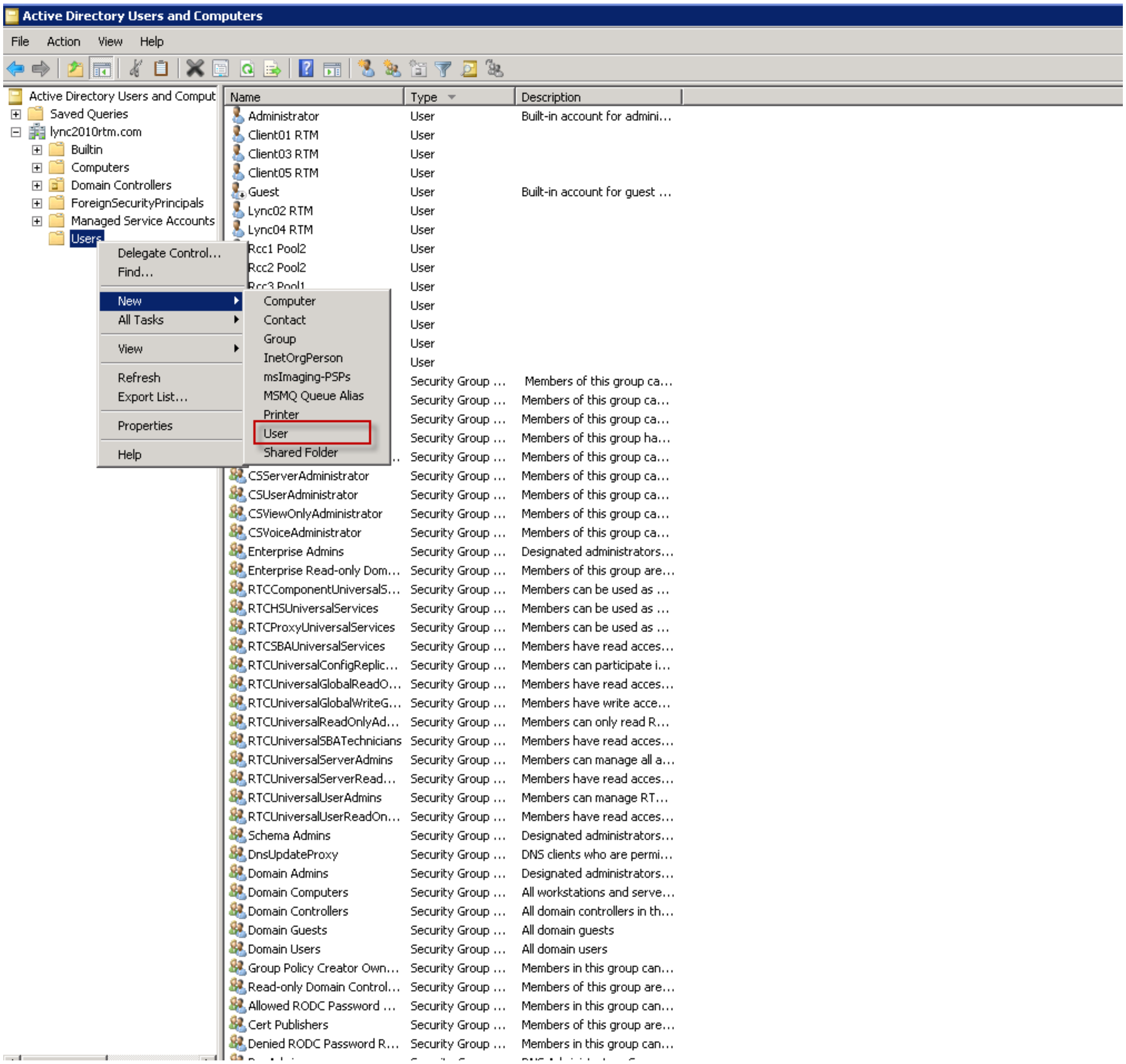

Lync Server 2010, Topology Builder
File Action View Help
Lync Server 2010
LyncRTM
Standard Edition Front End Servers
LYNC2010FE1-RTM.lync2010rtm.com
Enterprise Edition Front End pools
Director pools
A/V Conferencing pools
SQL stores
File stores
Mediation pools
LYNC2010FE1-RTM.lync2010rtm.com
MedServer34.lync2010rtm.com
MedServer4.lync2010rtm.com
PSTN gateways
CUCM-ExUM10.lync2010rtm.com
172.20.201.254
172.20.85.110
CUCM1.lync2010rtm.com
CUCM2.lync2010rtm.com
172.20.85.101
172.20.8.41
172.20.201.169
Monitoring Servers
Archiving Servers
Edge pools
Trusted application servers
Branch sites
Mediation Server PSTN gateway
TLS listening port: 5067
TCP listening port: 5068
PSTN Gateways:
Default Gateway Site
CUCM-ExUM10.lync2010rtm.com LyncRTM
172.20.201.254 LyncRTM
172.20.85.110 LyncRTM
172.20.85.101 LyncRTM
172.20.8.41 LyncRTM
172.20.201.169 LyncRTM
Actions
LYNC2010FE1-RTM.lync2010rt...
Edit Properties...
Topology
View
Help

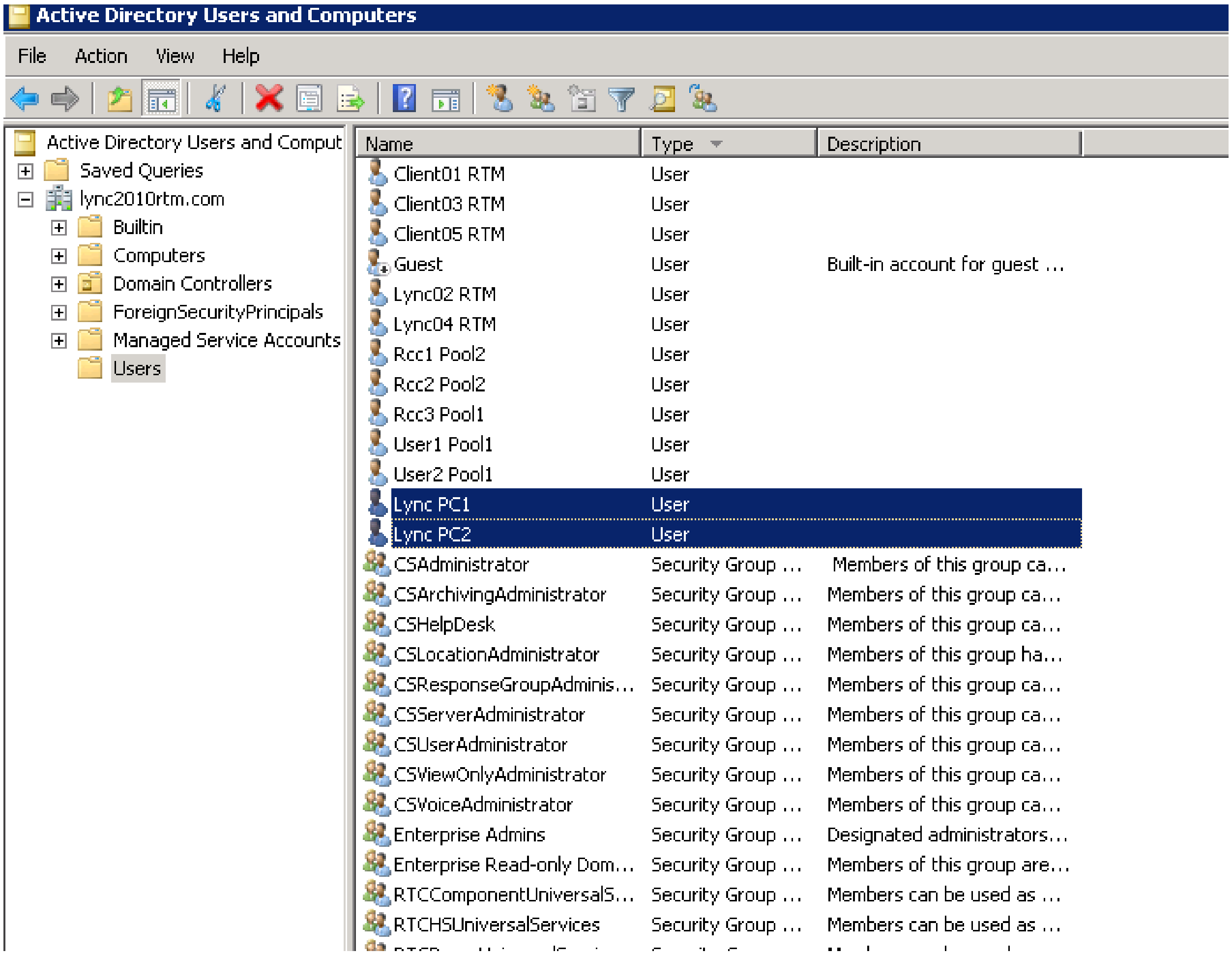


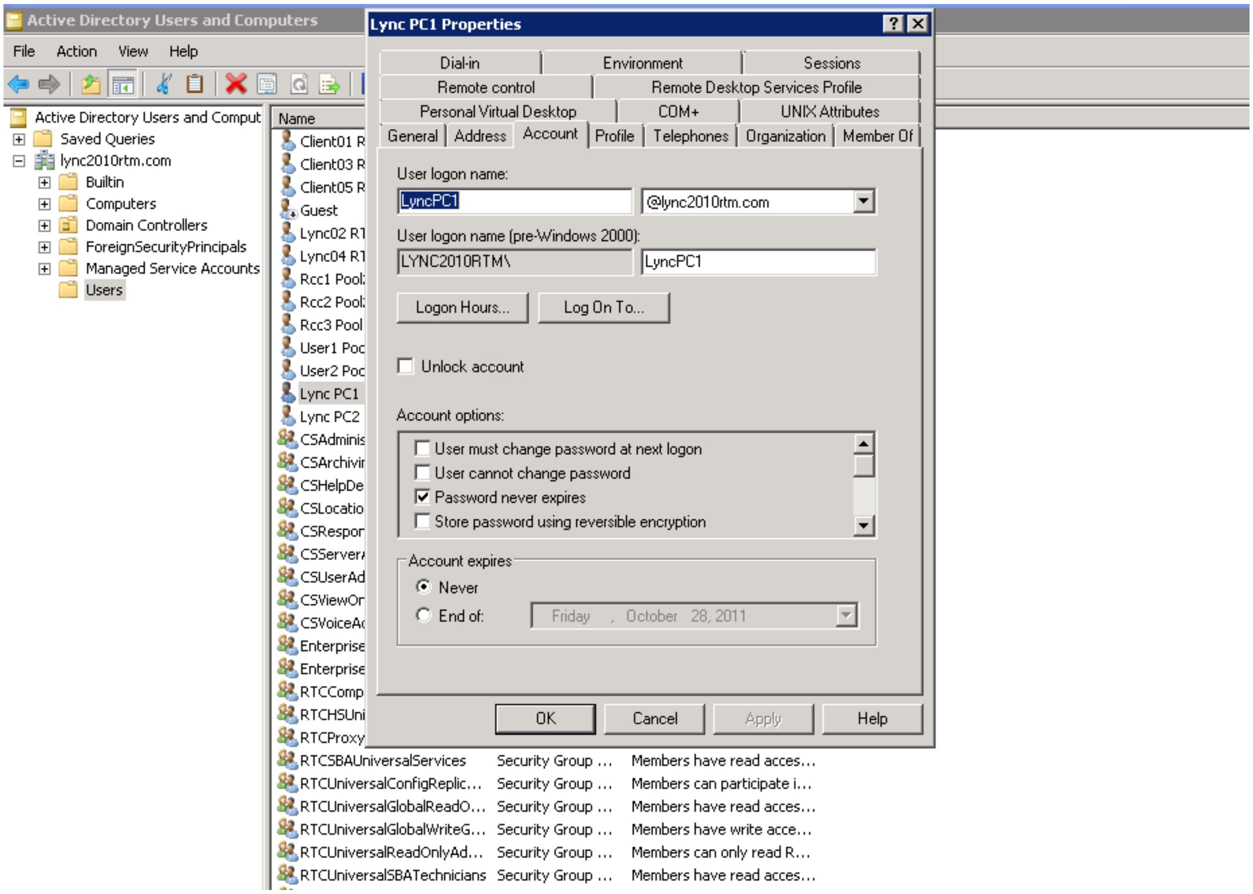
PSTN Gateway associated with the Mediation Pool

Active Directory User Configuration

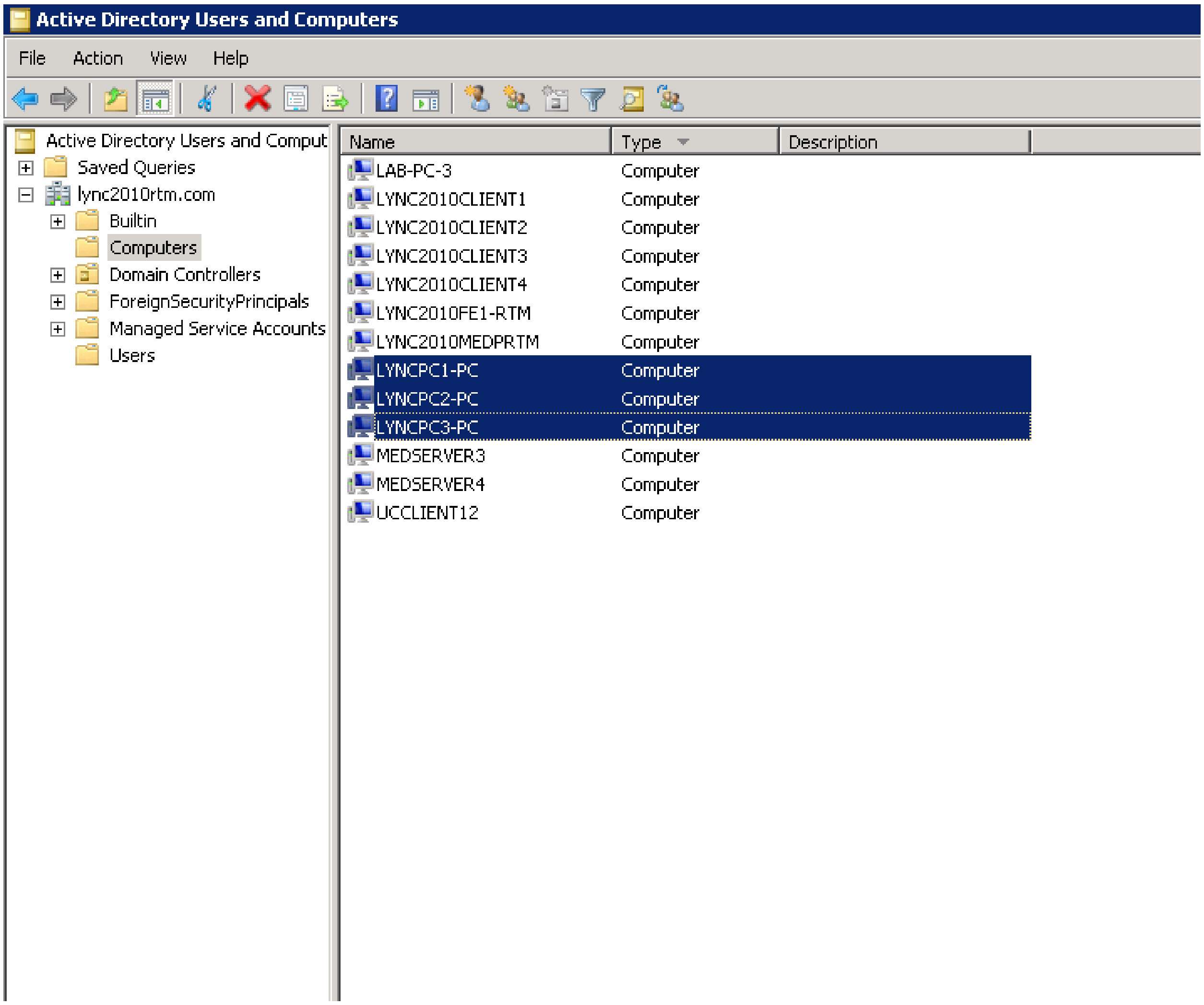
Create users from Front End Server by accessing the Active Directory Users and Computers window




Active Directory Users and Computers
File
Action
View
Help
Active Directory Users and Comput
Saved Queries
lync2010rtm.com
Builtin
Computers
Domain Controllers
ForeignSecurityPrincipals
Managed Service Accounts
Users
Name
Type
Description
Client01 RTM User
Client03 RTM User
Client05 RTM User
Guest User Built-in account for guest ...
Lync02 RTM User
Lync04 RTM User
Rcc1 Pool2 User
Rcc2 Pool2 User
Rcc3 Pool1 User
User1 Pool1 User
User2 Pool1 User
Lync PC1 User
Lync PC2 User
CSAdministrator Security Group ... Members of this group ca...
CSArchivingAdministrator Security Group ... Members of this group ca...
CSHelpDesk Security Group ... Members of this group ca...
CSLocationAdministrator Security Group ... Members of this group ha...
CSResponseGroupAdminis... Security Group ... Members of this group ca...
CSServerAdministrator Security Group ... Members of this group ca...
CSUserAdministrator Security Group ... Members of this group ca...
CSViewOnlyAdministrator Security Group ... Members of this group ca...
CSVoiceAdministrator Security Group ... Members of this group ca...
Enterprise Admins Security Group ... Designated administrators...
Enterprise Read-only Dom... Security Group ... Members of this group are...
RTCComponentUniversalS... Security Group ... Members can be used as ...
RTCHSUniversalServices Security Group ... Members can be used as ...


Active Directory Users and Computers
File Action View Help
Active Directory Users and Comput
Saved Queries
lync2010rtm.com
Builtin
Computers
Domain Controllers
ForeignSecurityPrincipals
Managed Service Accounts
Users
Name
Guest
Lync PC1
Lync PC2
RTCSBAUniversalServices Security Group ... Members have read acces...
RTCUniversalConfigReplic... Security Group ... Members can participate i...
RTCUniversalGlobalReadO... Security Group ... Members have read acces...
RTCUniversalGlobalWriteG... Security Group ... Members have write acce...
RTCUniversalReadOnlyAd... Security Group ... Members can only read R...
RTCUniversalSBATechnicians Security Group ... Members have read acces...
Lync PC1 Properties
Dial-in
Environment
Sessions
Remote control
Remote Desktop Services Profile
Personal Virtual Desktop
COM+
UNIX Attributes
General
Address
Account
Profile
Telephones
Organization
Member Of
User logon name:
LyncPC1
@lync2010rtm.com
User logon name (pre-Windows 2000):
LYNC2010RTM\
LyncPC1
Logon Hours...
Log On To...
Unlock account
Account options:
User must change password at next logon
User cannot change password
Password never expires
Store password using reversible encryption
Account expires
Never
End of:
Friday , October 28, 2011
OK
Cancel
Apply
Help

SME



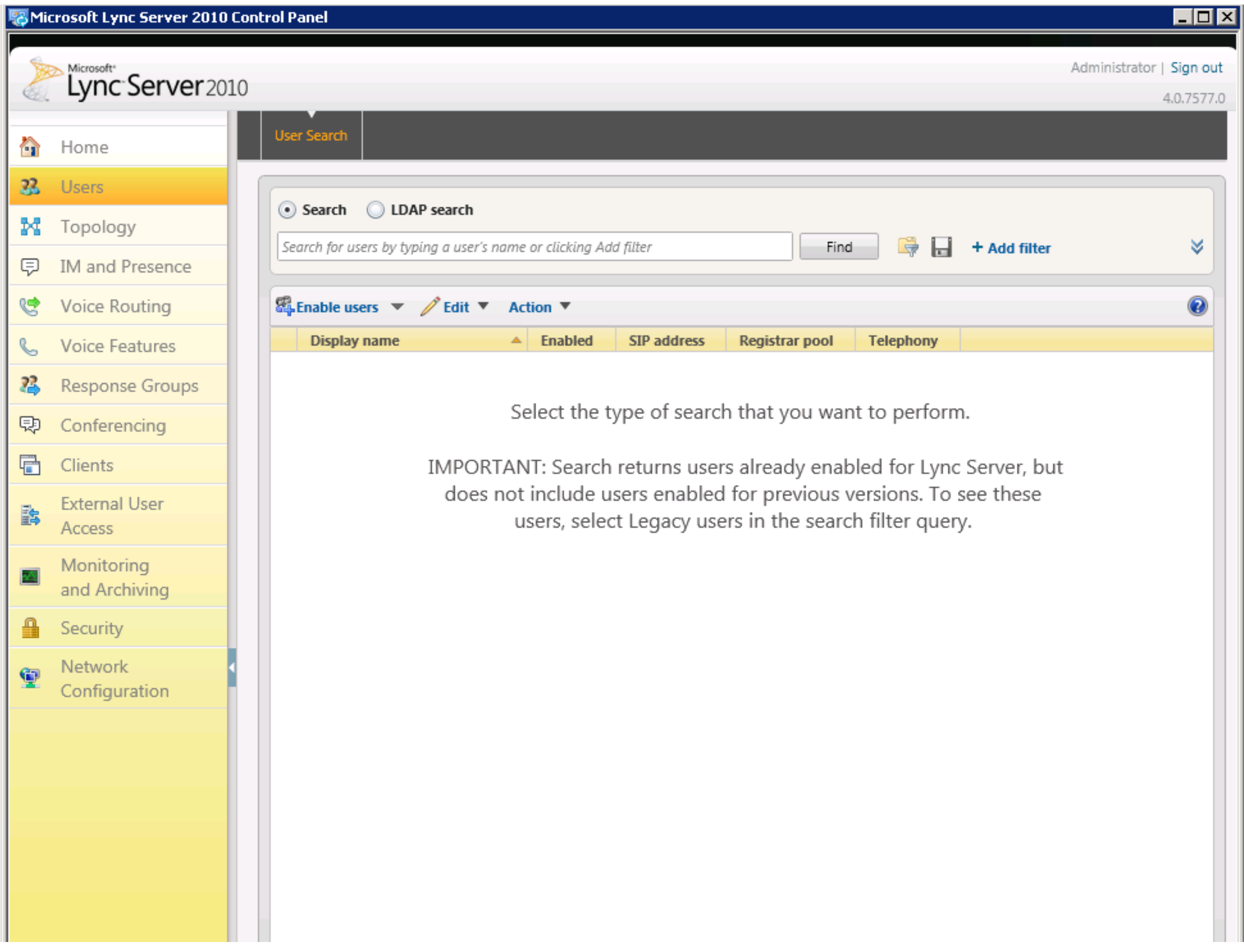
Computers on the Activ Directory running LYNC clients

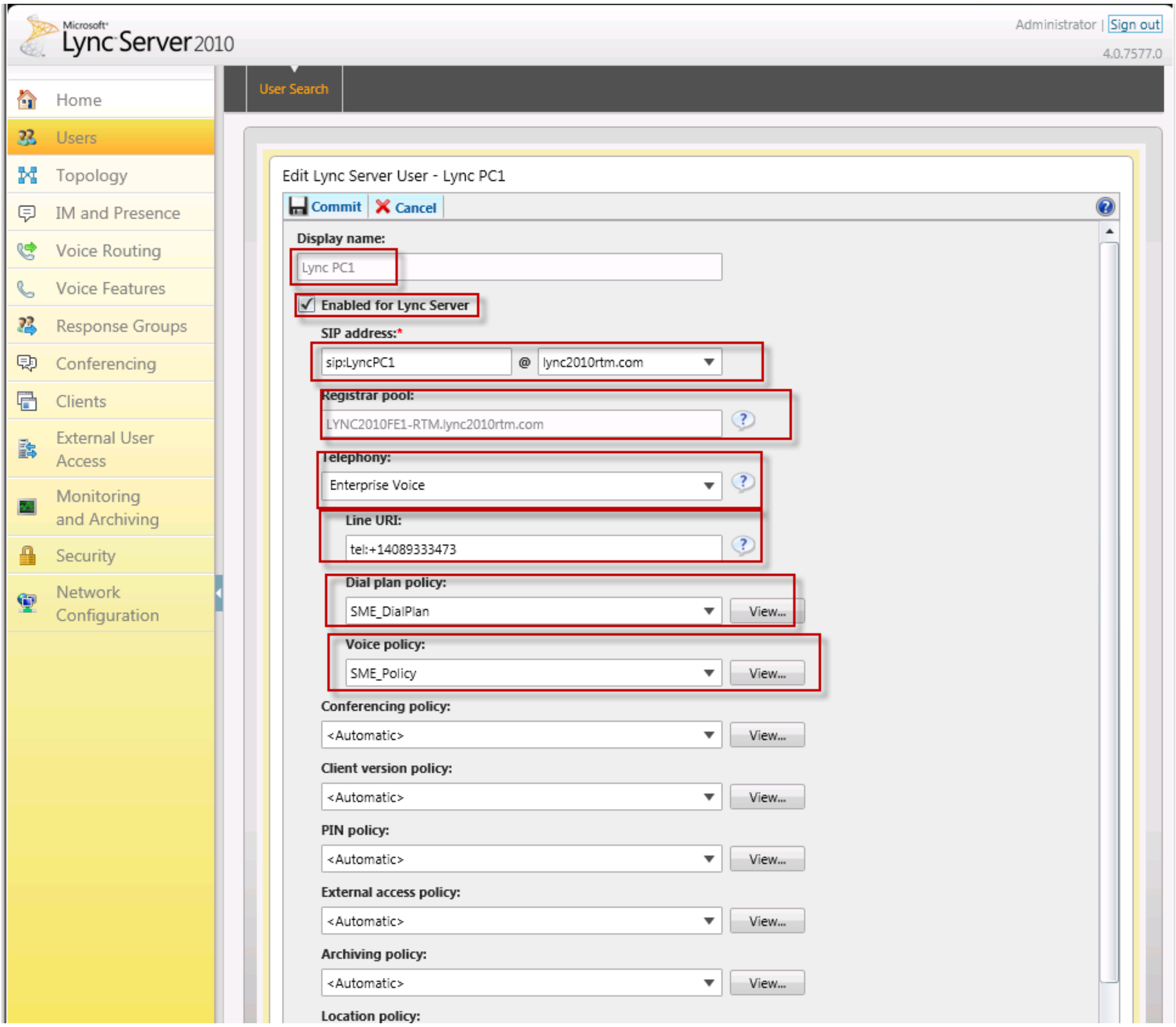
Lync Server 2010 Configuration

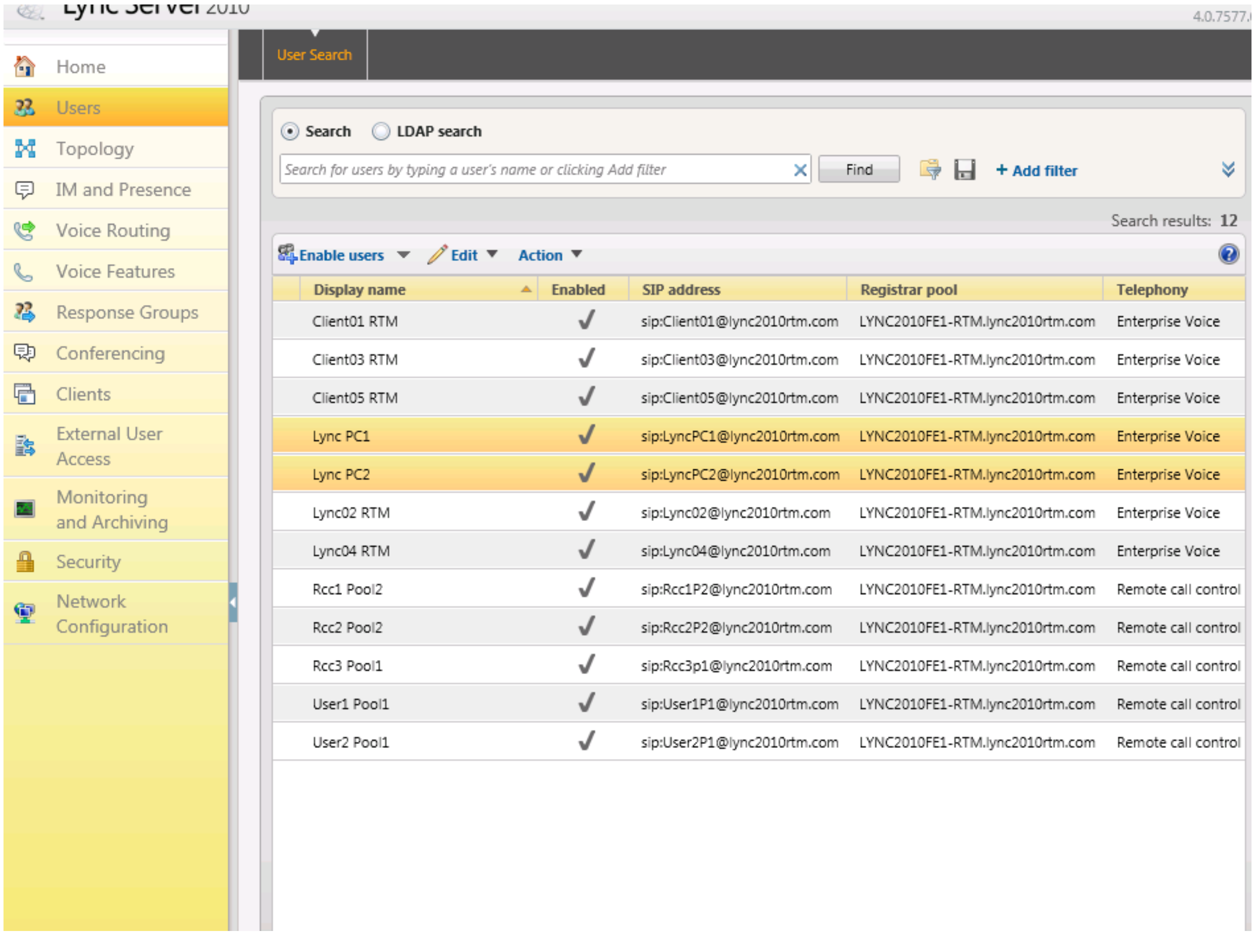
Lync Server configuration can be done in a couple of ways—through the Microsoft Lync Server Control Panel or through Lync Server Management Shell. For the purposes of configuring a direct SIP connection with CUCM, we will illustrate the configuration by using the Lync Server Control Panel and Topology Builder.

User Configuration from Control Panel

Start > All Programs > Microsoft Lync Server 2010 > Lync Server Control Panel
To add users got to Home > Enable users for Lync Server and then go to Users > Add and then do a find as shown below.

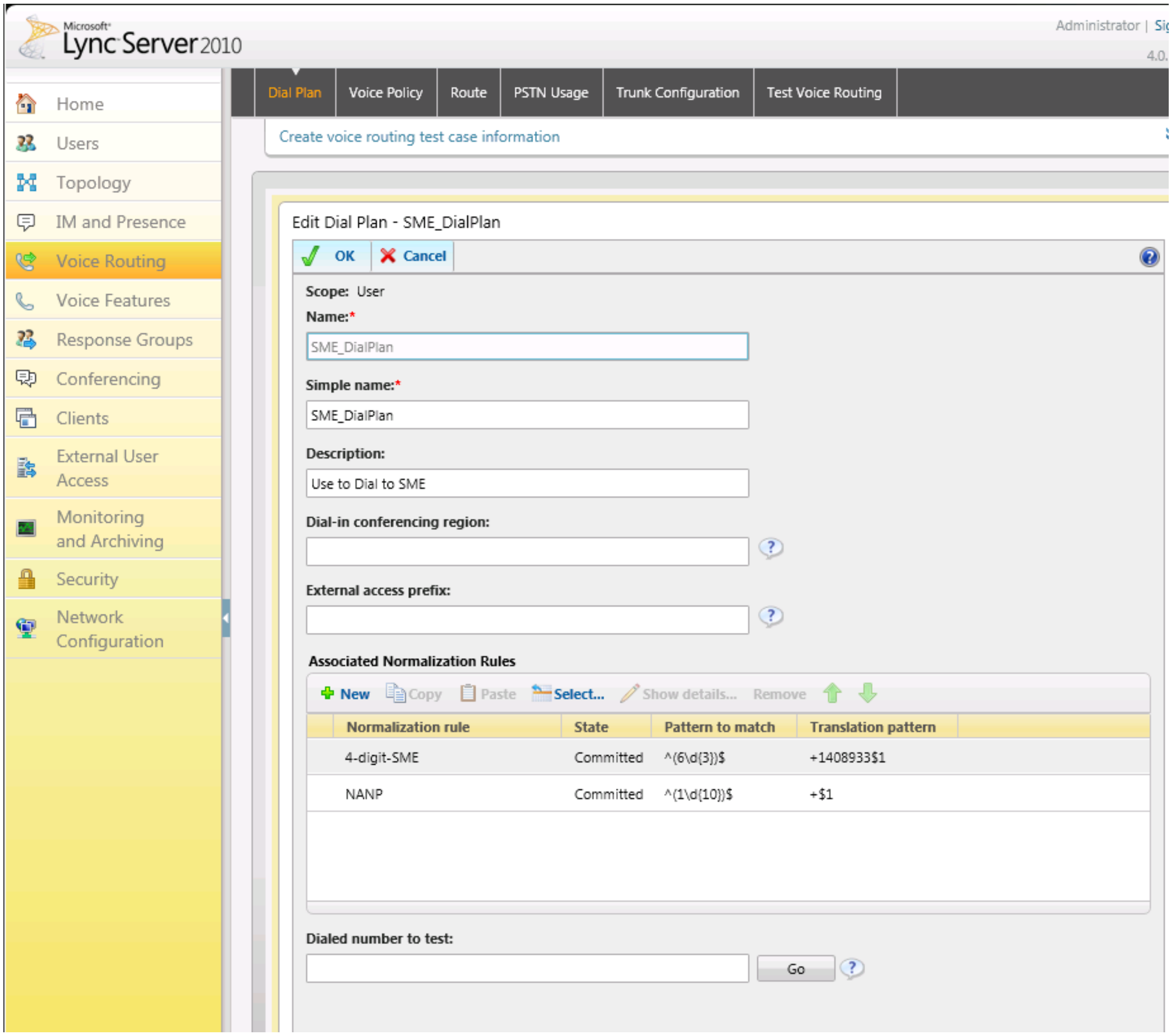



Microsoft Lync Server 2010
Administrator | Sign out
4.0.7577.0
Home
Users
Topology
IM and Presence
Voice Routing
Voice Features
Response Groups
Conferencing
Clients
External User Access
Monitoring and Archiving
Security
Network Configuration
User Search
Edit Lync Server User - Lync PC1
Commit
Cancel
Display name:
Lync PC1
Enabled for Lync Server
SIP address:*
sip:LyncPC1
@
lync2010rtm.com
Registrar pool:
LYNC2010FE1-RTM.lync2010rtm.com
Telephony:
Enterprise Voice
Line URI:
tel:+14089333473
Dial plan policy:
SME_DialPlan
View...
Voice policy:
SME_Policy
View...
Conferencing policy:
<Automatic>
View...
Client version policy:
<Automatic>
View...
PIN policy:
<Automatic>
View...
External access policy:
<Automatic>
View...
Archiving policy:
<Automatic>
View...
Location policy:

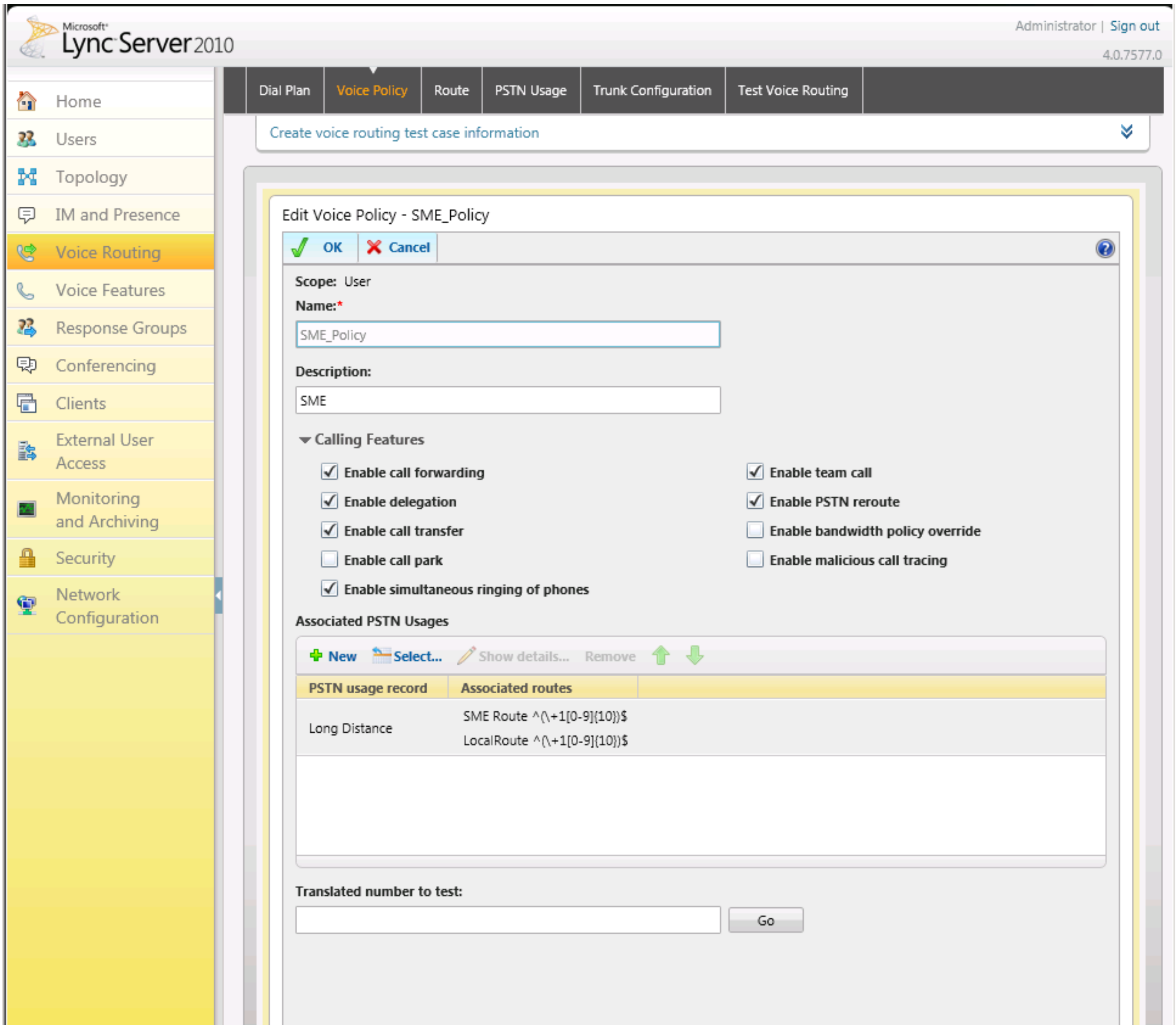


In Lync Server, best practice is to use a dial plan that is based on the E.164 standard. This allows easier routing and troubleshooting as well as a scalable model for growth. It is best practices to represent numbers in an E.164 format (External DIDs as well as internal extensions).

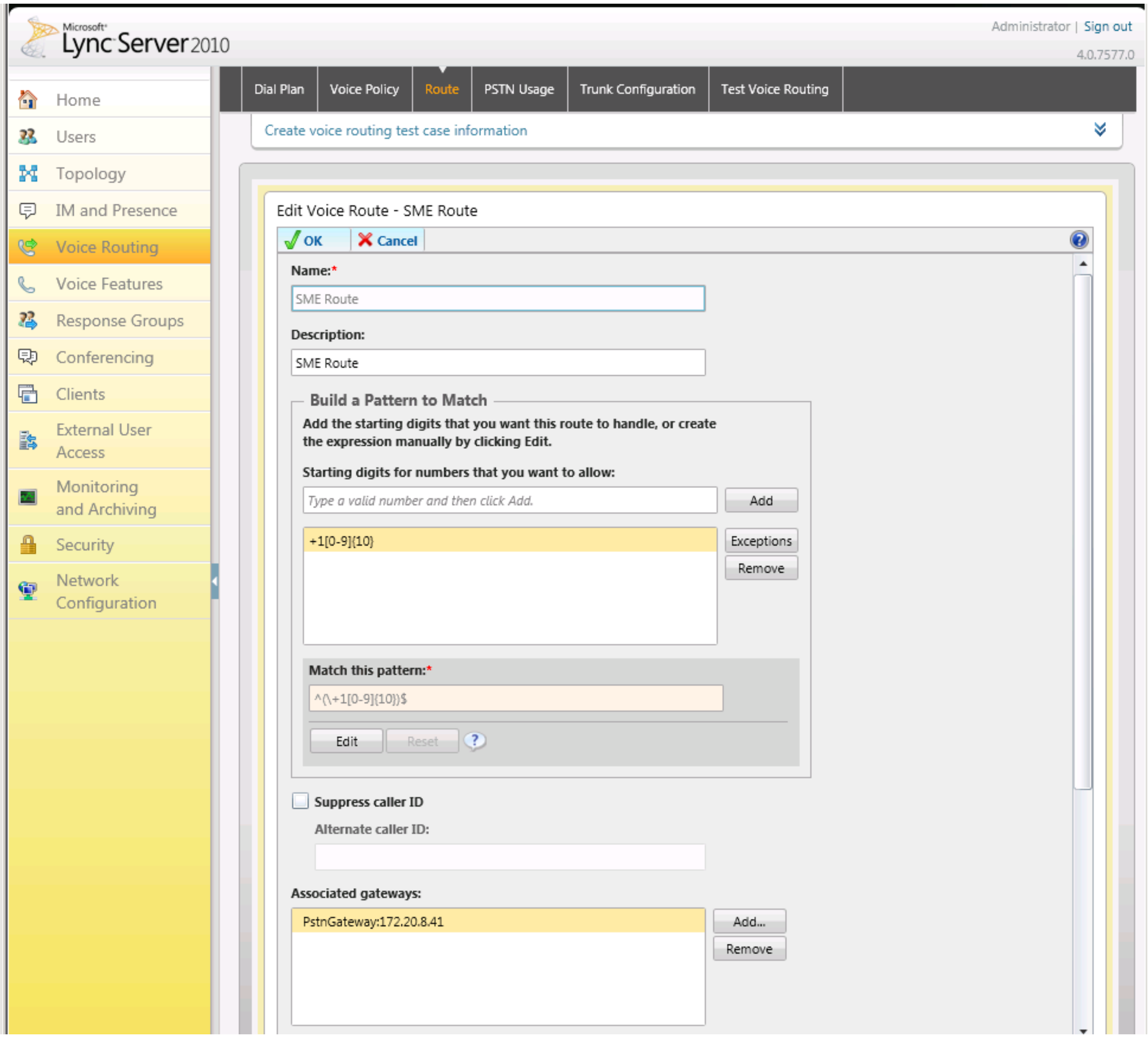
Voice Routing Configuration



Lync Server normalizes all outbound calls as E.164. This allows uniform routing that scales globally across the Lync Server deployment. Phone numbers are normalized to E.164 by normalization rules in this case the rule 4dig-ext. Normalization rules are added in the Dial Plan tab.

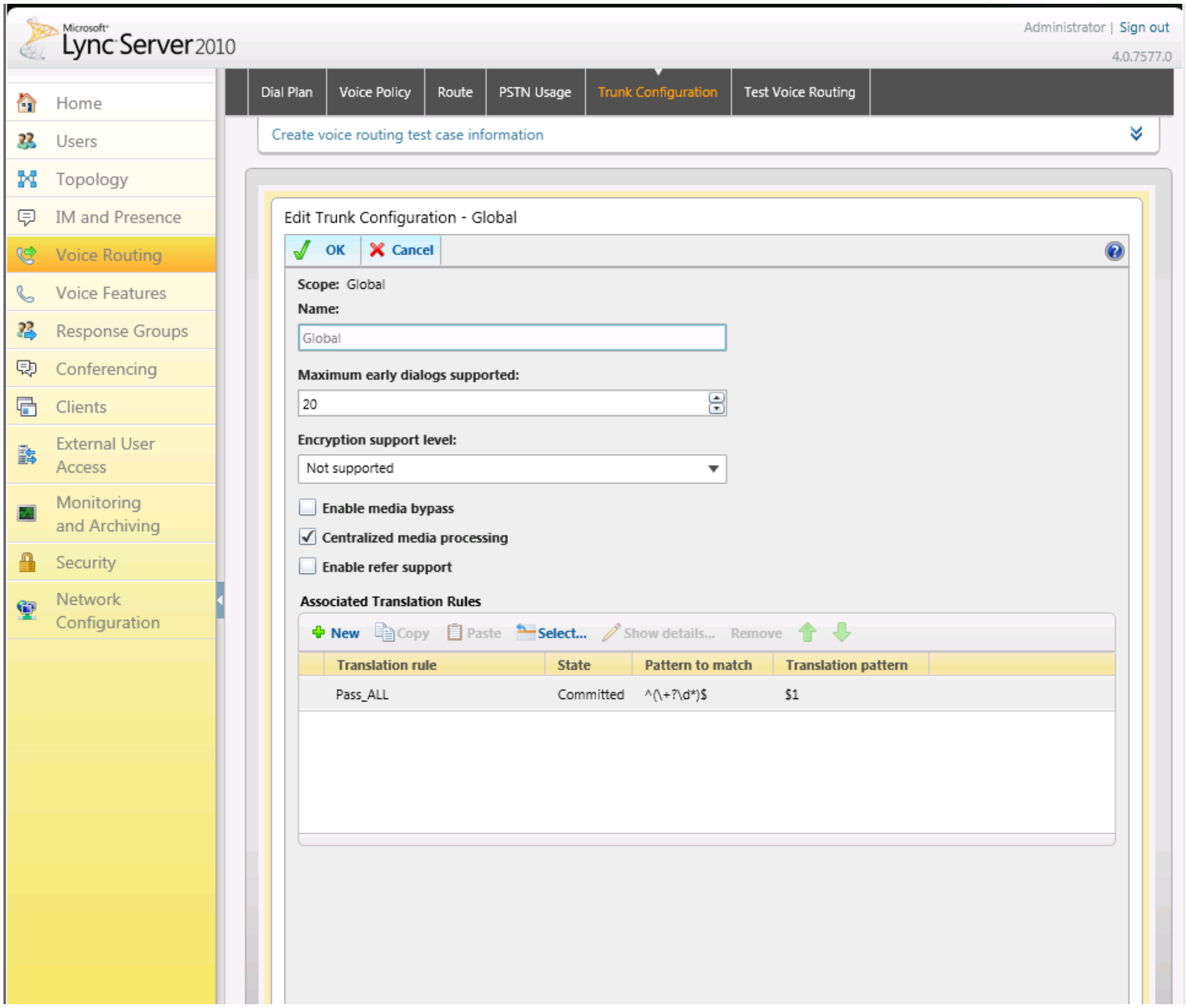


PSTN usage is associated on the Voice Policy. Calling features are selected in Voice policy tab. New user policy is created to dial to the SME.



Route to SME is associated with the PSTN gateway(SME in our case) here.

After the trunk is configured, routes can be configured to route CUCM extensions to the trunk to SME. If PSTN connectivity is configured through SME, calls to PSTN from Lync 2010 can be routed to SME through the trunk.


Microsoft Lync Server 2010
Administrator | Sign out
4.0.7577.0
Home
Users
Topology
IM and Presence
Voice Routing
Voice Features
Response Groups
Conferencing
Clients
External User Access
Monitoring and Archiving
Security
Network Configuration
Dial Plan
Voice Policy
Route
PSTN Usage
Trunk Configuration
Test Voice Routing
Create voice routing test case information
Edit Trunk Configuration - Global
OK
Cancel
Scope: Global
Name:
Global
Maximum early dialogs supported:
20
Encryption support level:
Not supported
Enable media bypass
Centralized media processing
Enable refer support
Associated Translation Rules
New
Copy
Paste
Select...
Show details...
Remove
Translation rule
State
Pattern to match
Translation pattern
Pass_ALL
Committed
^(\+?\d*)$
$1

Lync 2010 Server Management Shell Commands

Lync Server/Client Address Book Updating
With the default server/client settings, the Address Book is not updated right away. To ensure that the Address Book is updated with the latest users added to the Active Directory and their configurations, force the update on the server side. Then force the Lync 2010 Client to pull down the latest files to update its local GalContacts.db file.

On Lync Server 2010, enter the following command in the Lync Server Management Shell:

Update-CsAddressBook

This triggers the Lync Server to synchronize current Active Directory information in the SQL database into the downloadable client and device address book files. Wait 5 minutes for this process to complete.

On Lync Client 2010, execute the following command from the Windows Command Prompt run as an administrator:

reg add HKLM\Software\Policies\Microsoft\Communicator /v GalDownloadInitialDelay /t REG_DWORD /d 0 /f

Setting this value to 0 will force Lync to immediately download the address book instead of randomly selecting a time to check the server.

On Lync Client 2010, if the **GalContacts.db** and **GalContacts.db.idx** files already exist, delete them from the user's profile directory (directory location may vary depending on your Client OS). Make sure you exit and restart the Lync Client, after log in you should see a new set of files downloaded, and you should see the latest updated users appear during a search for contacts.

Lync 2010 Server draining mode
Stop-CsWindowsService -Graceful rtcmedsrv

Some Lync2010 useful commands
Get-CsTrunkConfiguration
Set-CsTrunkConfiguration
Get-CsMediaConfiguration
Set-CsMediaConfiguration

PS C:\Users\administrator.OCS2010> Get-CsTrunkConfiguration

```
Identity                          : Global
OutboundTranslationRulesList      : {}
SipResponseCodeTranslationRulesList : {}
Description                       :
ConcentratedTopology              : True
EnableBypass                      : False
EnableMobileTrunkSupport          : False
EnableReferSupport                : True
```

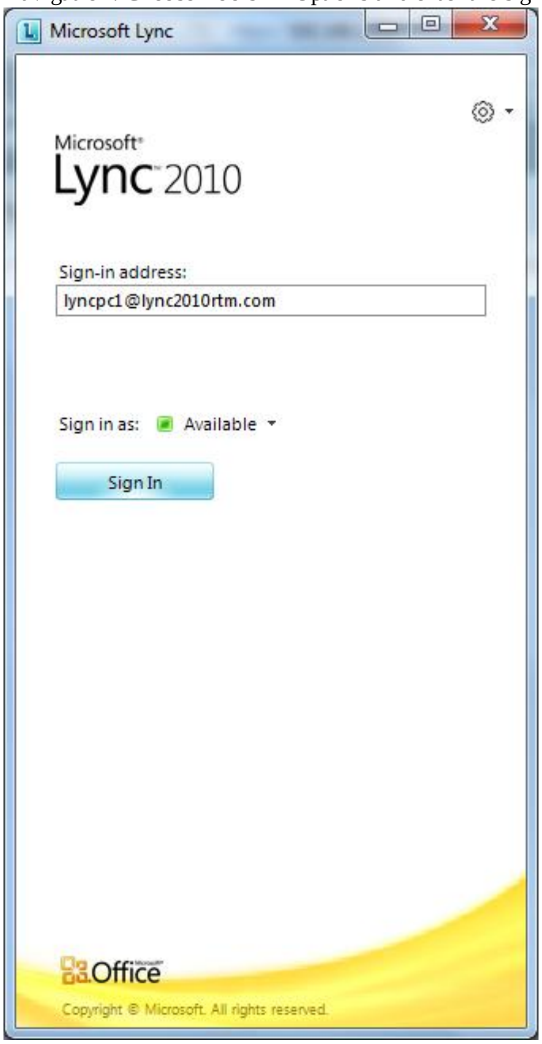
```
EnableSessionTimer          : False
EnableSignalBoost           : False
MaxEarlyDialogs             : 20
RemovePlusFromUri           : False
RTCPActiveCalls             : True
RTCPCallsOnHold             : True
SRTPMode                    : Required
EnablePIDFLOSupport         : False

PS C:\Users\administrator.OCS2010> Get-CsMediaConfiguration


Identity           : Global
EnableQoS          : False
EncryptionLevel    : RequireEncryption
EnableSiren        : False
MaxVideoRateAllowed : VGA600K
Identity           : Site:lab
EnableQoS          : False
EncryptionLevel    : RequireEncryption
EnableSiren        : False
MaxVideoRateAllowed : VGA600K
```

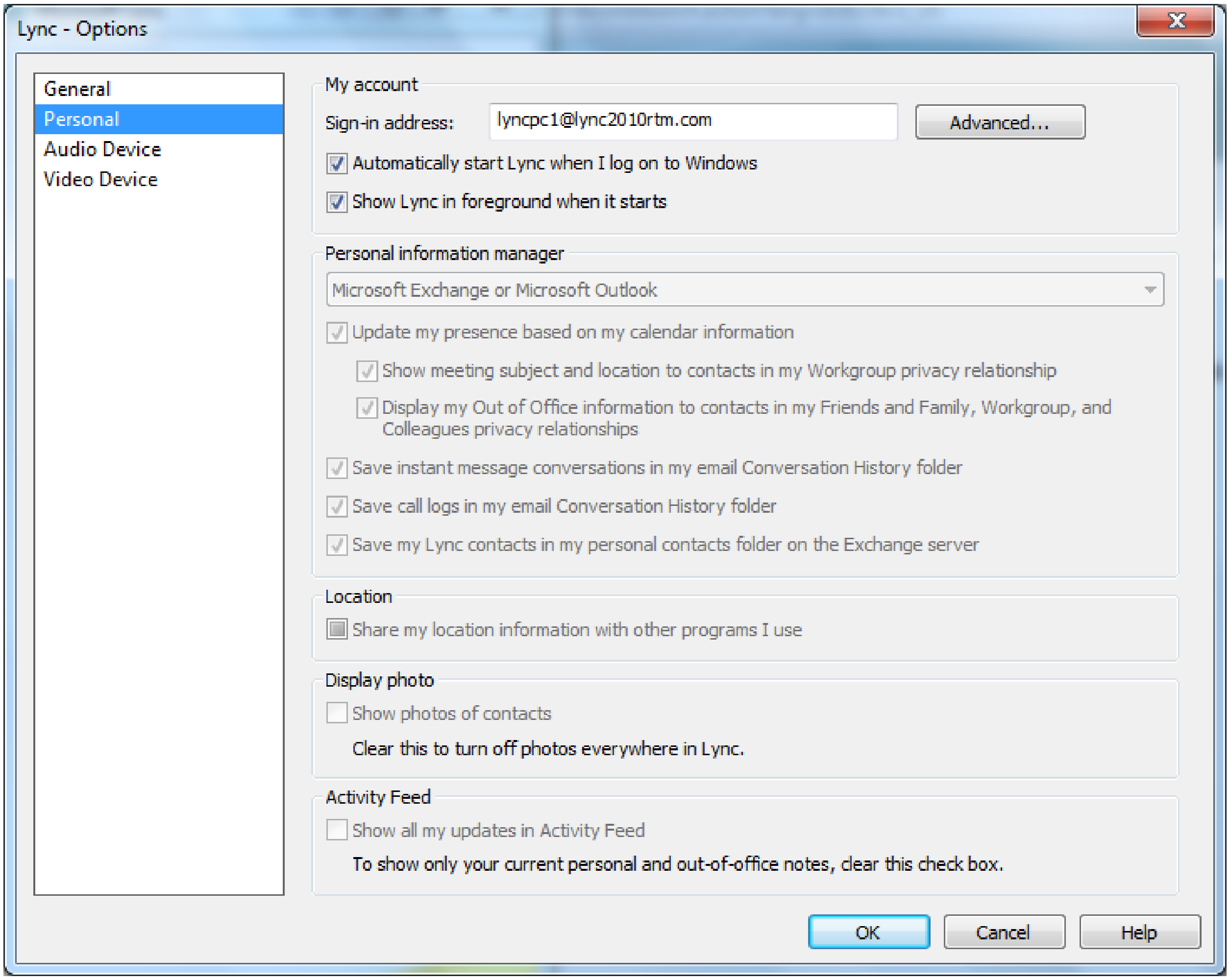
Lync2010 - enable music on hold:

set-csclientpolicy -EnableClientMusicOnHold $TRUE

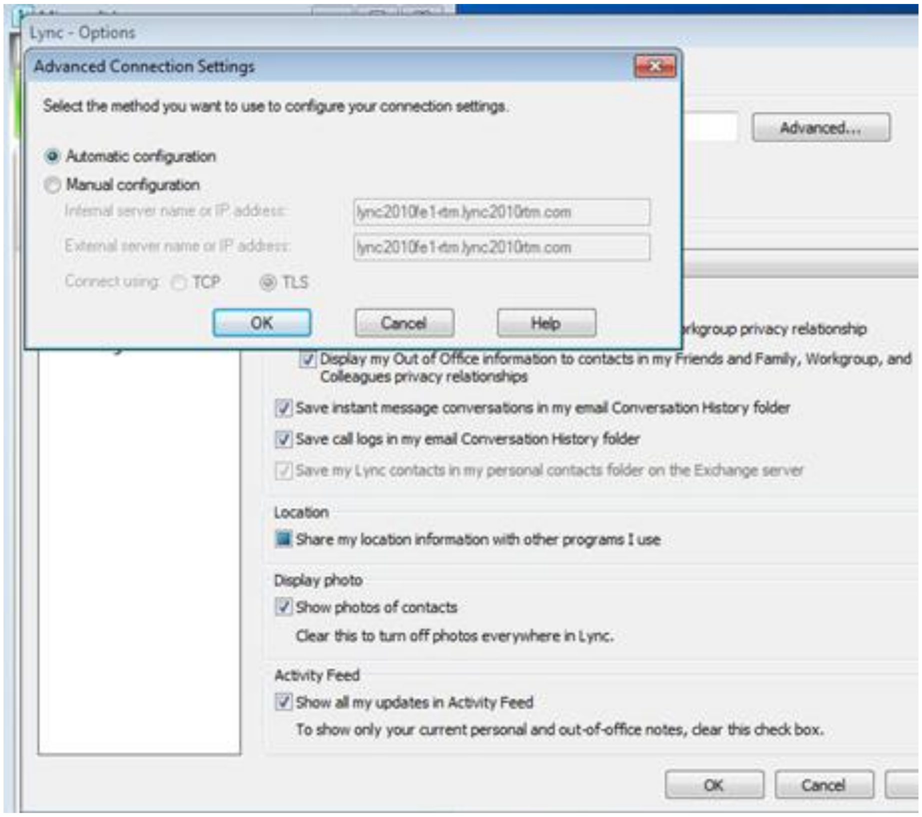
Microsoft LYNC 2010 Enterprise Voice Client Configuration

Navigation: Choose Tools → Options and enter the sign-in information.



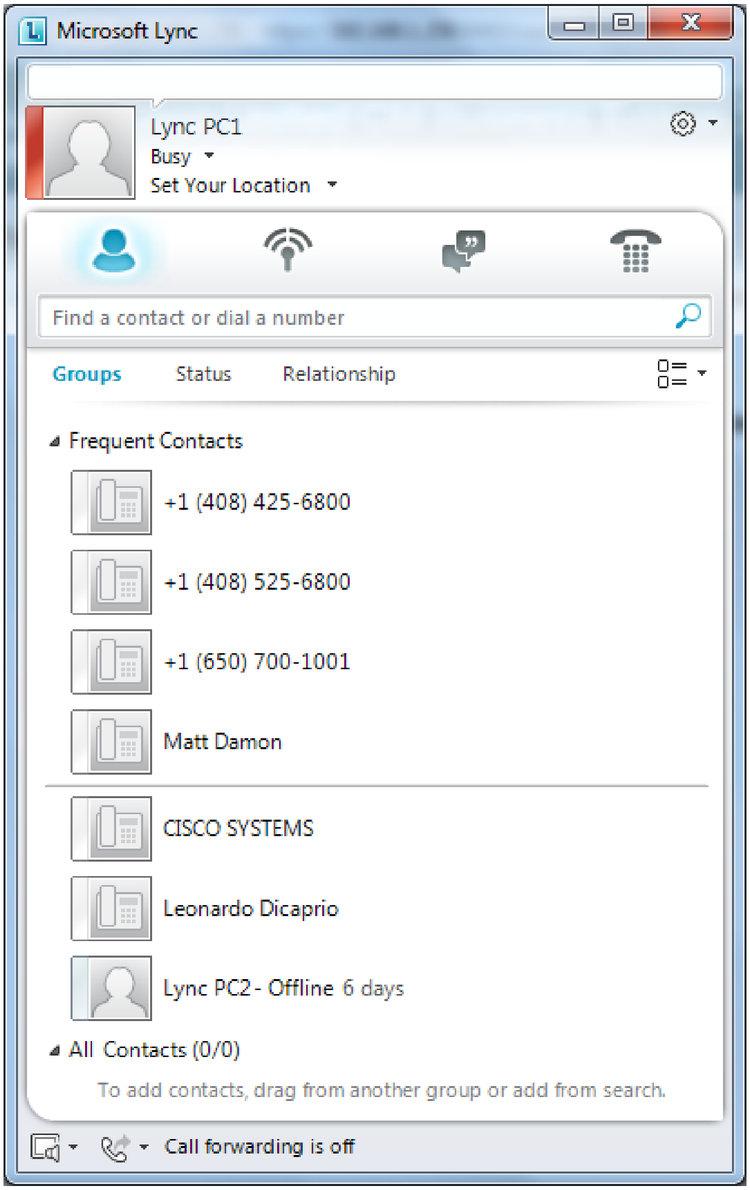


Click Advanced button to select the Advanced Connection Settings.

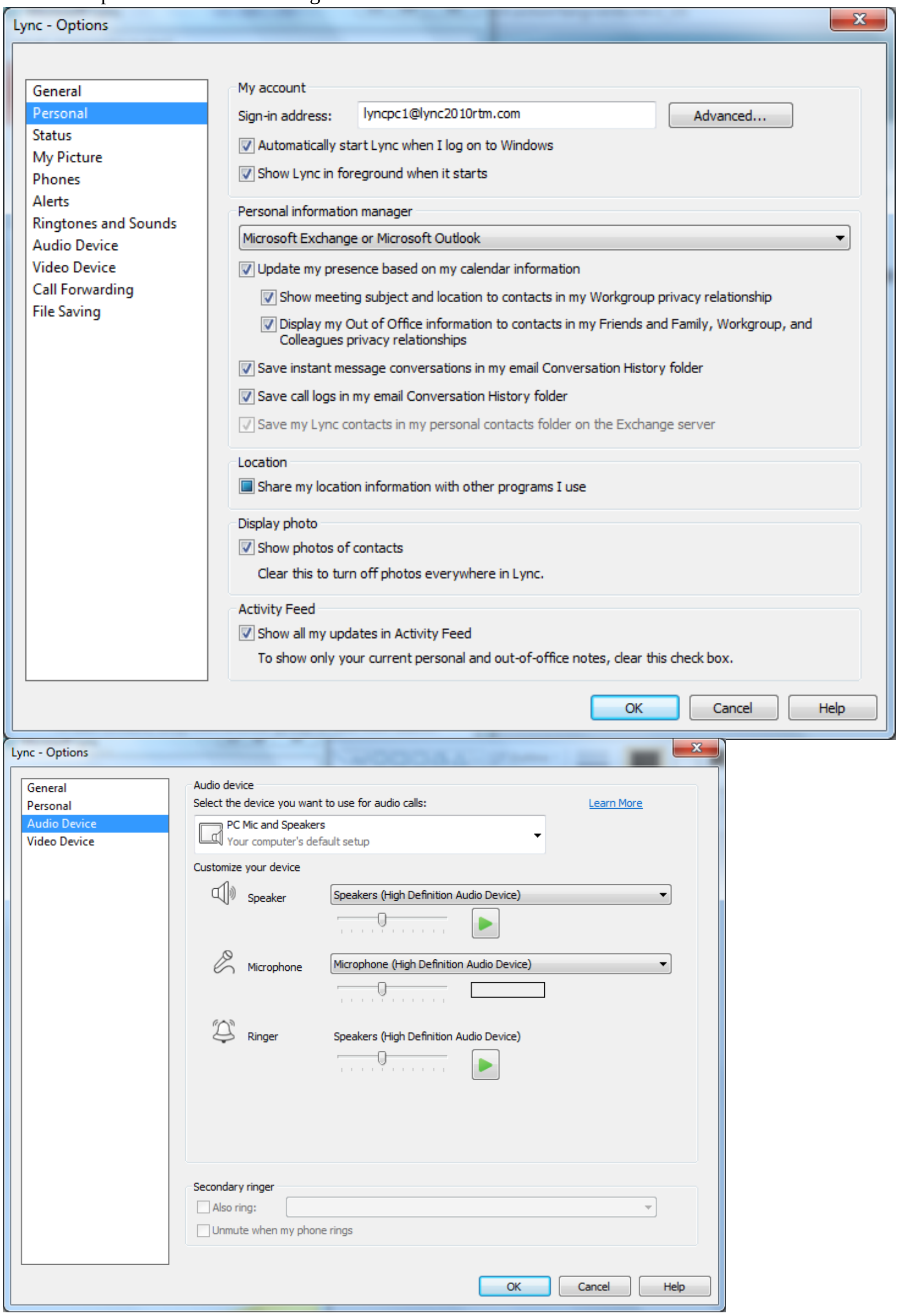


If there are DNS entries for this Microsoft Lync, automatic configuration can be used if not select manual configuration.

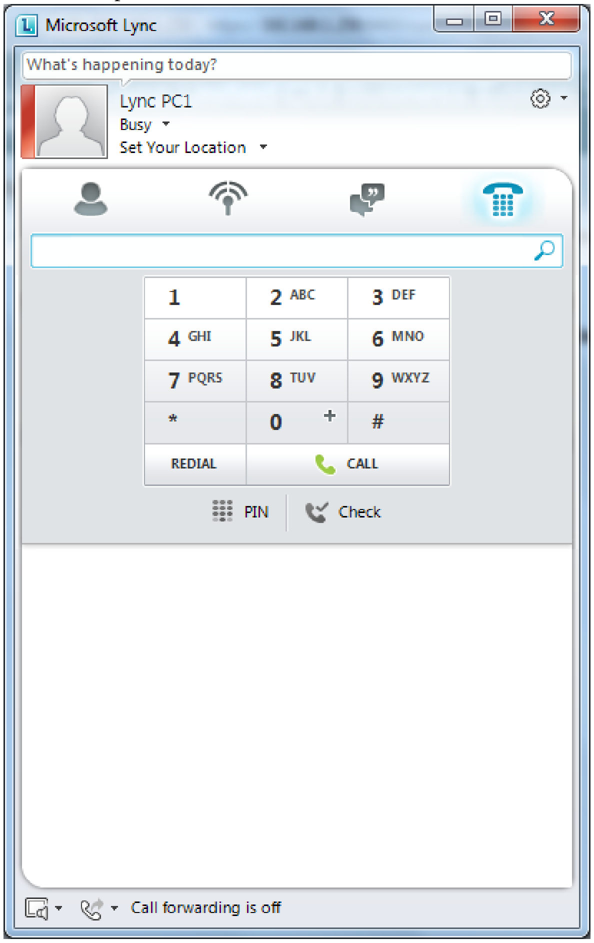
After signing add contacts



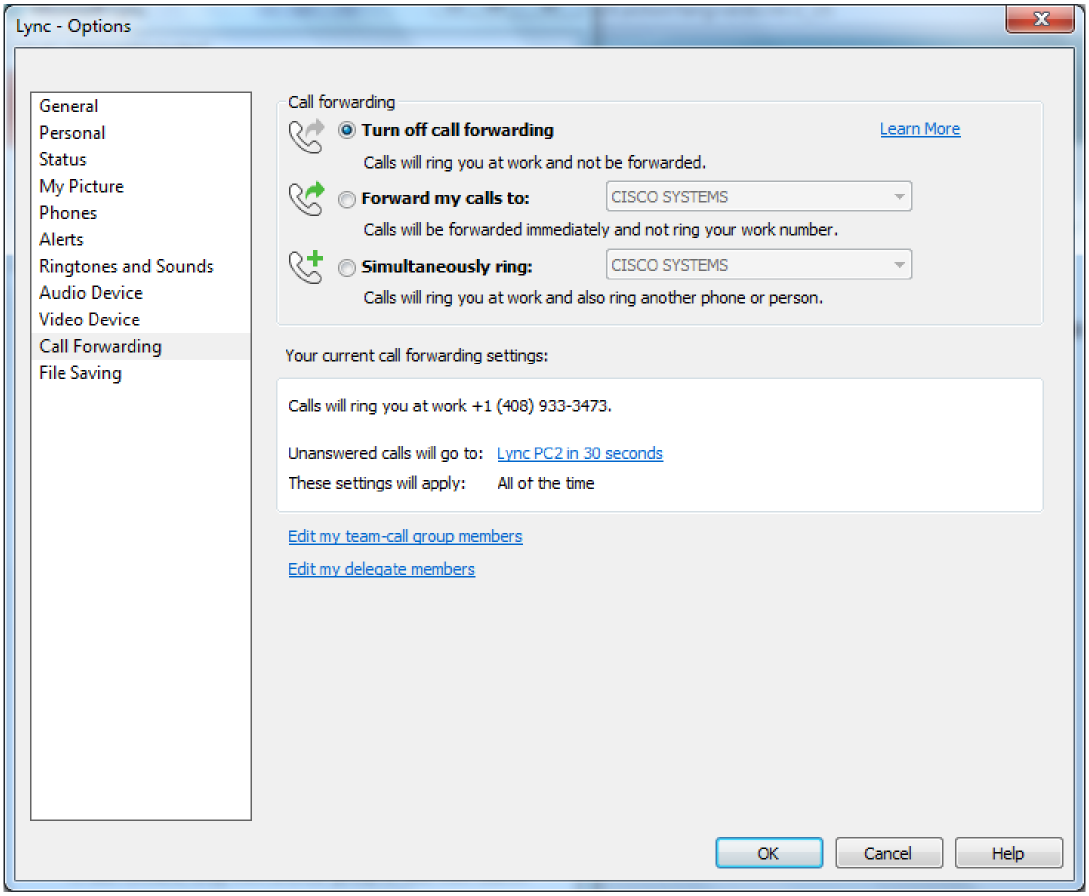
Personal Options and Audio Settings



LYNC Dialpad



Call Forwarding Configurations



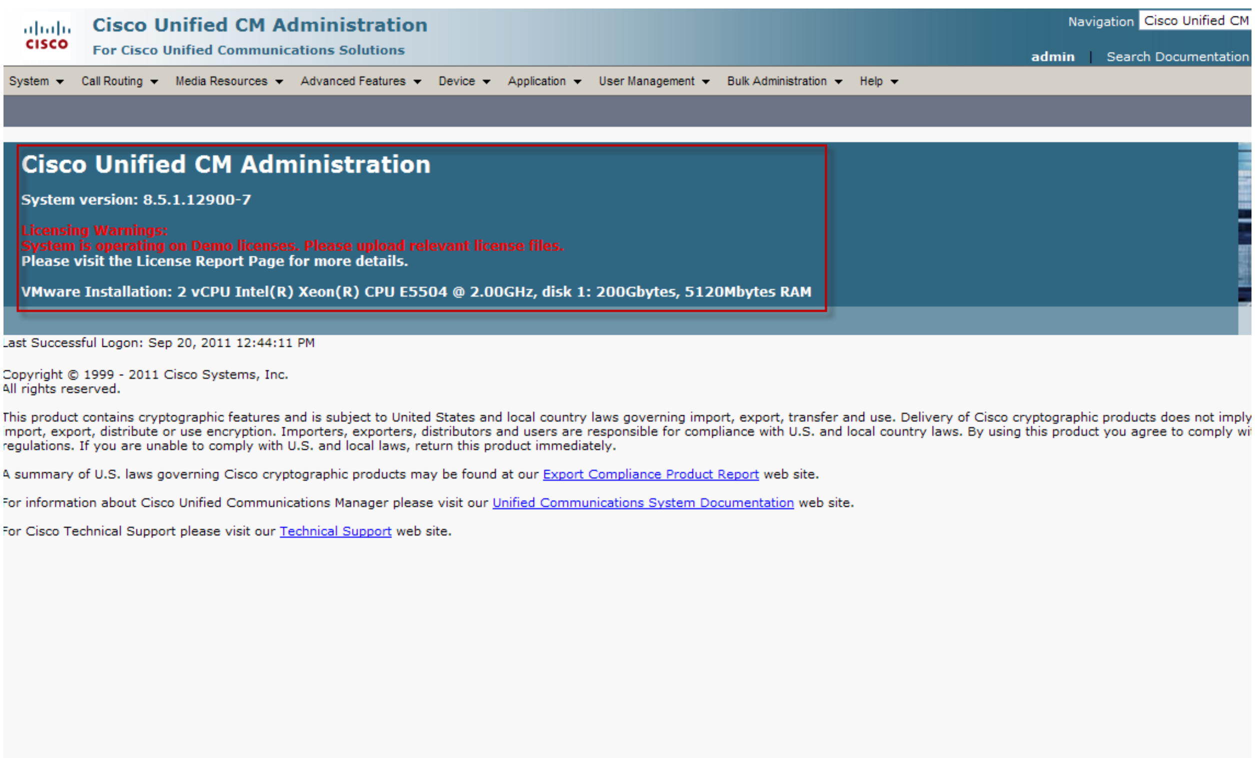
All call forward settings are selected in this tab. You can turn on/off the call forwarding settings and also select for phones to simultaneously ring.

Cisco Unified Communications Manager- Session Manager Edition Configurations

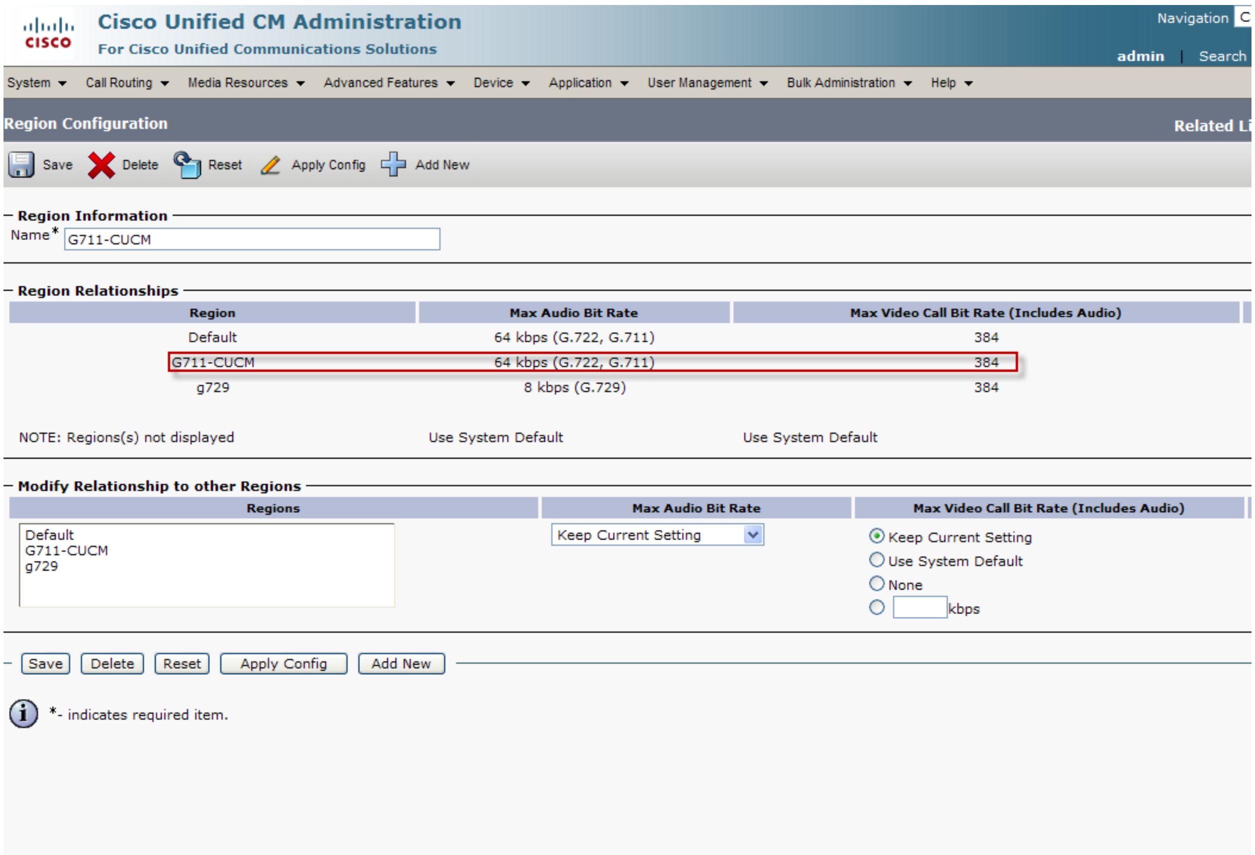
UC deployments using Unified CM Session Manager Edition are a variation on the Multi-site Distributed Call Processing deployment model and are typically used where large numbers of UC end systems need to be interconnected via a single UC system - i.e. The Unified CM Session Manager. A deployment using Cisco Unified CM Session Manager Edition is essentially a Unified CM cluster with trunk interfaces only (although IP endpoints are also allowed if required). It allows aggregation of multiple Unified Communications systems, referred to as leaf systems. UC SME is deployed to create and manager centralized dialplan, provide centralized PSTN access, aggregate PBX trunks, interconnect and interoperate across different protocols, load balance inbound and outbound calls.

Note: Cisco Session Manager Edition is set to manage dialplan as a best practice. The Cisco Unified Border Element – Enterprise edition passes all the digits over to the Session Manager Edition and the Session Manager routes the calls to the leaf nodes i.e., Cisco Unified Communications Manager and Microsoft LYNC 2010 Server.

Cisco Unified Communications Manager- Session Manager Edition – Software release

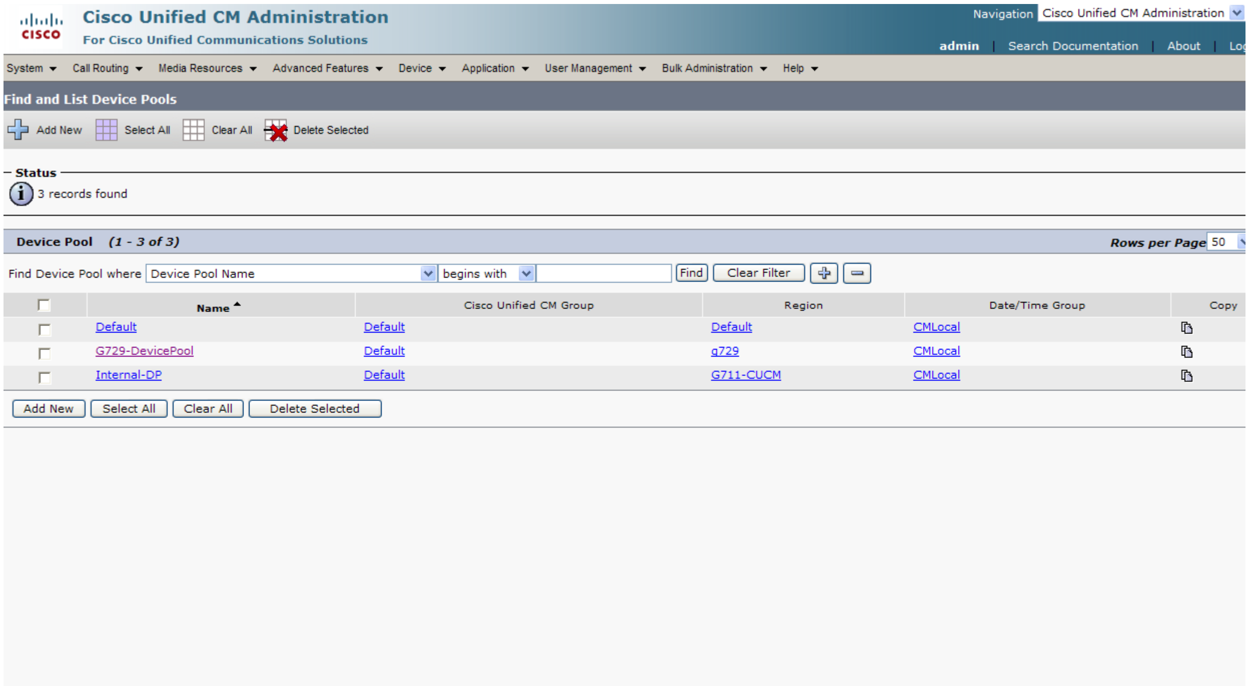


Cisco Unified Communications Manager-Session Manager Edition-Region Configurations

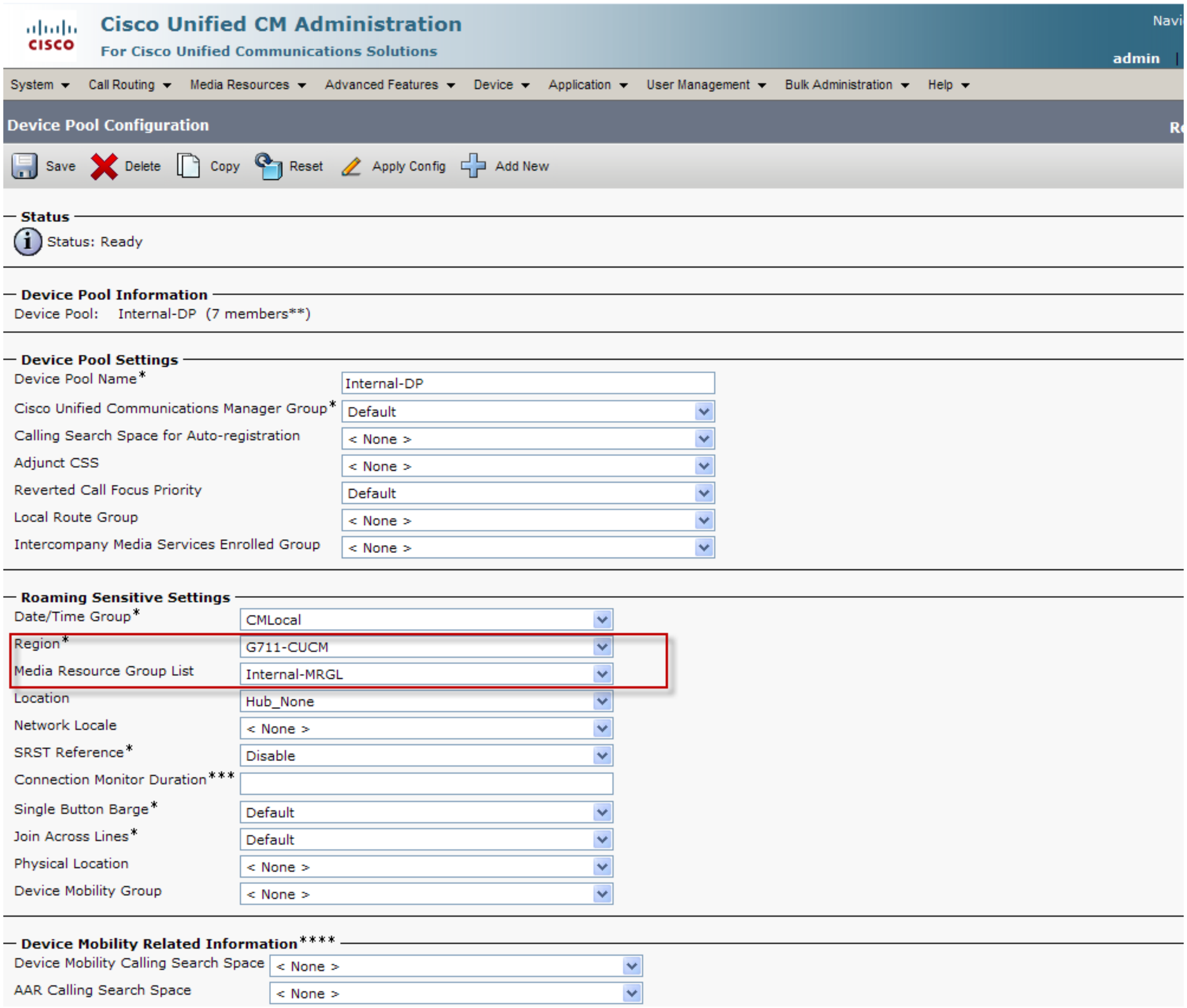


Region configurations are used to determine the codecs used by the endpoint within the region and across the region.

Cisco Unified Communications Manager – Session Manager Edition – Device Pools

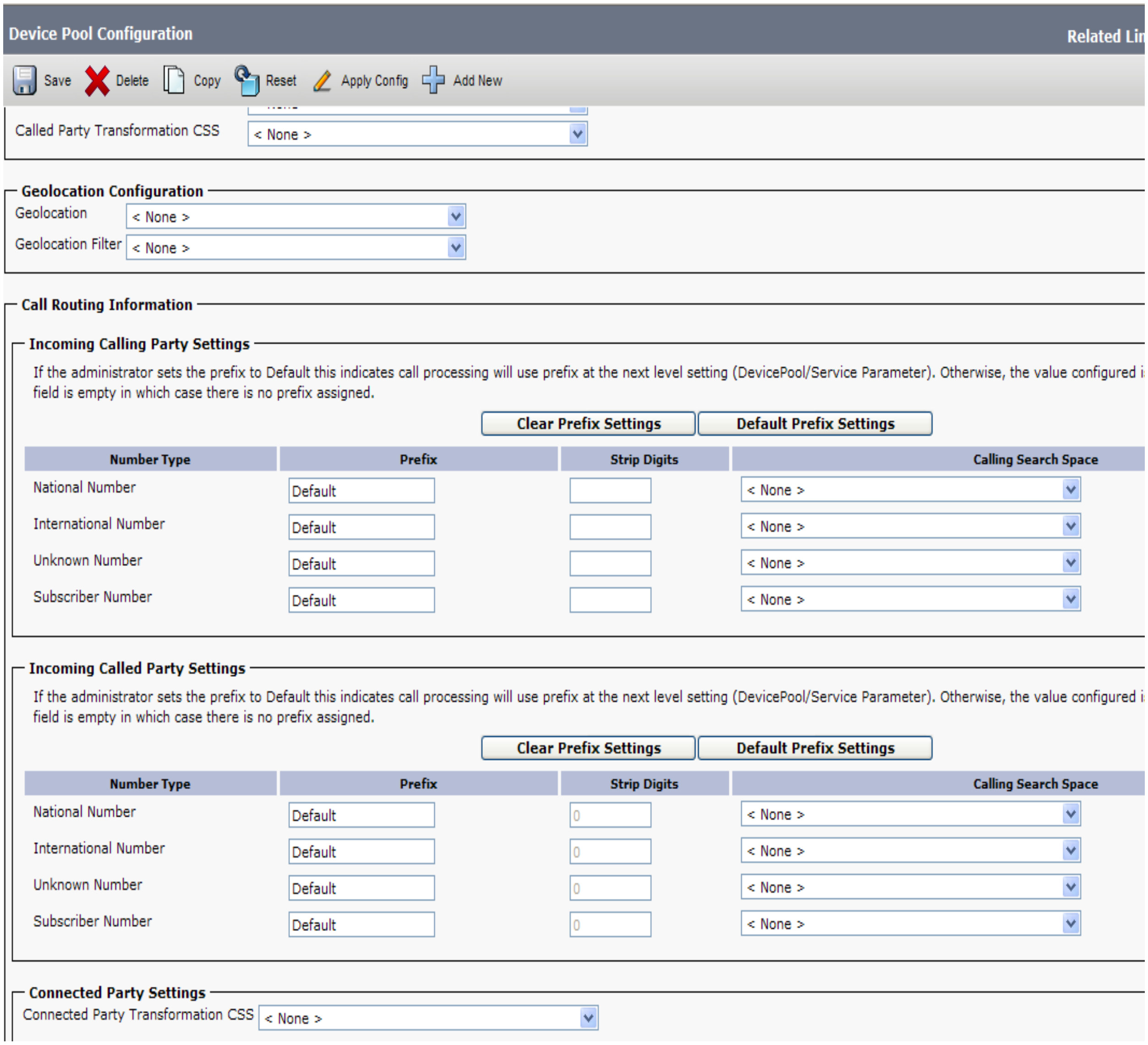


Cisco Unified Communications Manager – Session Manager Edition – Device Pool Configurations G711

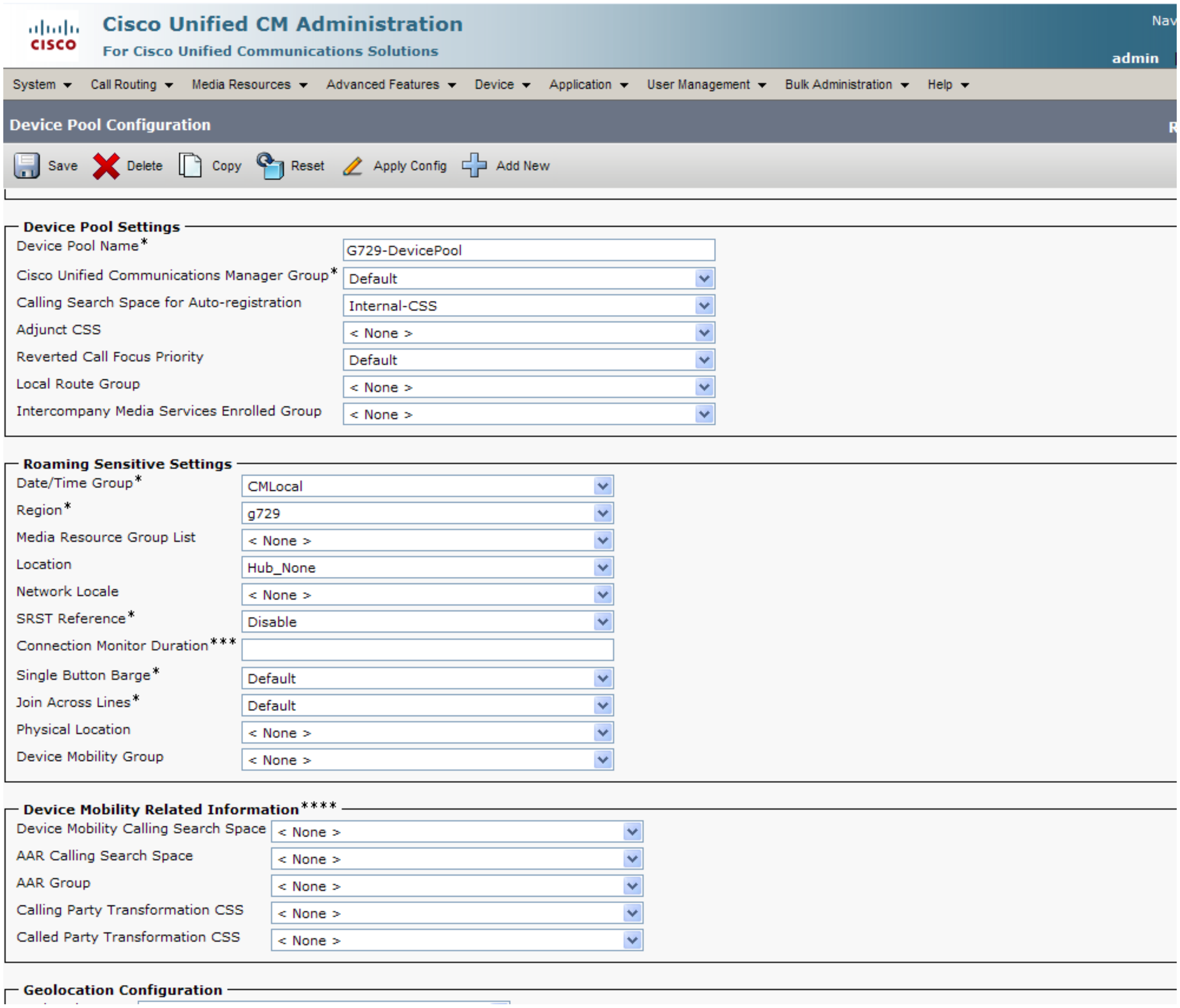


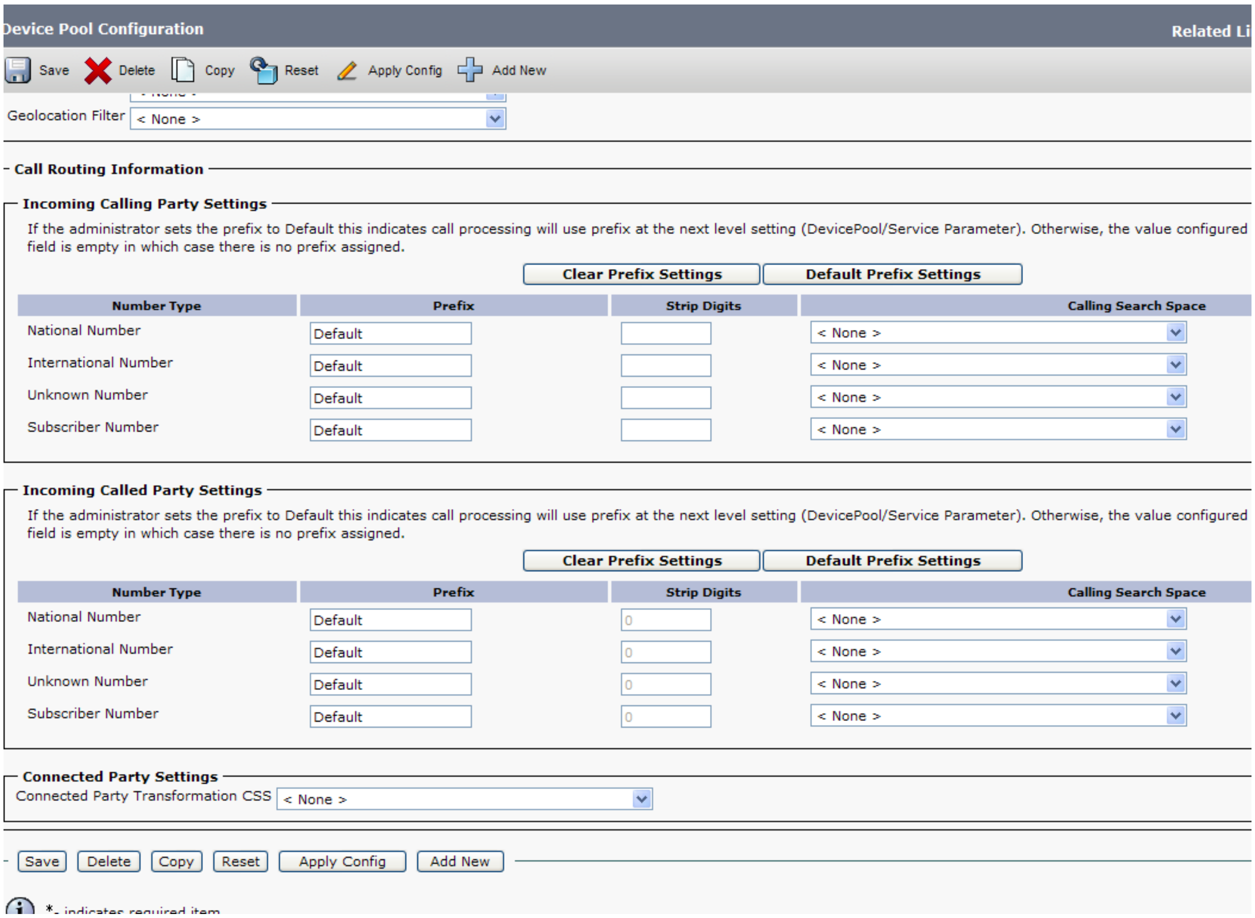
Region is associated with a device pool which is assigned to the endpoint or a trunk.

Cisco Unified Communications Manager – Session Manager Edition – Device Pool Configurations G711

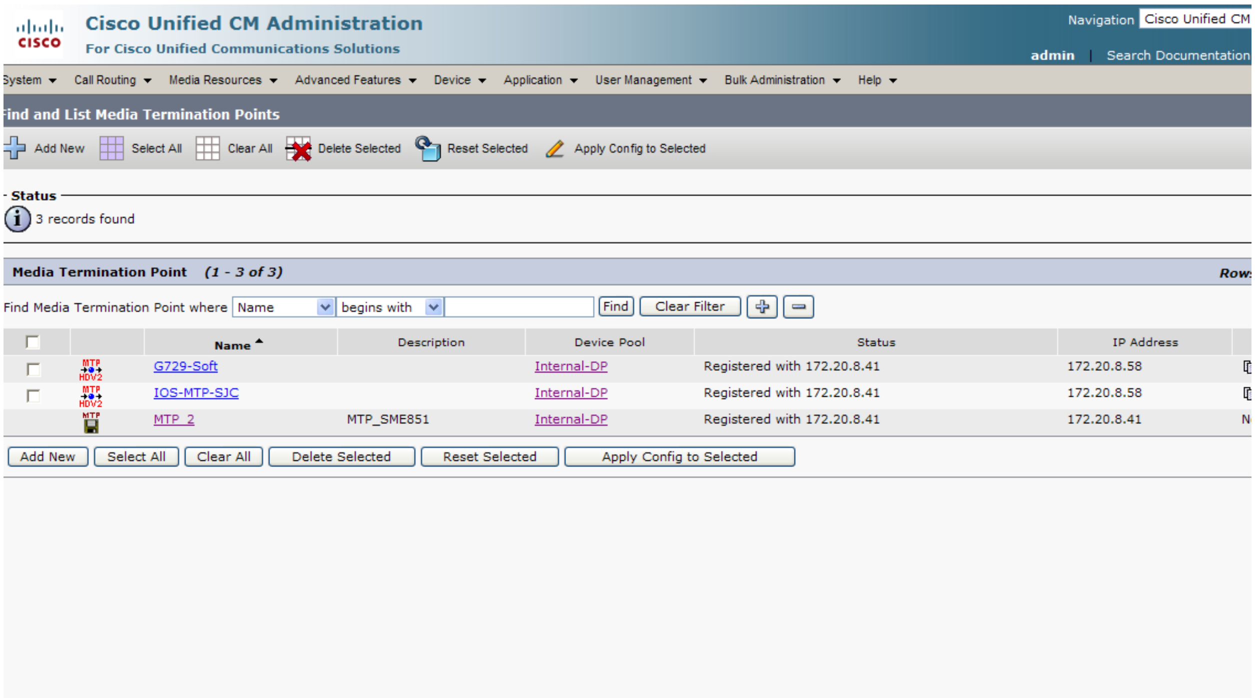


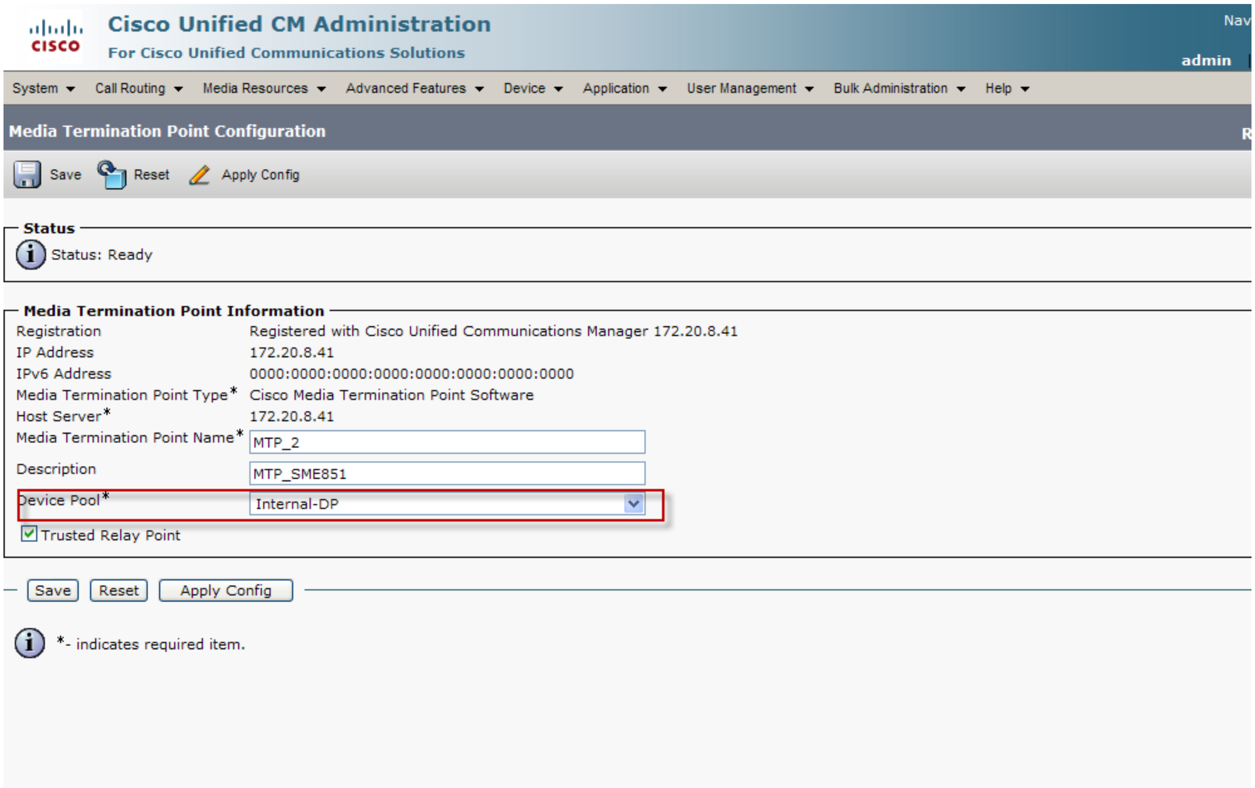
Cisco Unified Communications Manager – Session Manager Edition – Device Pool Configurations-G729



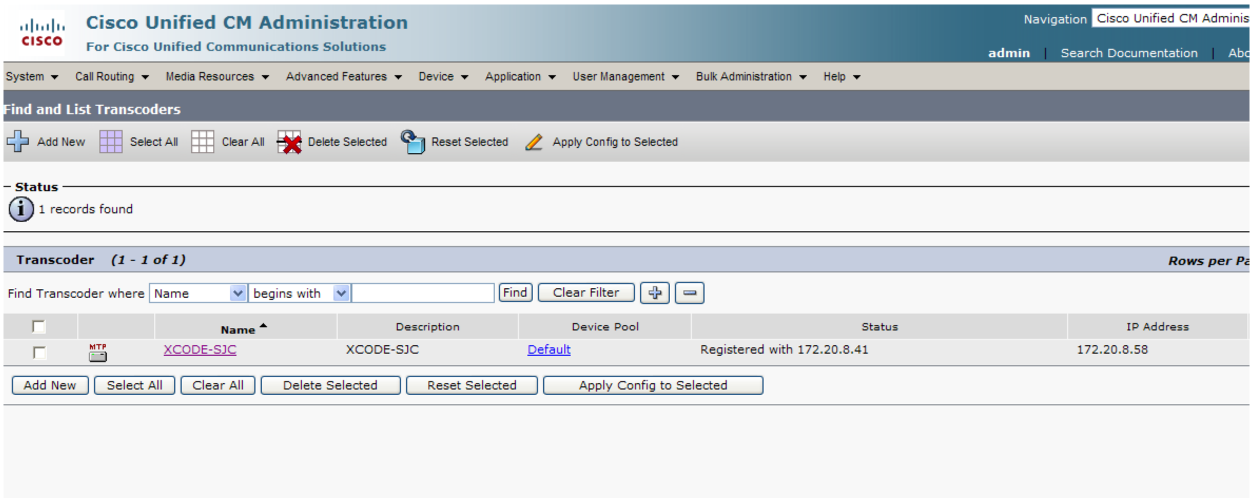

Device Pool Configuration
Related Li
Save
Delete
Copy
Reset
Apply Config
Add New
Geolocation Filter
< None >
Call Routing Information
Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured
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
International Number
Unknown Number
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
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
International Number
Unknown Number
Subscriber Number
Default
0
< None >
Connected Party Settings
Connected Party Transformation CSS
< None >
Save
Delete
Copy
Reset
Apply Config
Add New

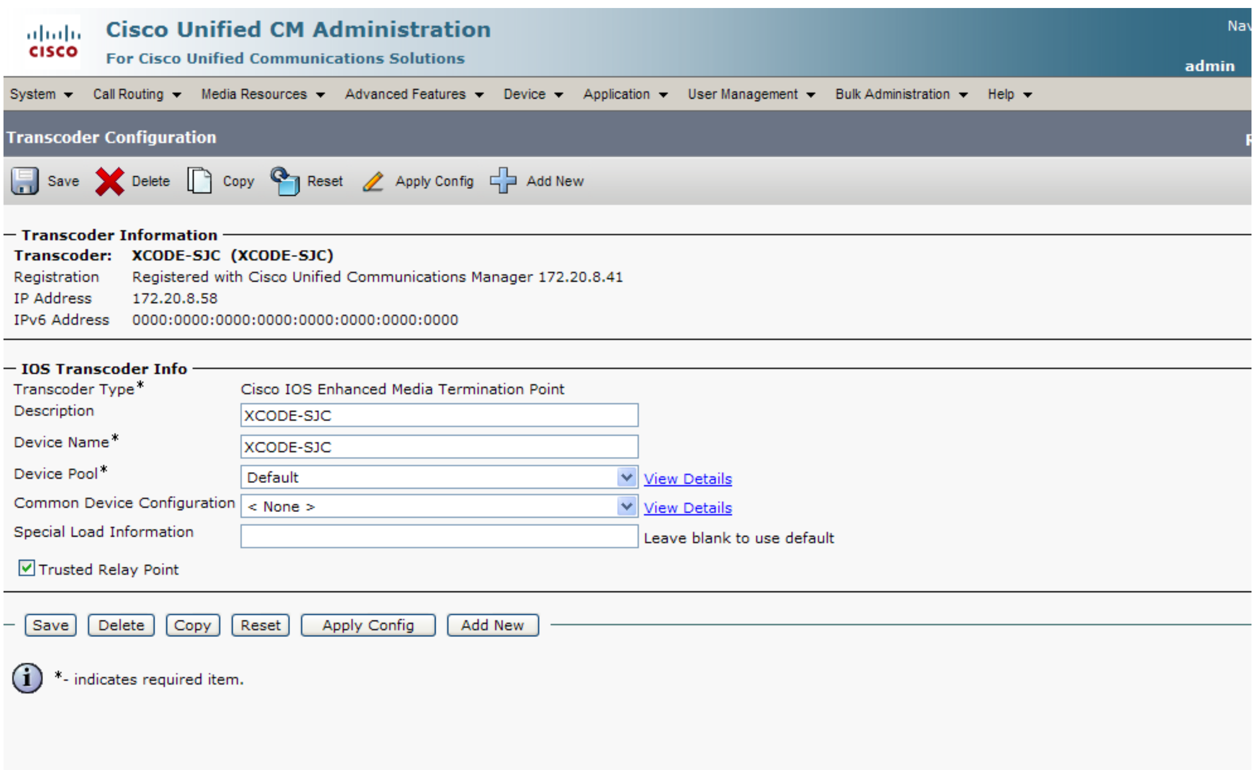
Cisco Unified Communications Manager – Session manager Edition - Media Termination Points Configurations



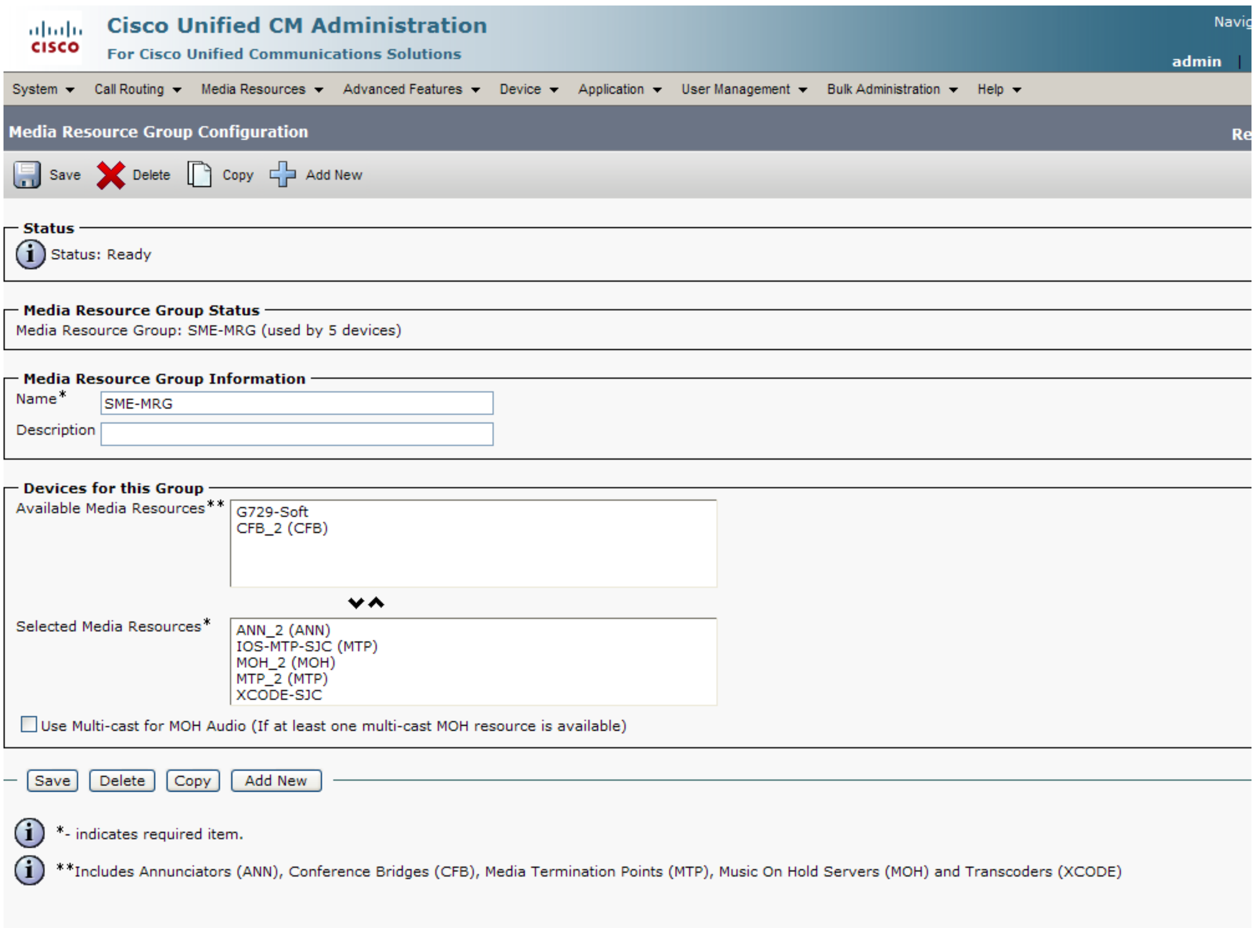

Cisco Unified CM Administration
For Cisco Unified Communications Solutions
admin
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Media Termination Point Configuration
Save Reset Apply Config
Status
Status: Ready
Media Termination Point Information
Registration Registered with Cisco Unified Communications Manager 172.20.8.41
IP Address 172.20.8.41
IPv6 Address 0000:0000:0000:0000:0000:0000:0000:0000
Media Termination Point Type* Cisco Media Termination Point Software
Host Server* 172.20.8.41
Media Termination Point Name* MTP_2
Description MTP_SME851
Device Pool* Internal-DP
Trusted Relay Point
Save Reset Apply Config
*- indicates required item.

Cisco Unified Communications Manager – Session manager Edition – Transcoder Configuration

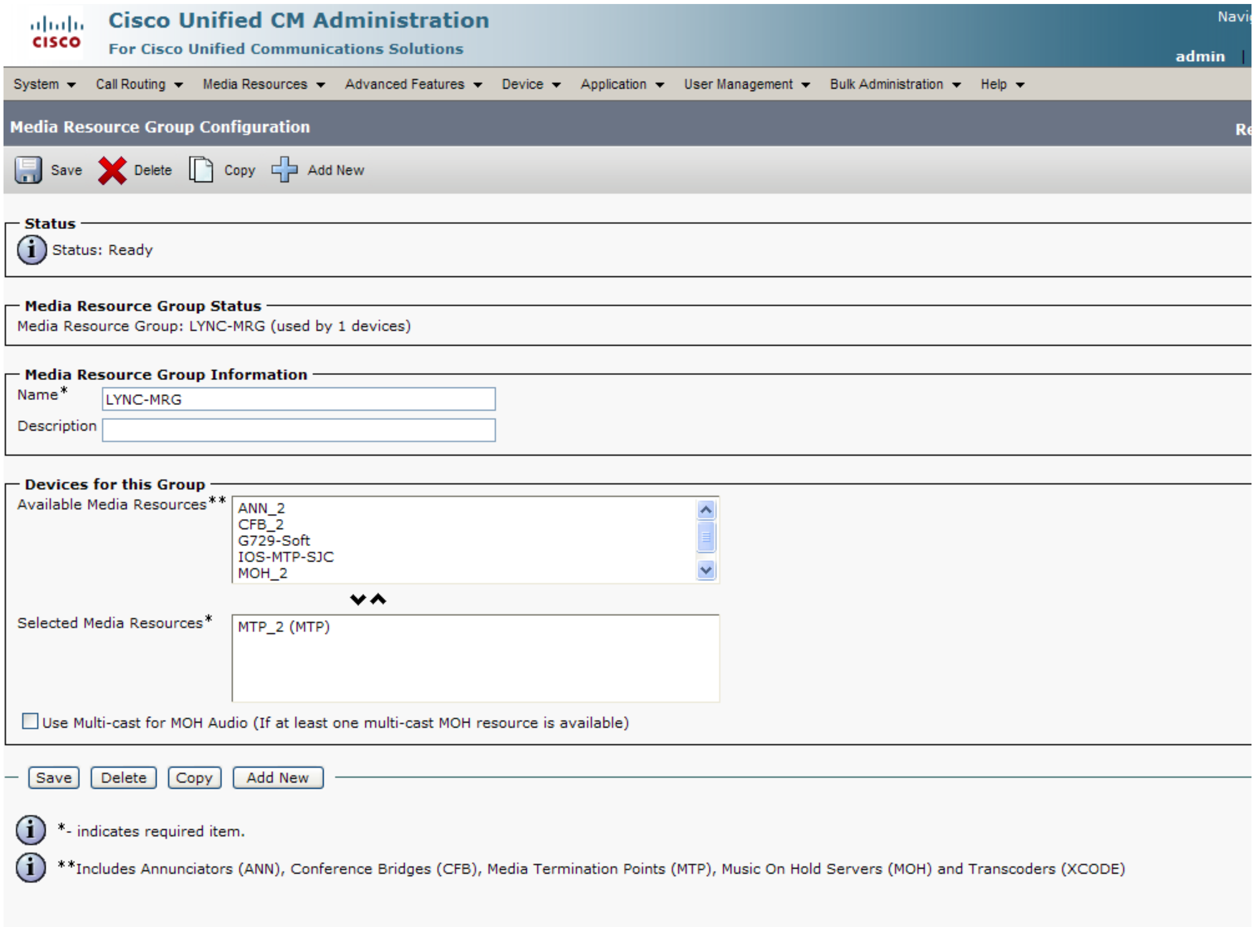




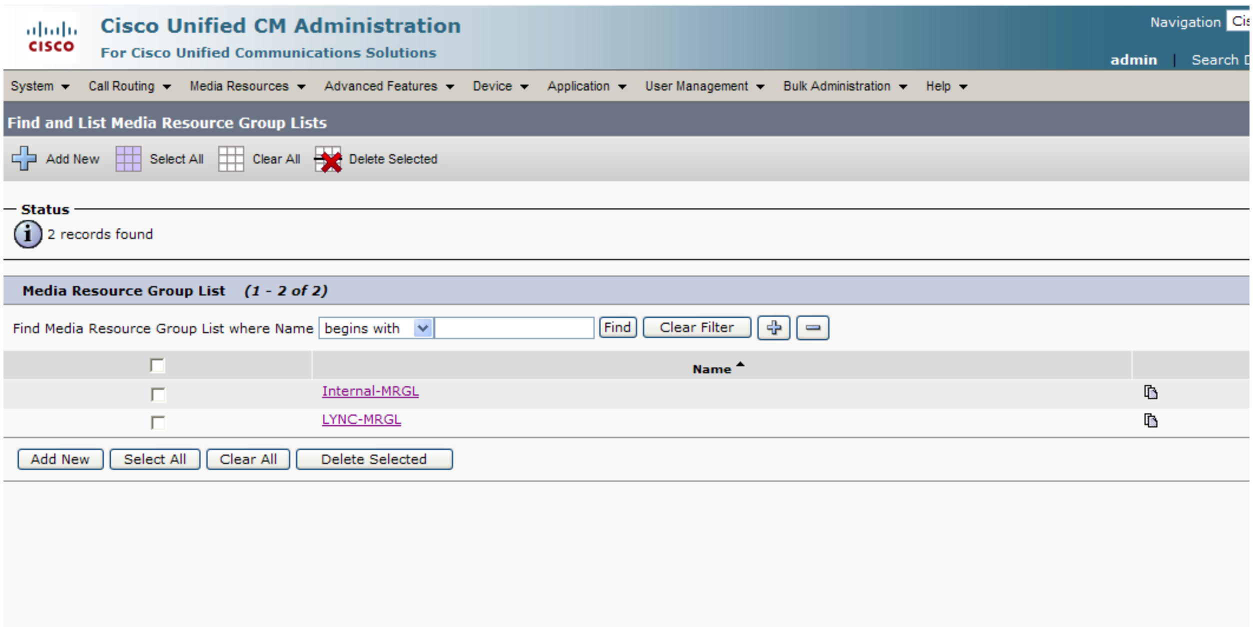
Cisco Unified Communications Manager – Session Manager Edition – Media Resource Groups configurations



MRG for LYNC MRGL

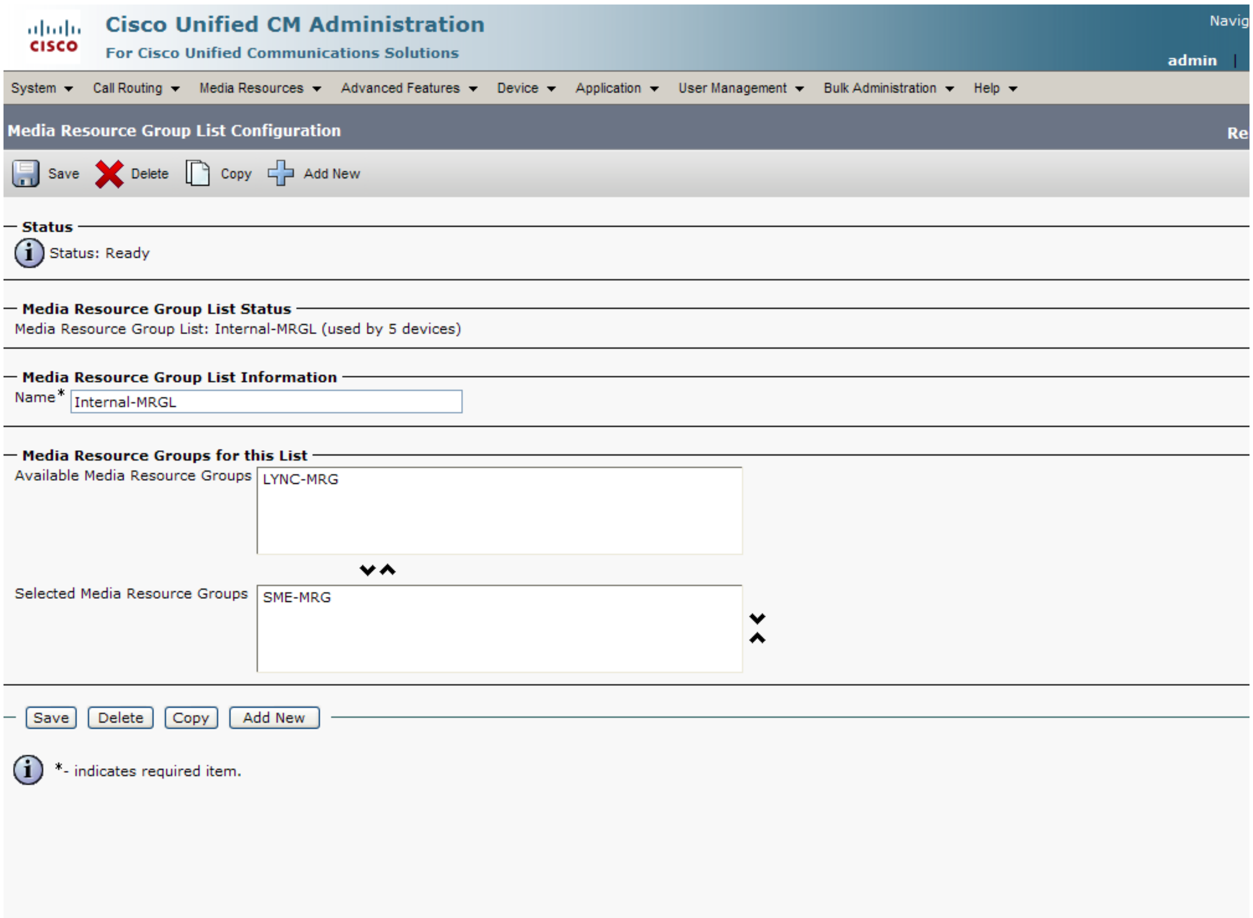


Cisco Unified Communications Manager – Session Manager Edition – Media Resource Group Lists configurations

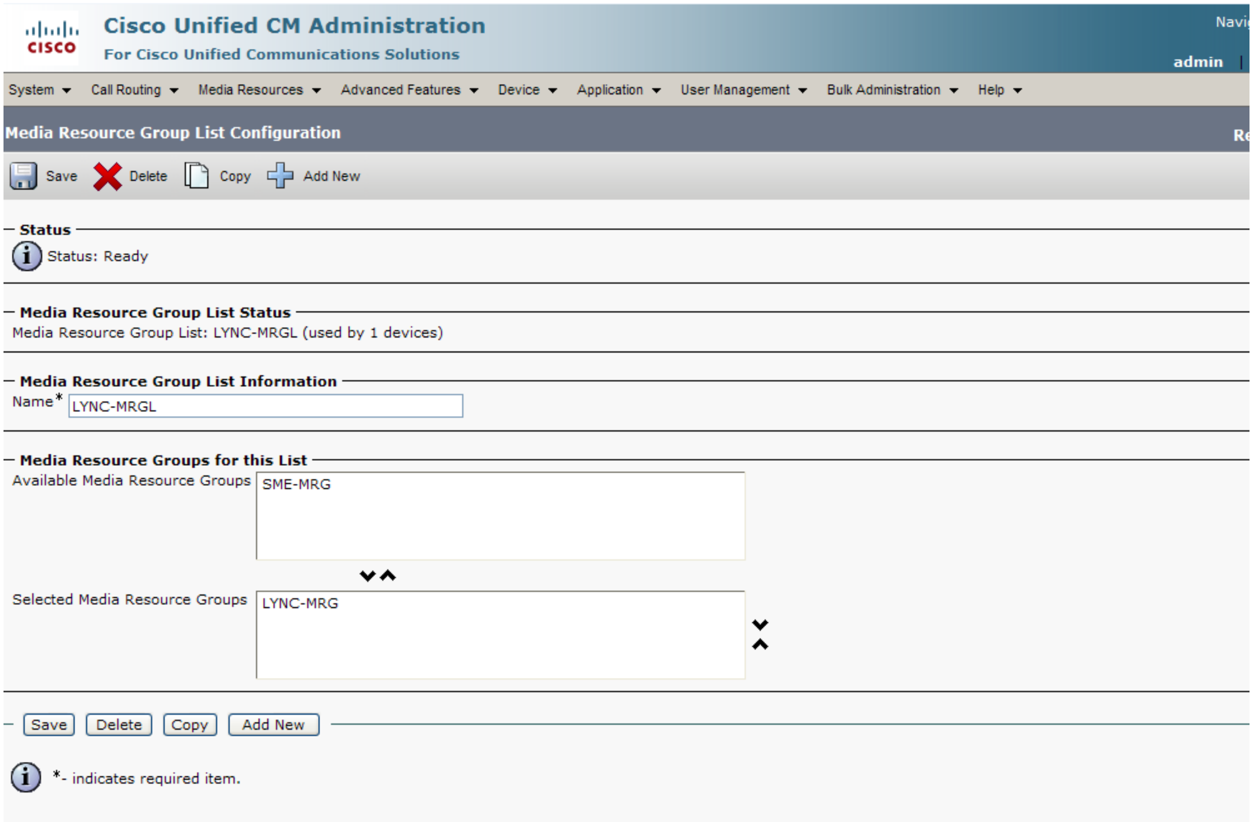


LYNC will need a separate MRGL to force MTP based early offer from the SME

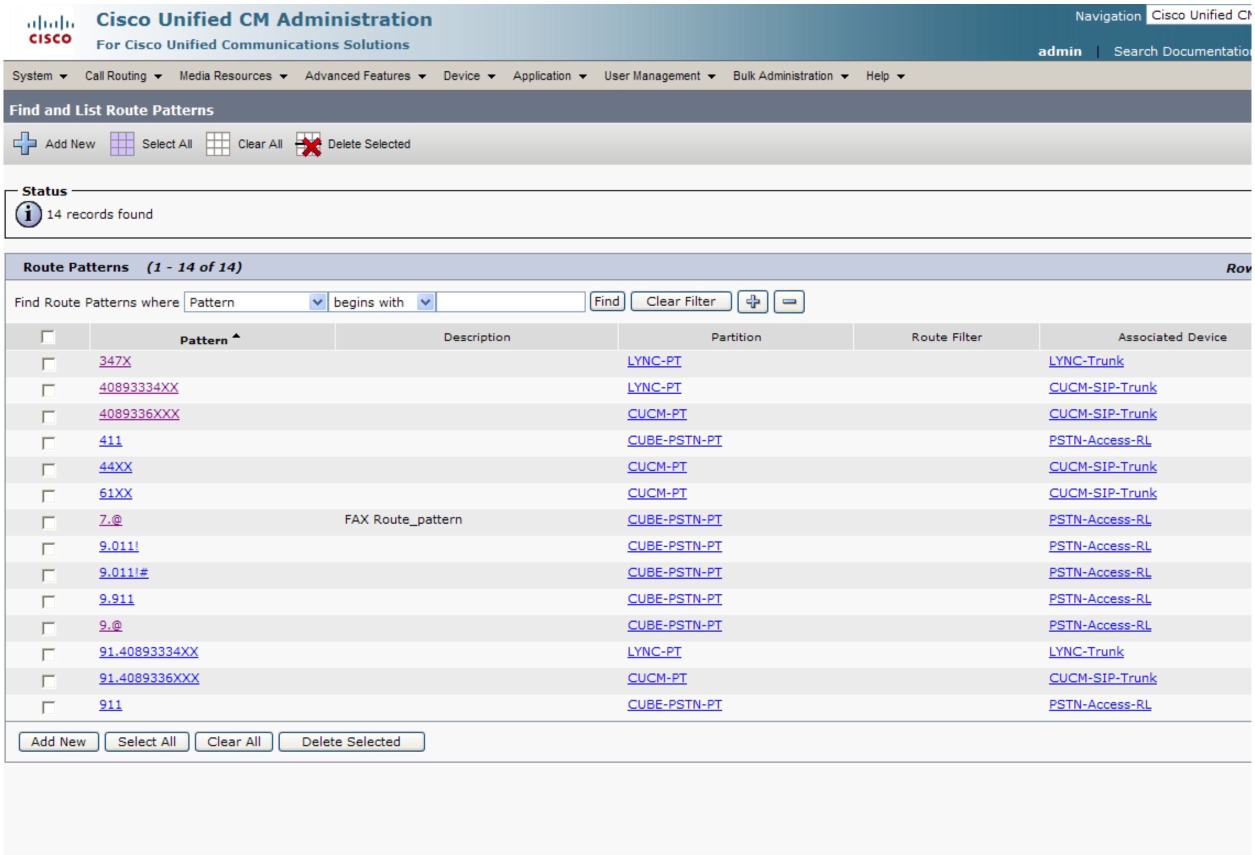
MRGL to be associated with SIP trunks except LYNC trunk



MRGL to be associated with SIP trunk connecting Microsoft LYNC 2010 server

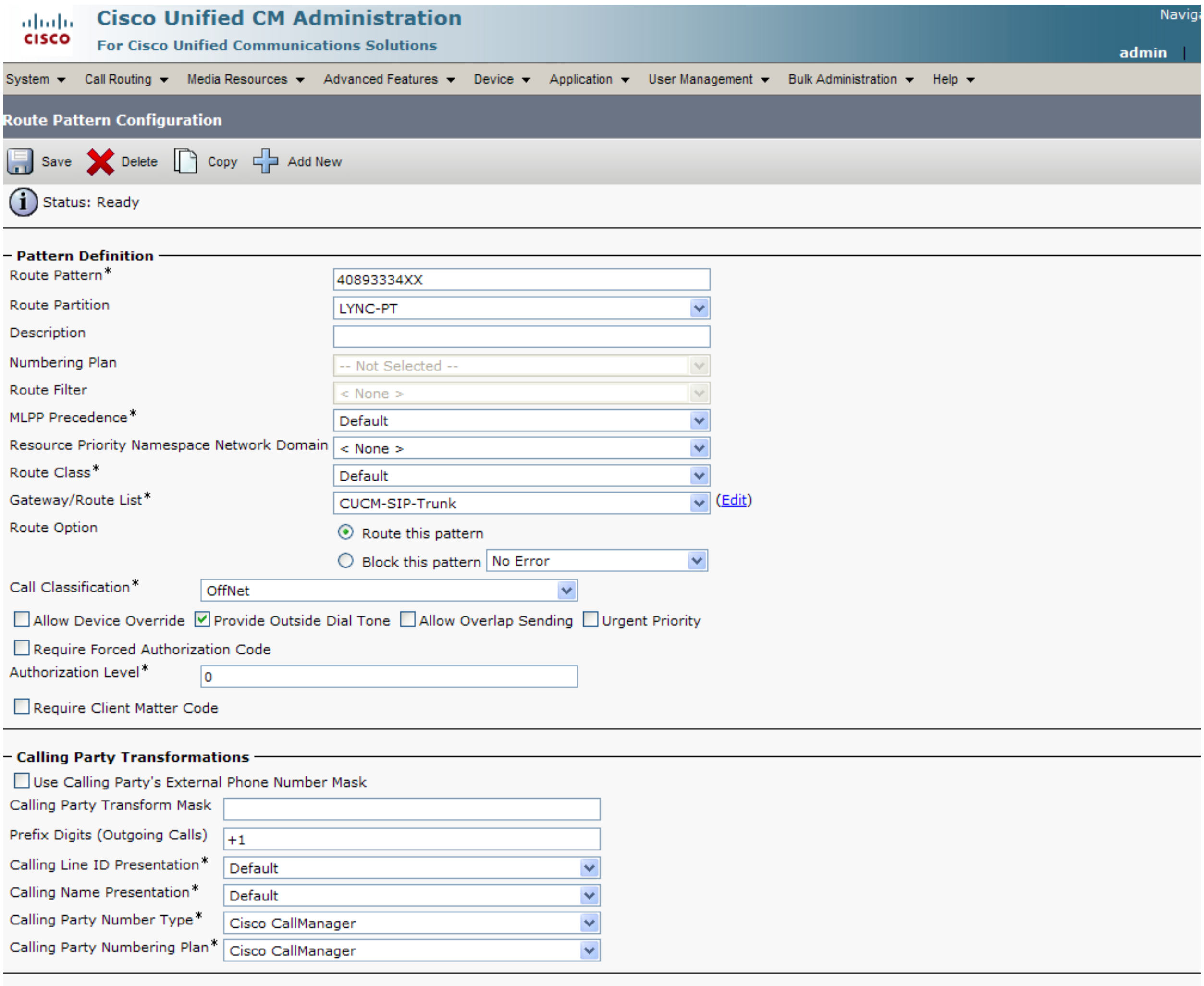


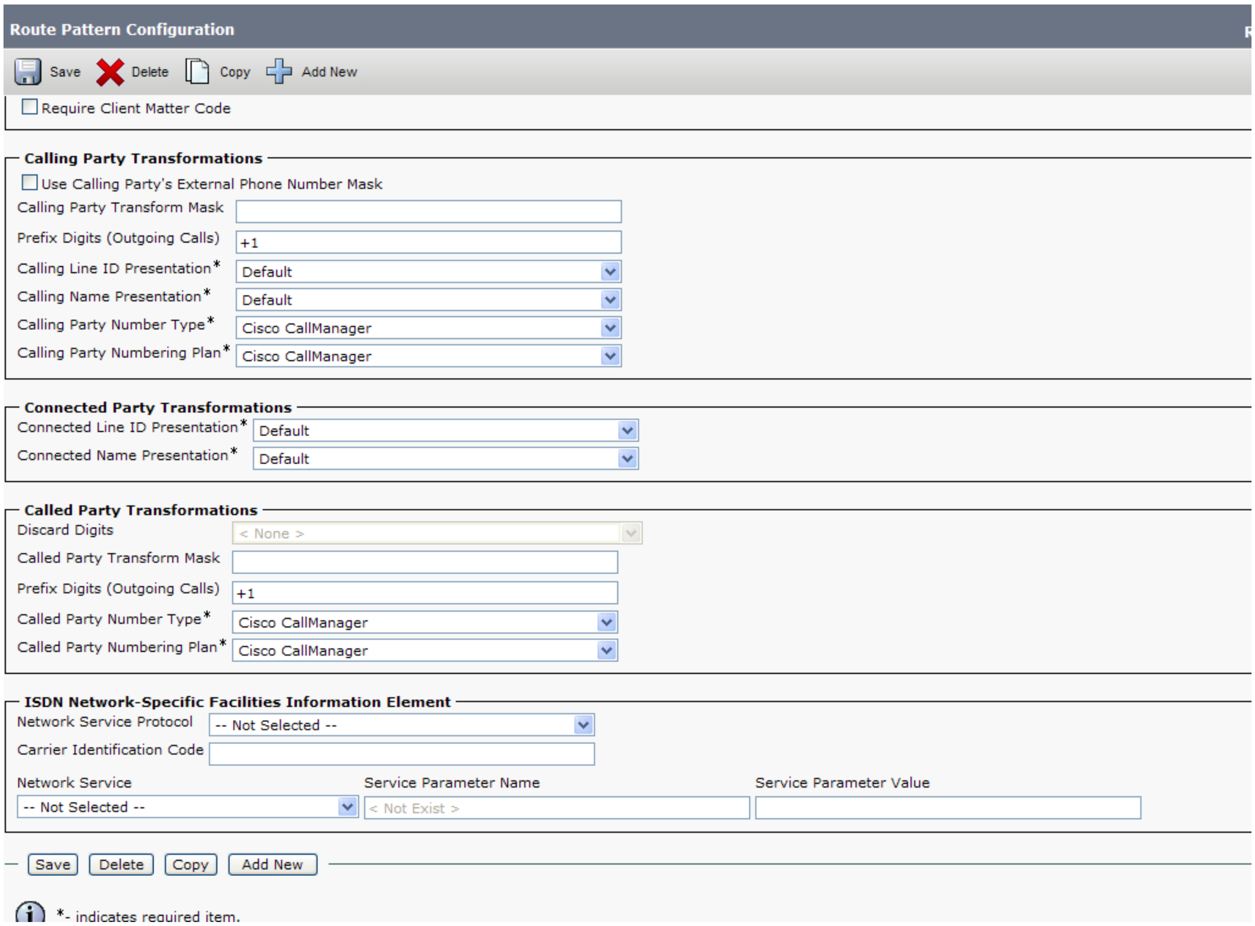
Cisco Unified Communications Manager –Session Manager Edition – Route Patterns



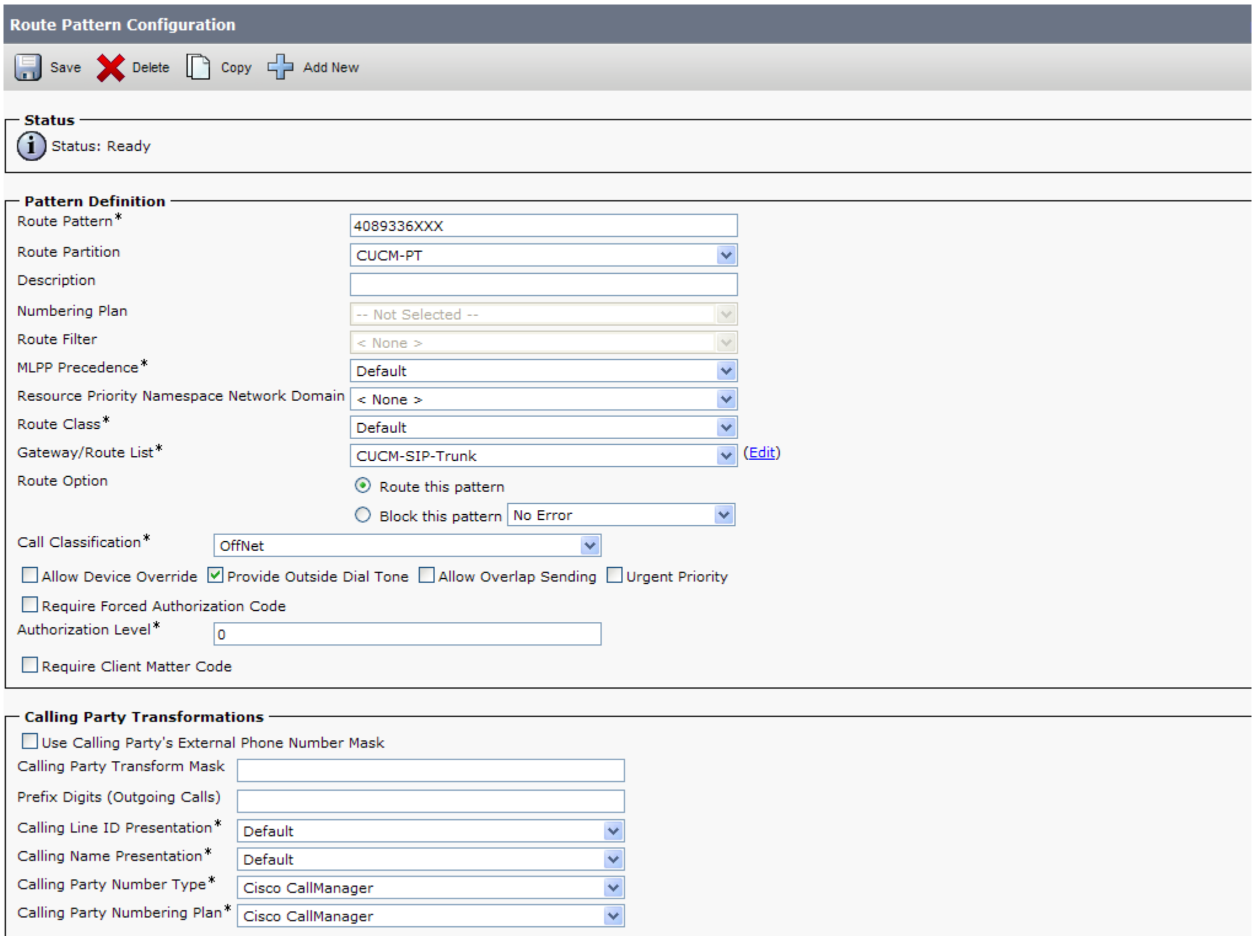
	Pattern	Description	Partition	Route Filter	Associated Device
	347X		LYNC-PT		LYNC-Trunk
	40893334XX		LYNC-PT		CUCM-SIP-Trunk
	4089336XXX		CUCM-PT		CUCM-SIP-Trunk
	411		CUBE-PSTN-PT		PSTN-Access-RL
	44XX		CUCM-PT		CUCM-SIP-Trunk
	61XX		CUCM-PT		CUCM-SIP-Trunk
	7.@	FAX Route_pattern	CUBE-PSTN-PT		PSTN-Access-RL
	9.011!		CUBE-PSTN-PT		PSTN-Access-RL
	9.011!#		CUBE-PSTN-PT		PSTN-Access-RL
	9.911		CUBE-PSTN-PT		PSTN-Access-RL
	9.@		CUBE-PSTN-PT		PSTN-Access-RL
	91.40893334XX		LYNC-PT		LYNC-Trunk
	91.4089336XXX		CUCM-PT		CUCM-SIP-Trunk
	911		CUBE-PSTN-PT		PSTN-Access-RL

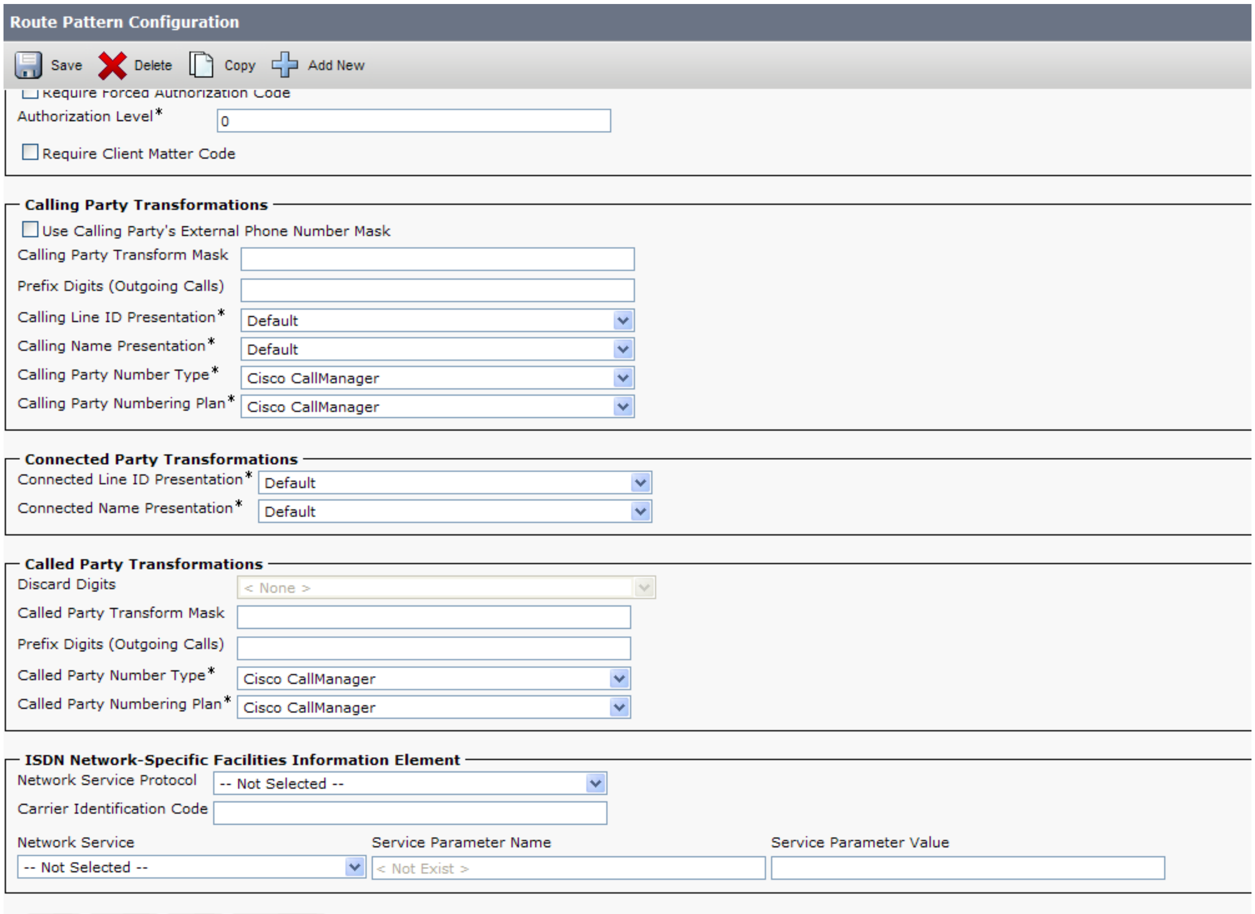
Route Pattern to the Microsoft LYNC 2010 server



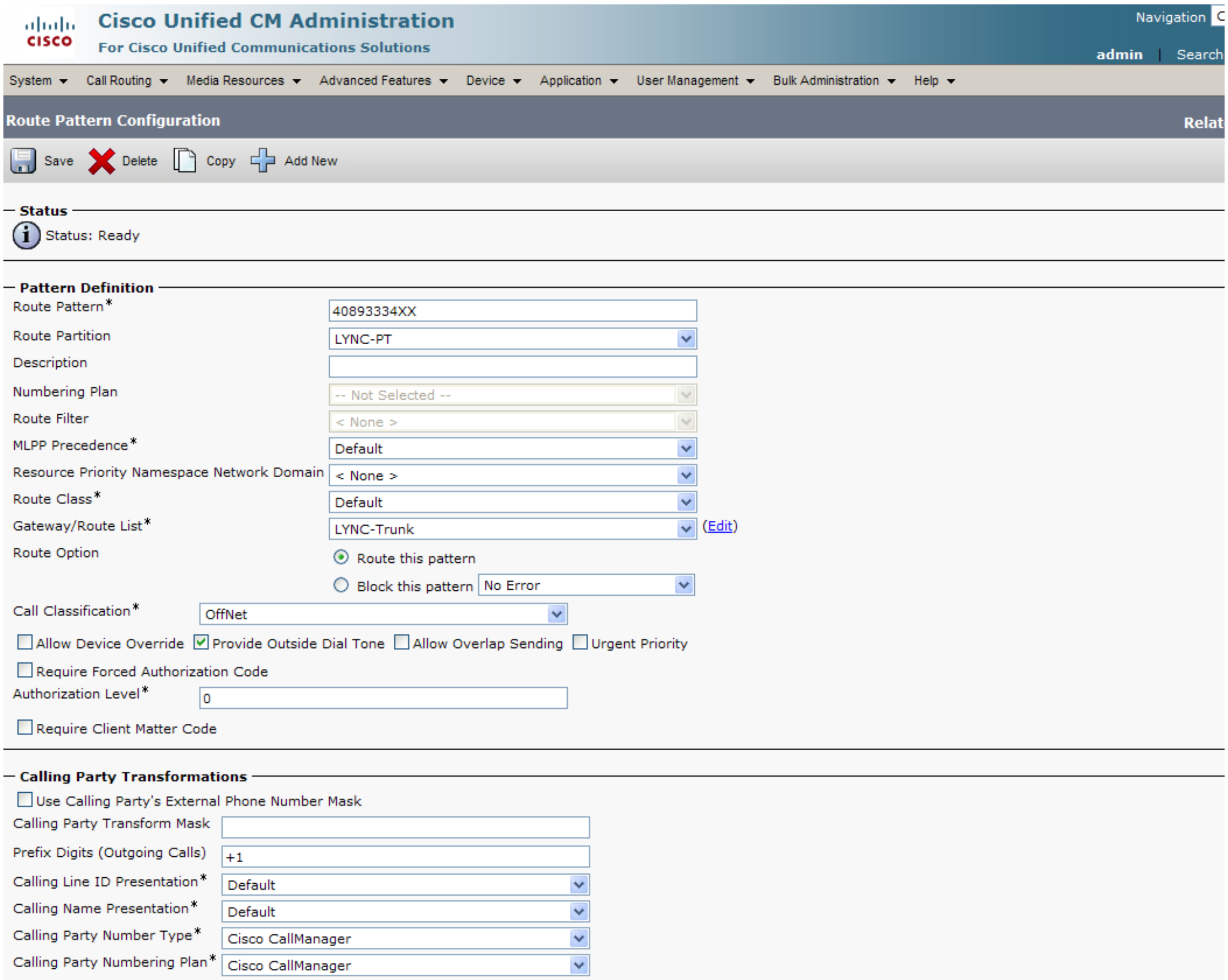

Route Pattern Configuration
Save
Delete
Copy
Add New
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
+1
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling Party Number Type*
Cisco CallManager
Calling Party Numbering Plan*
Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Called Party Transformations
Discard Digits
< None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
+1
Called Party Number Type*
Cisco CallManager
Called Party Numbering Plan*
Cisco CallManager
ISDN Network-Specific Facilities Information Element
Network Service Protocol
-- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save
Delete
Copy
Add New
*- indicates required item.

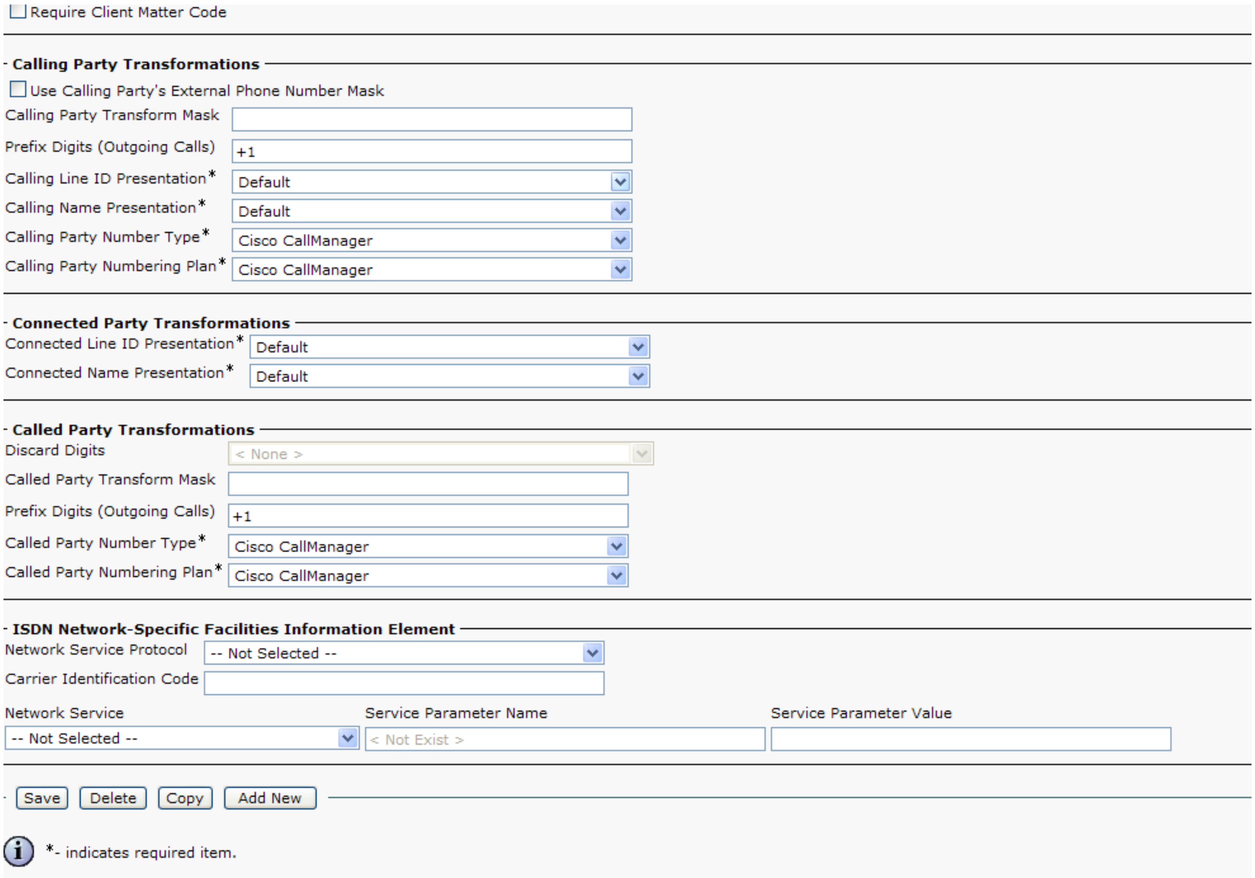
Route Pattern to the Cisco Unified Communications Manager 8.5 (Leaf node)




Route Pattern Configuration
Save
Delete
Copy
Add New
Authorization Level*
0
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling Party Number Type*
Cisco CallManager
Calling Party Numbering Plan*
Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Called Party Transformations
Discard Digits
< None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Cisco CallManager
Called Party Numbering Plan*
Cisco CallManager
ISDN Network-Specific Facilities Information Element
Network Service Protocol
-- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >

Route Pattern to route calls to the LYNC server





☐ Require Client Matter Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	+1
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Called Party Transformations

Discard Digits	< None >
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	+1
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

ISDN Network-Specific Facilities Information Element

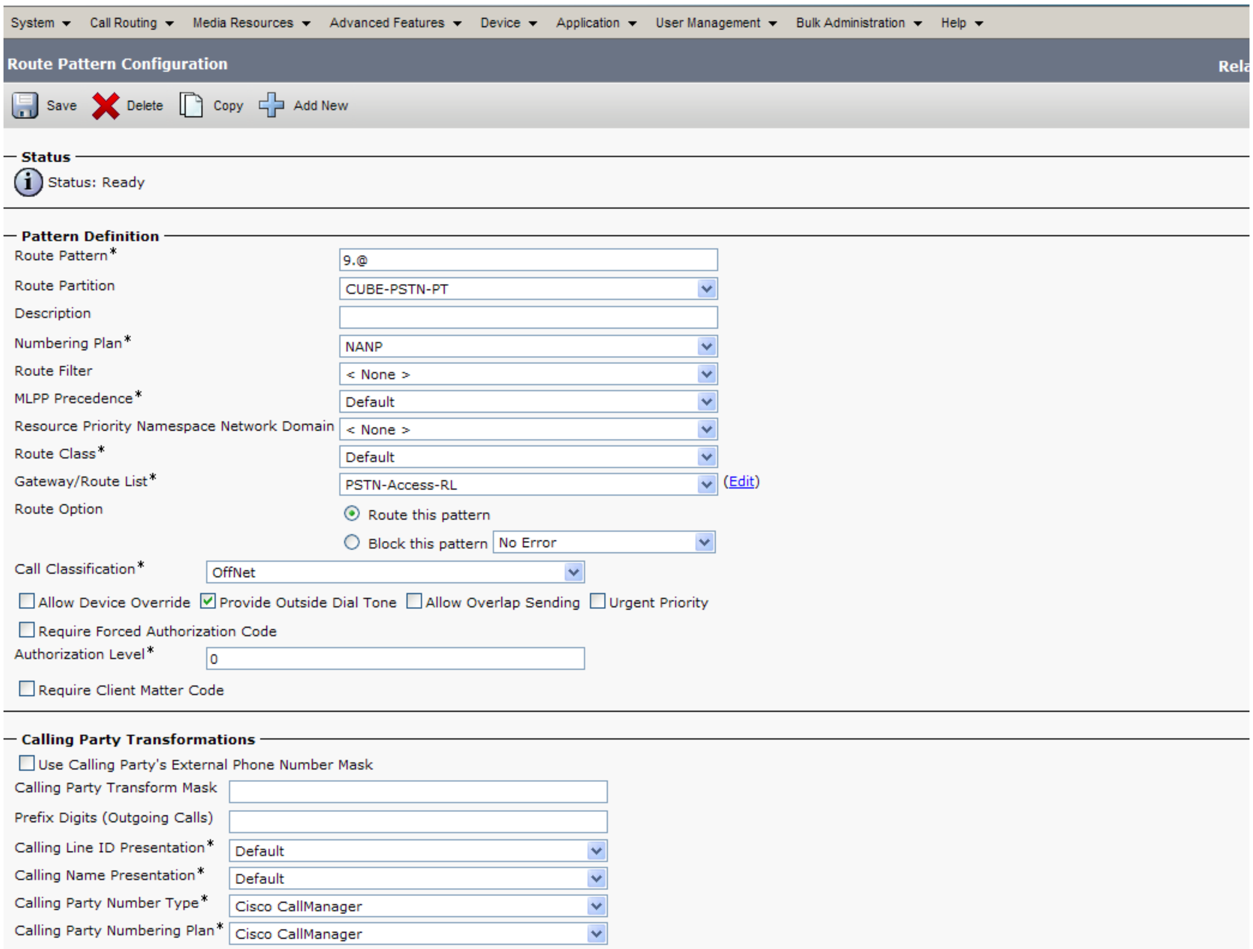
Network Service Protocol	-- Not Selected --
Carrier Identification Code	

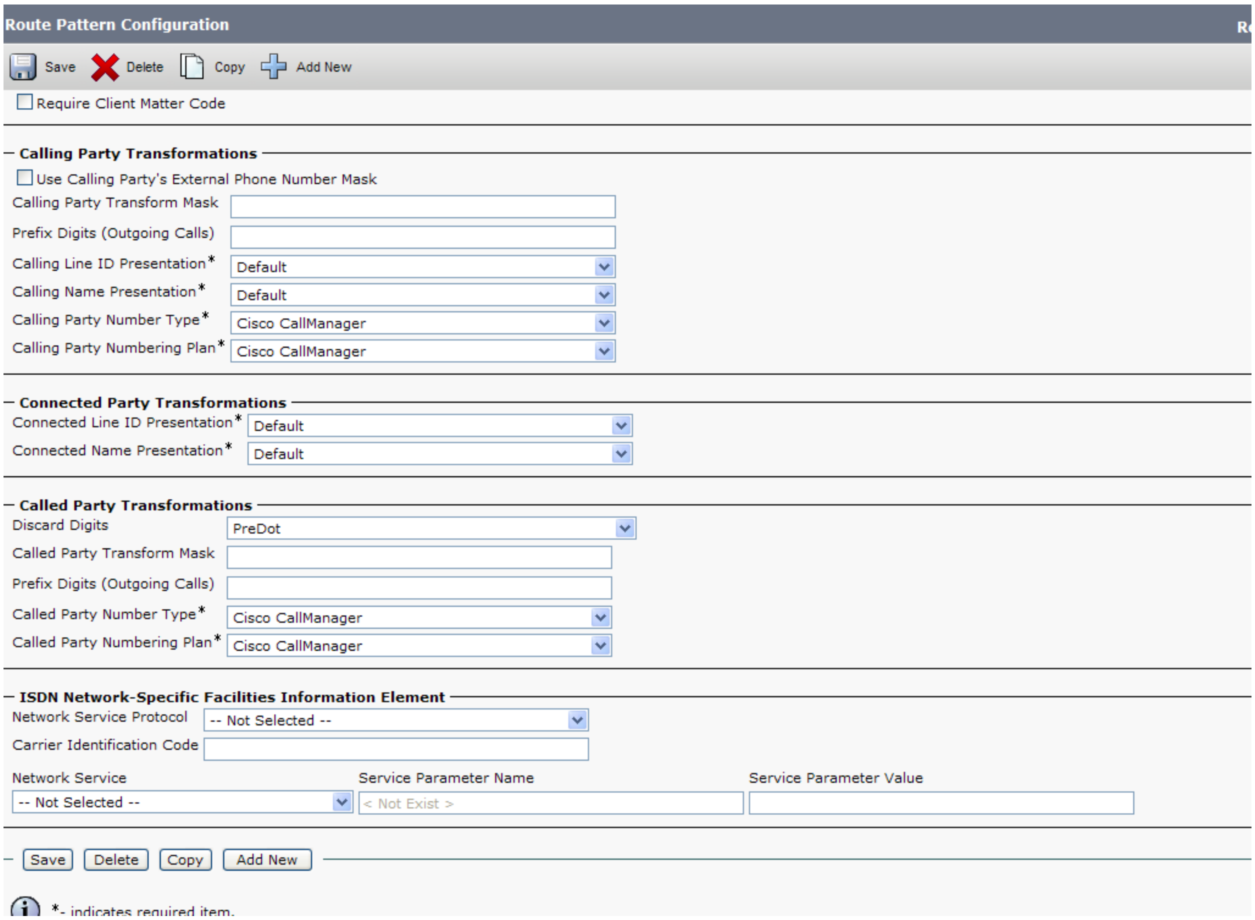
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Save | Delete | Copy | Add New

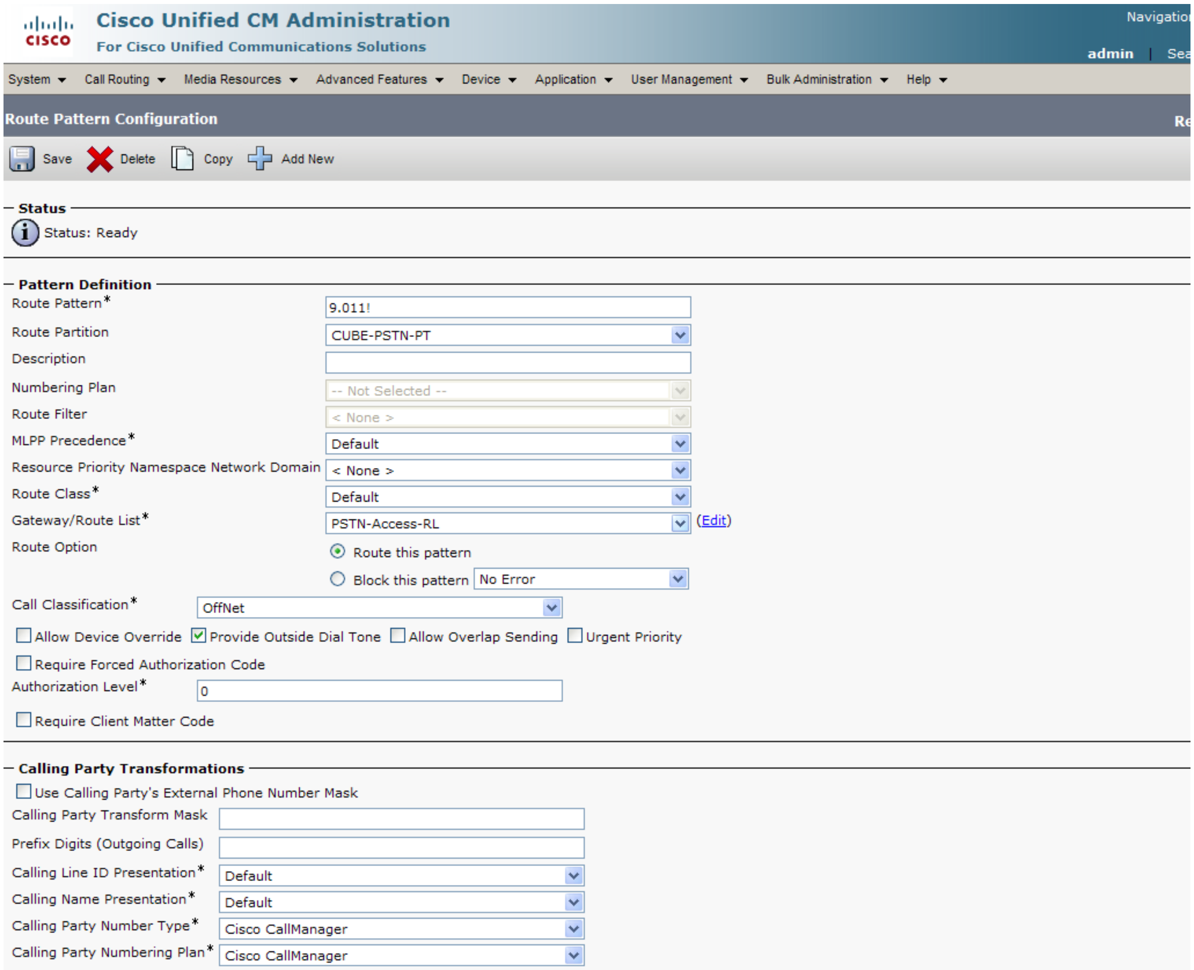
*- indicates required item.

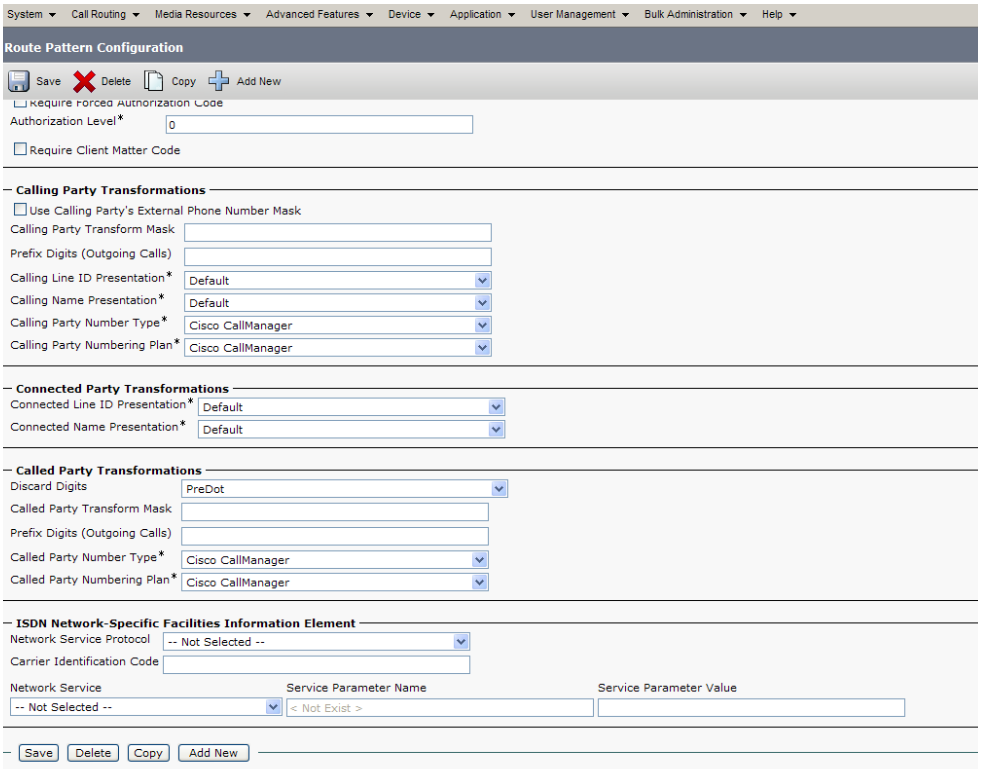
Route Pattern to Service Provider



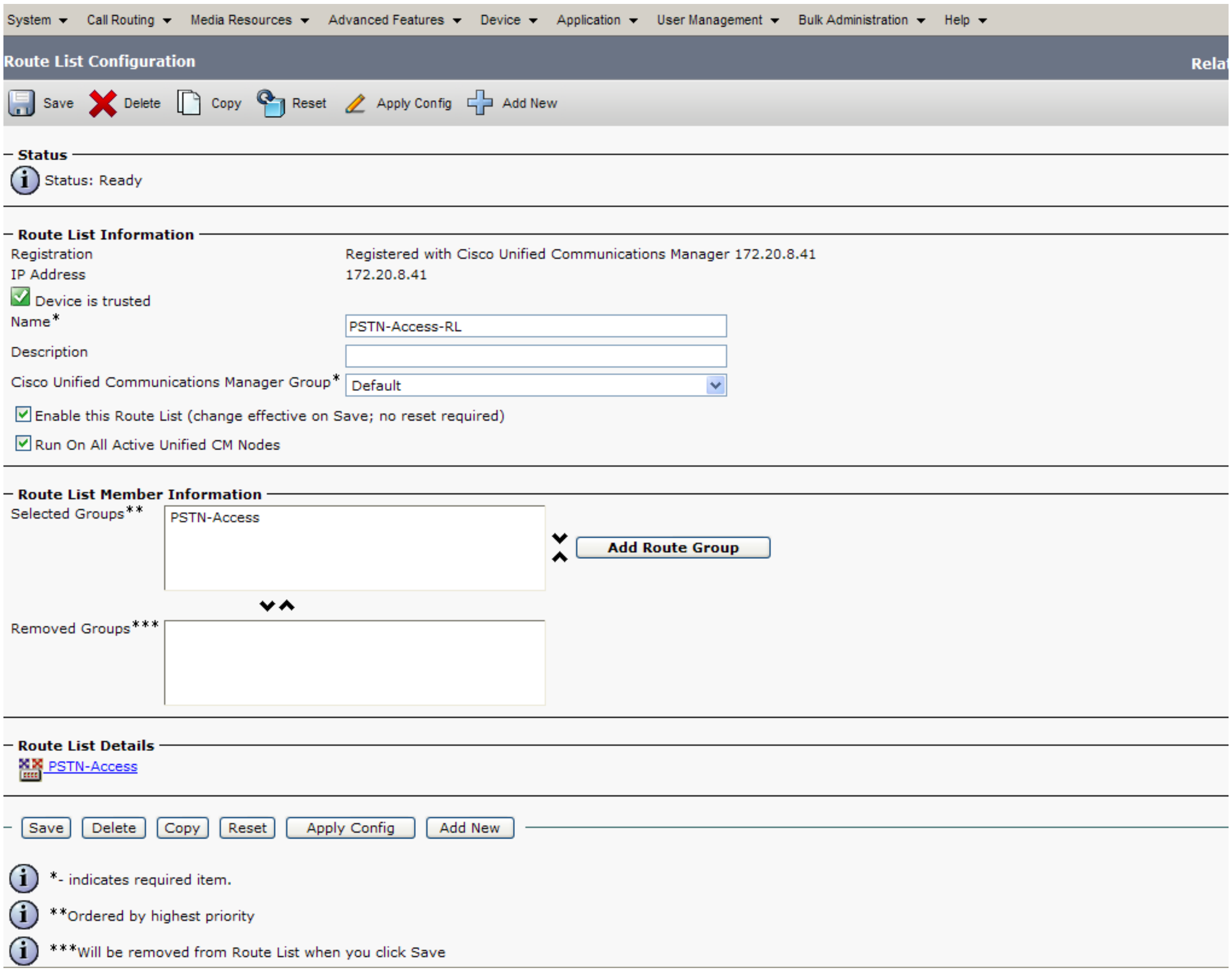

Route Pattern Configuration
Save
Delete
Copy
Add New
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling Party Number Type*
Cisco CallManager
Calling Party Numbering Plan*
Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Called Party Transformations
Discard Digits
PreDot
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Cisco CallManager
Called Party Numbering Plan*
Cisco CallManager
ISDN Network-Specific Facilities Information Element
Network Service Protocol
-- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save
Delete
Copy
Add New
*- indicates required item.

International Route Pattern

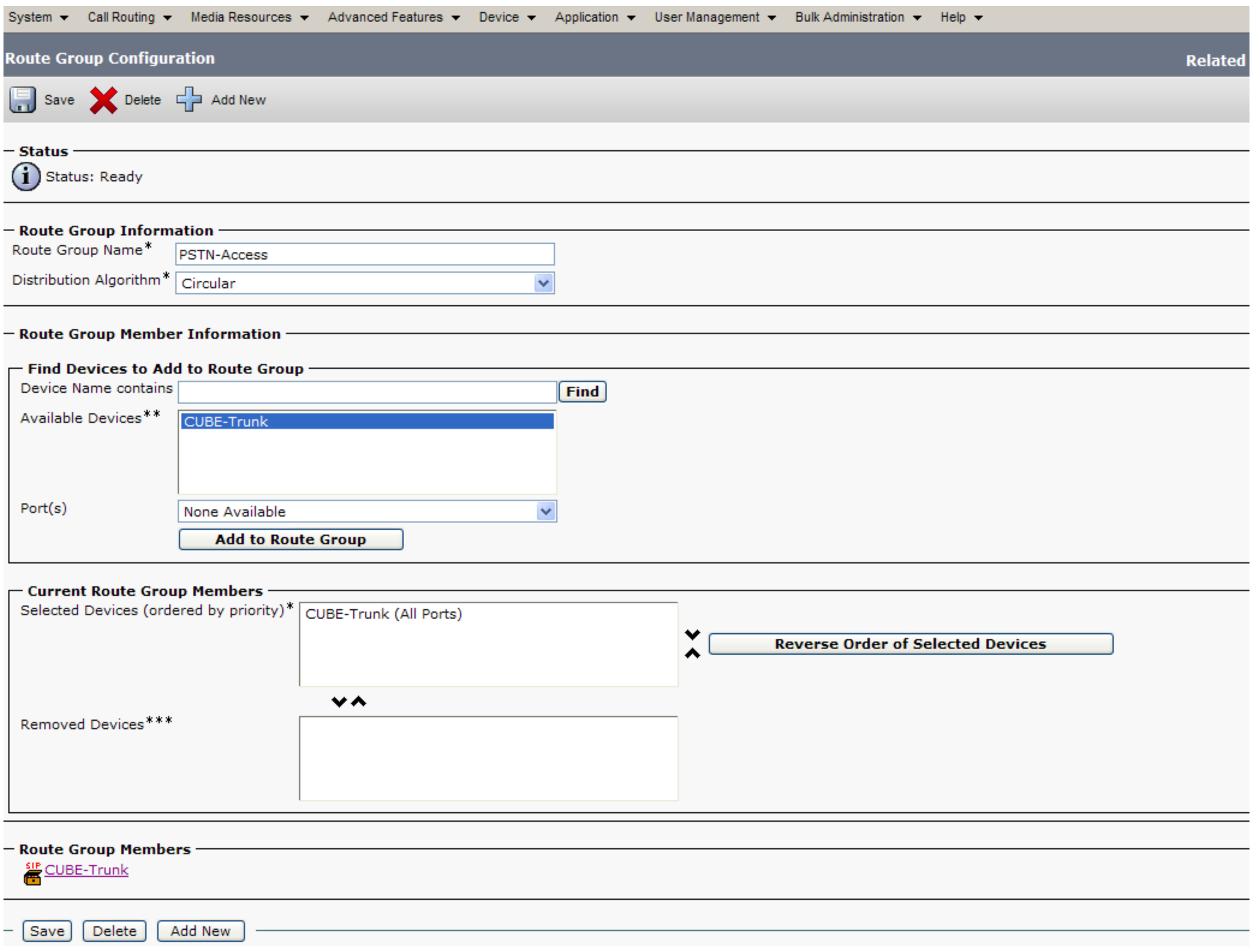



System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Route Pattern Configuration
Save
Delete
Copy
Add New
Require Forced Authorization Code
Authorization Level*
0
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling Party Number Type*
Cisco CallManager
Calling Party Numbering Plan*
Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Called Party Transformations
Discard Digits
PreDot
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Cisco CallManager
Called Party Numbering Plan*
Cisco CallManager
ISDN Network-Specific Facilities Information Element
Network Service Protocol
-- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save
Delete
Copy
Add New

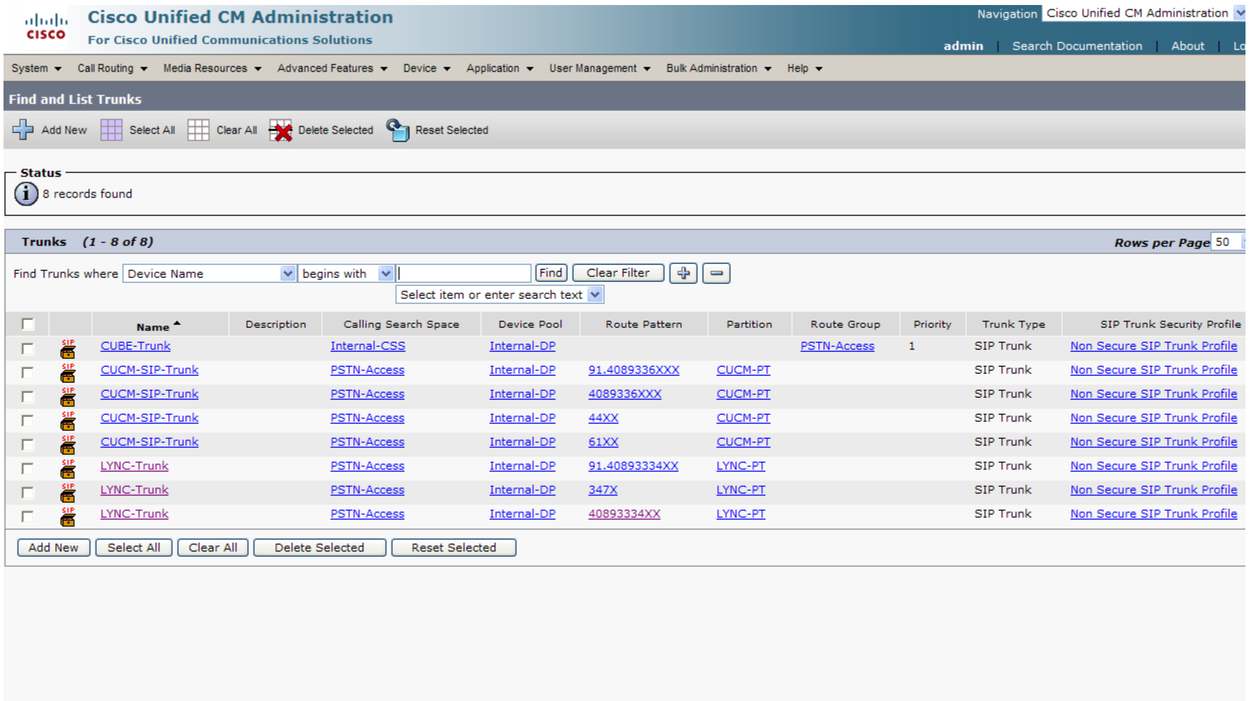
Cisco UCM-SME Route list for PSTN (CUBE) access



SME Route Group for PSTN Trunk Access

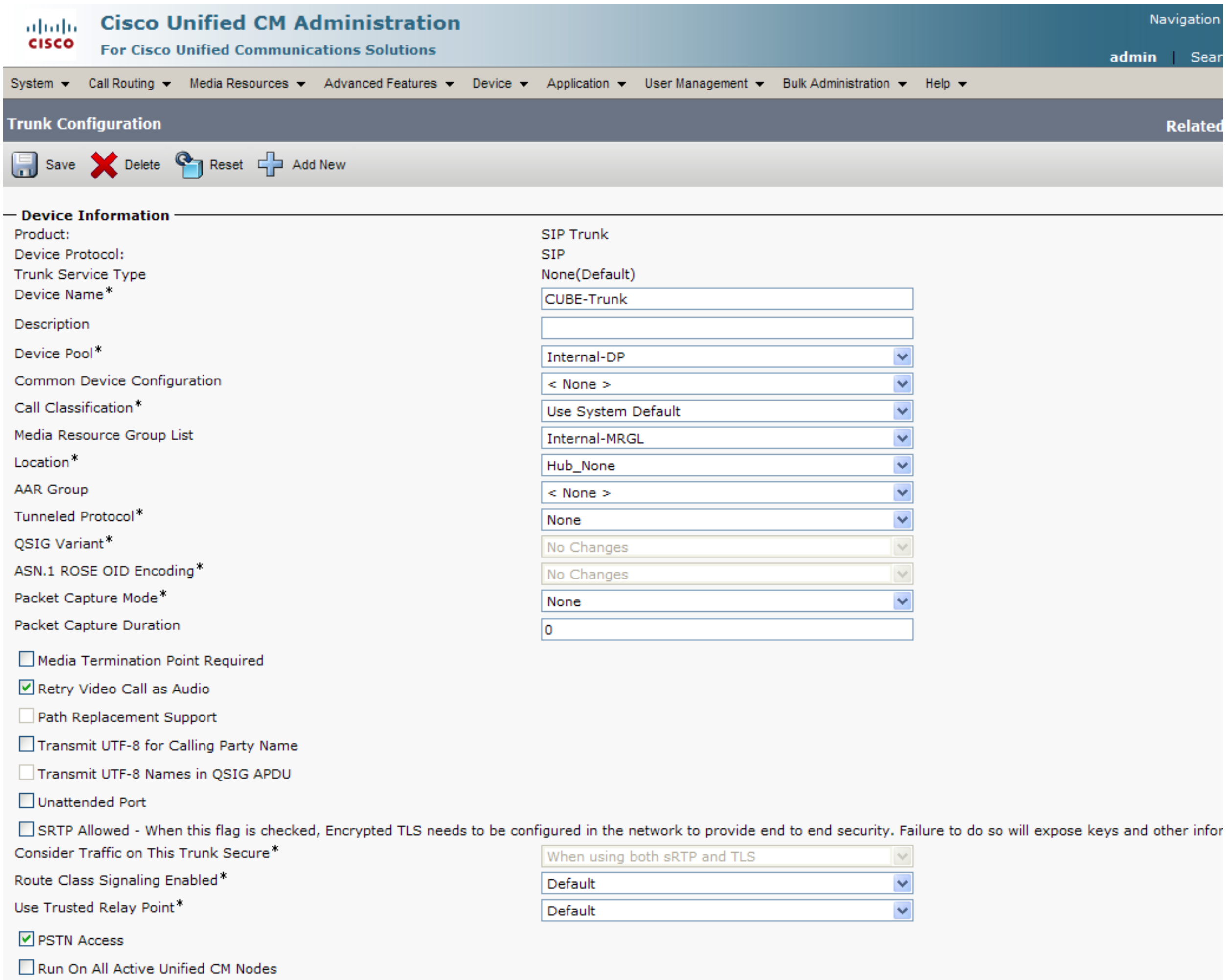


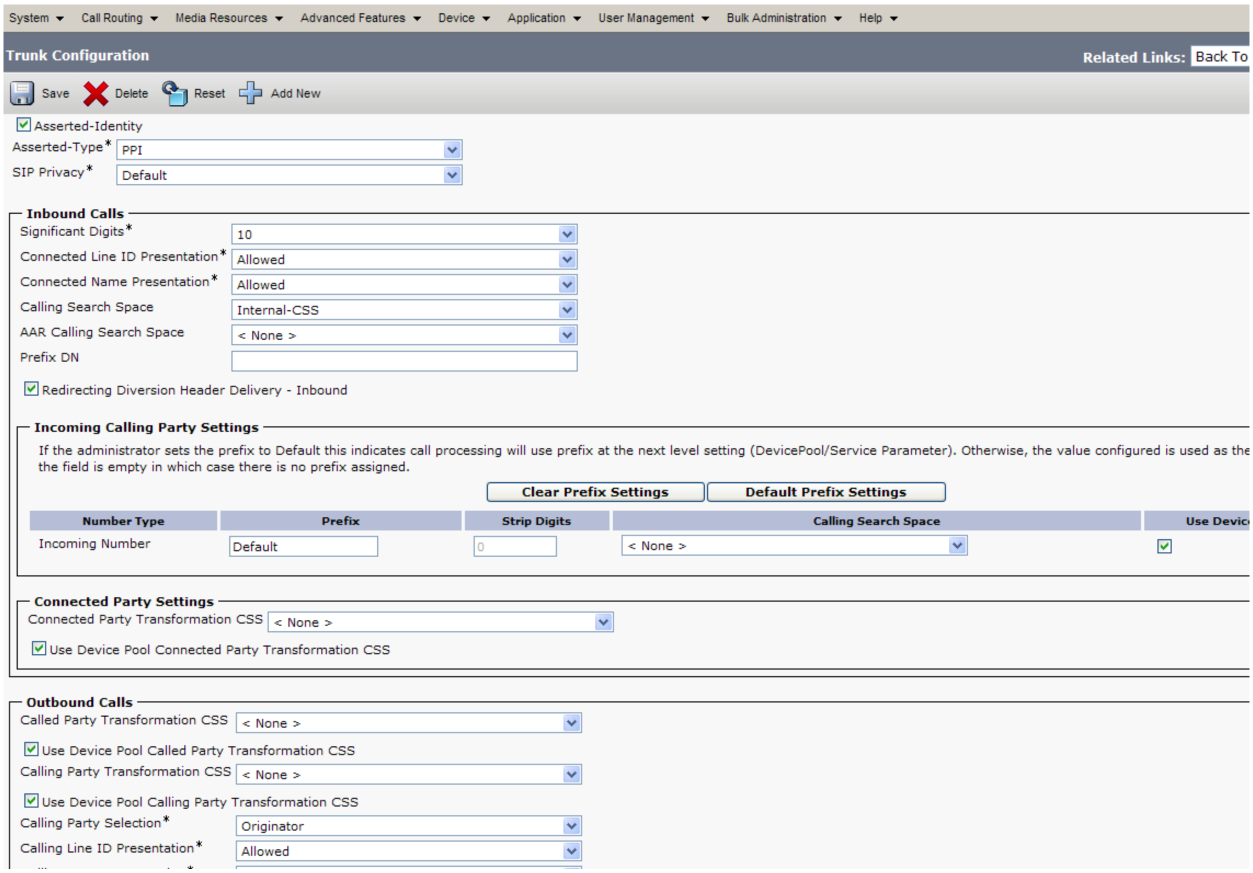
Cisco UCM-SME SIP Trunks:

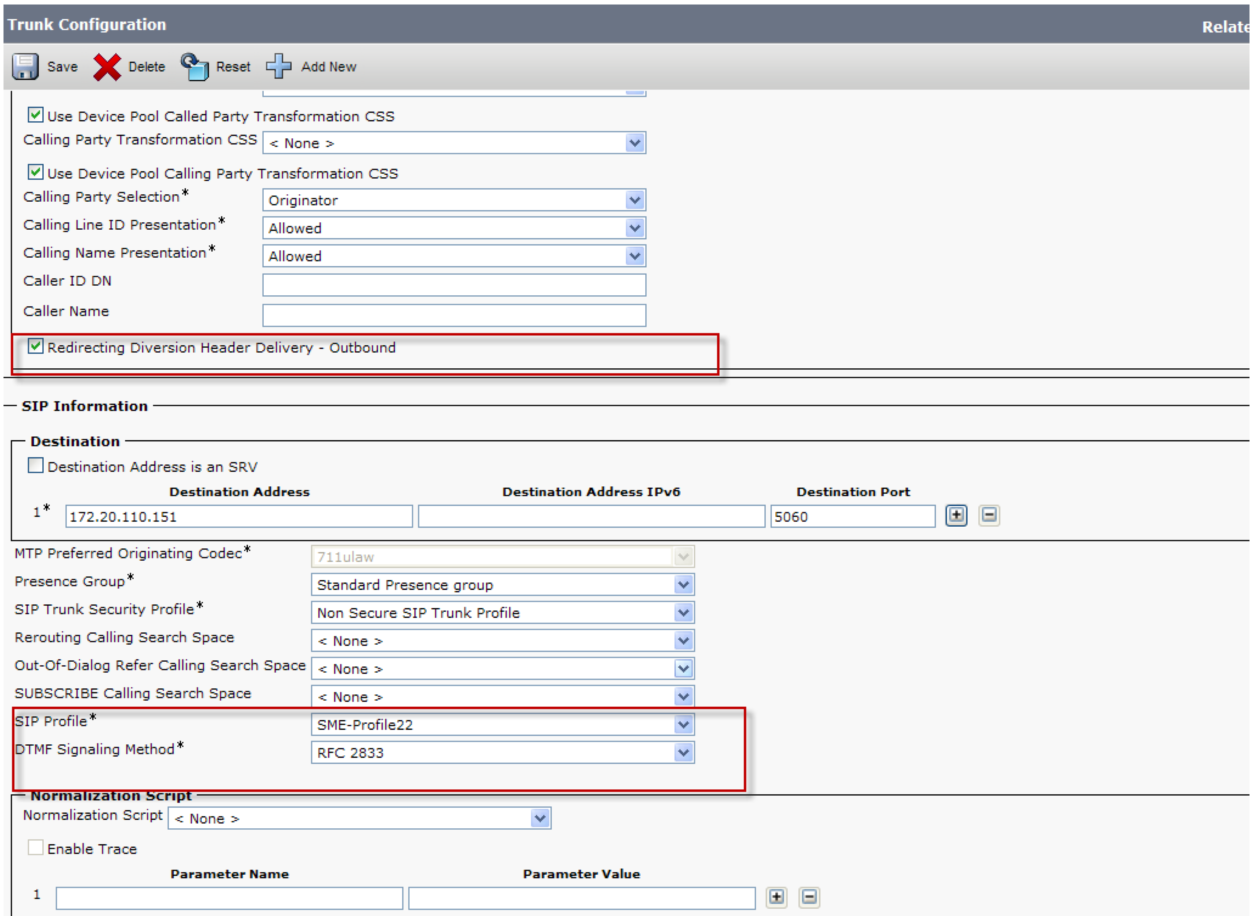


Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Security Profile
CUBE-Trunk		Internal-CSS	Internal-DP			PSTN-Access	1	SIP Trunk	Non Secure SIP Trunk Profile
CUCM-SIP-Trunk		PSTN-Access	Internal-DP	91.4089336XXX	CUCM-PT			SIP Trunk	Non Secure SIP Trunk Profile
CUCM-SIP-Trunk		PSTN-Access	Internal-DP	4089336XXX	CUCM-PT			SIP Trunk	Non Secure SIP Trunk Profile
CUCM-SIP-Trunk		PSTN-Access	Internal-DP	44XX	CUCM-PT			SIP Trunk	Non Secure SIP Trunk Profile
CUCM-SIP-Trunk		PSTN-Access	Internal-DP	61XX	CUCM-PT			SIP Trunk	Non Secure SIP Trunk Profile
LYNC-Trunk		PSTN-Access	Internal-DP	91.40893334XX	LYNC-PT			SIP Trunk	Non Secure SIP Trunk Profile
LYNC-Trunk		PSTN-Access	Internal-DP	347X	LYNC-PT			SIP Trunk	Non Secure SIP Trunk Profile
LYNC-Trunk		PSTN-Access	Internal-DP	40893334XX	LYNC-PT			SIP Trunk	Non Secure SIP Trunk Profile

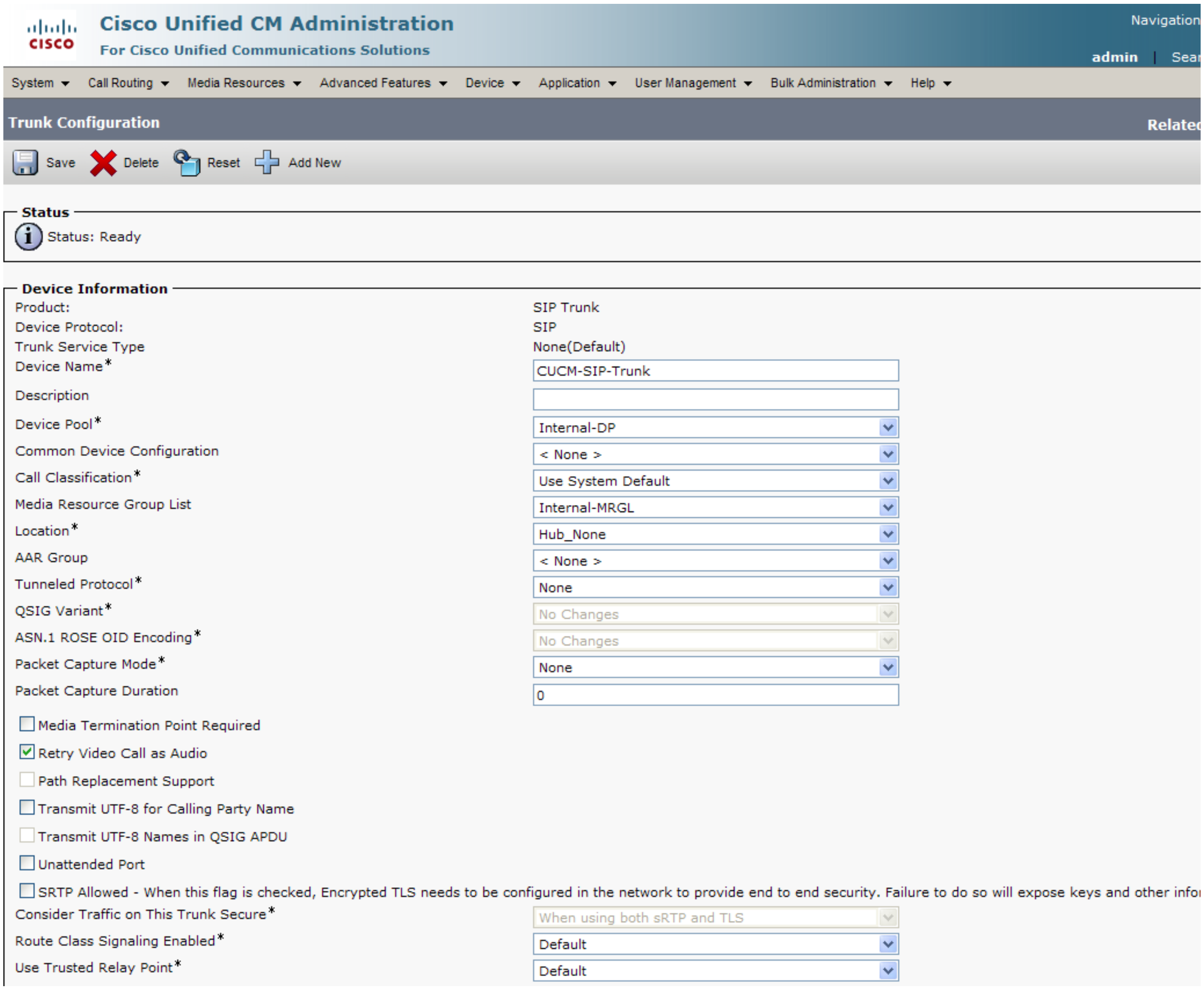
SME SIP trunk to Service Provider via ASR CUBE

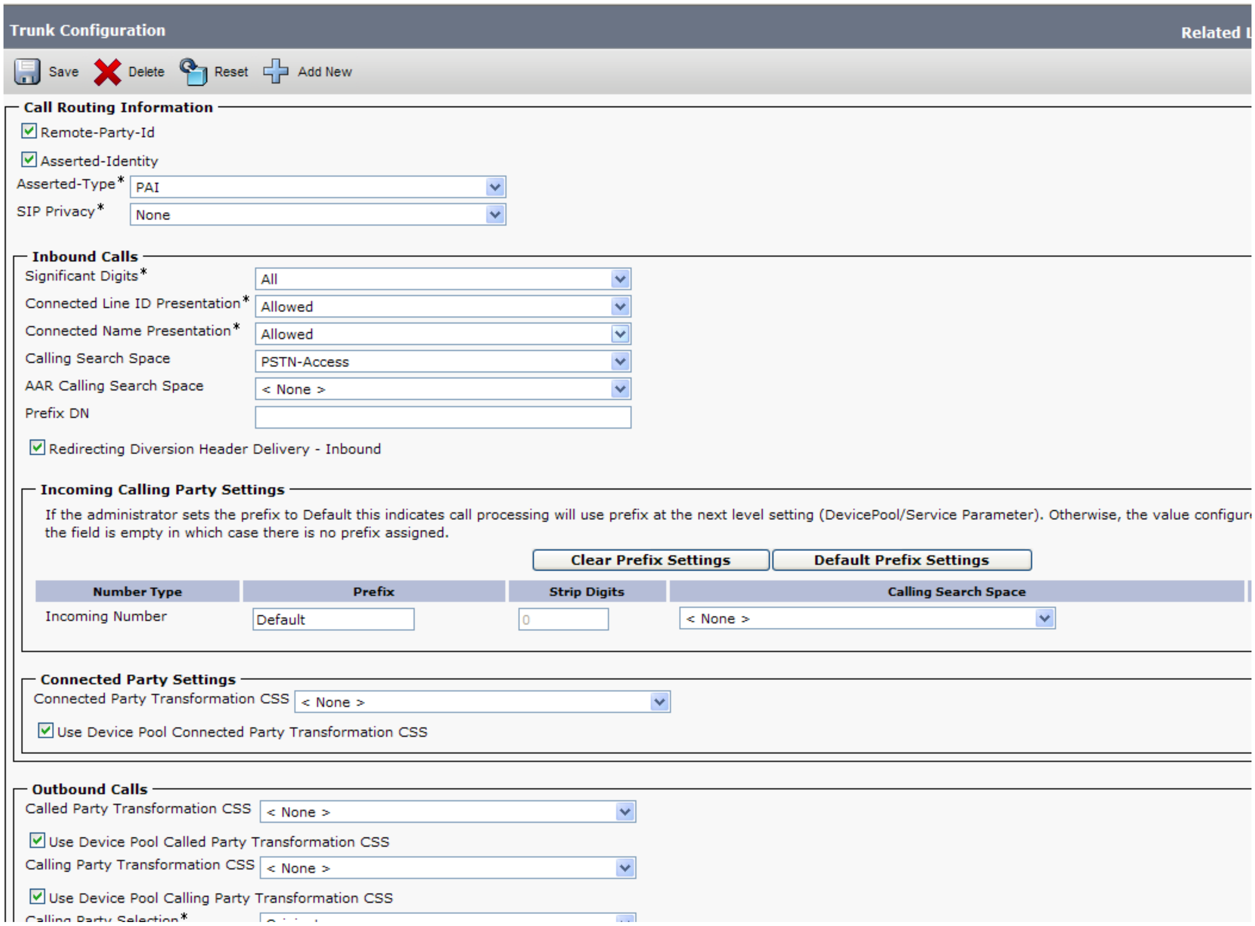


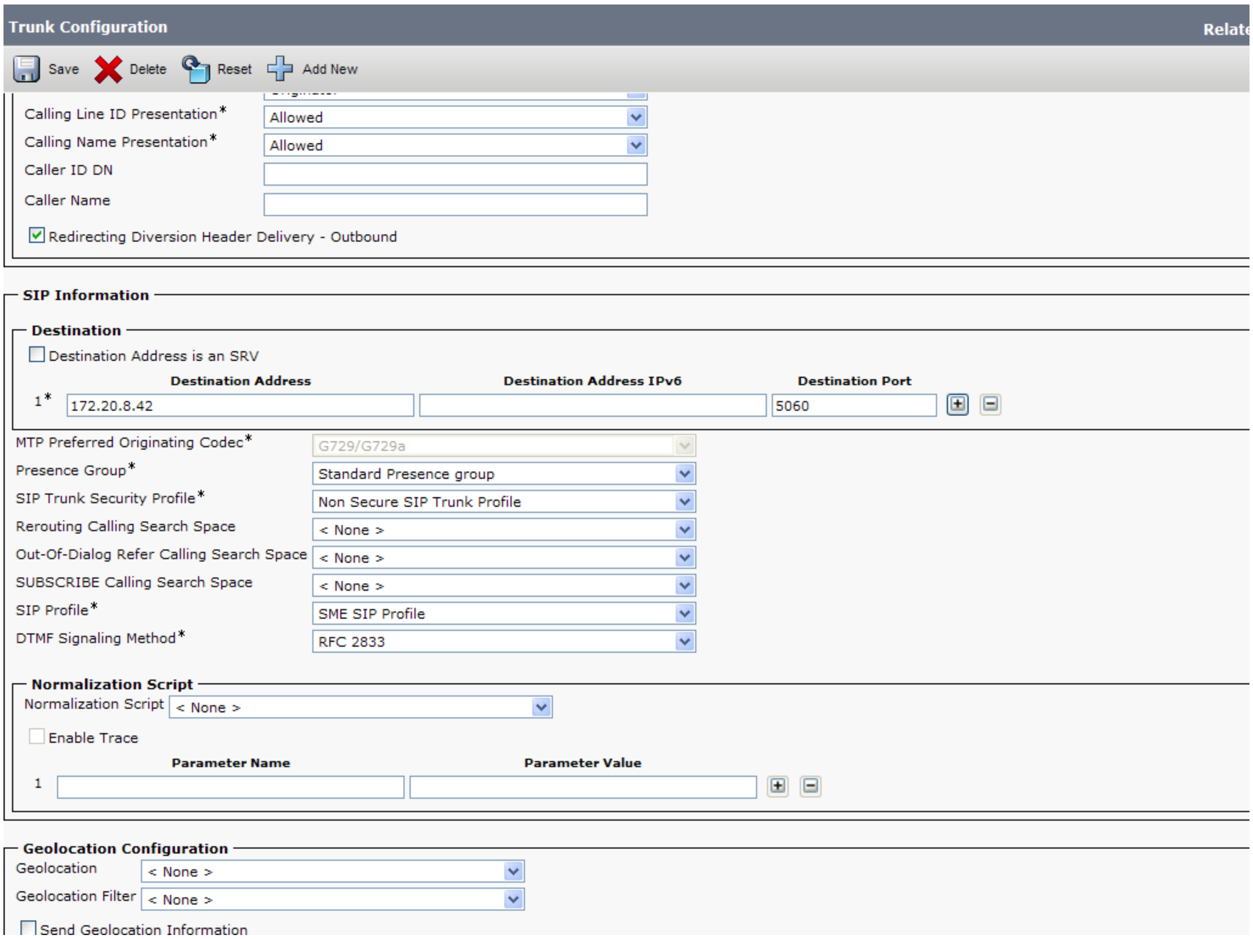

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Trunk Configuration
Related Links: Back To
Save
Delete
Reset
Add New
Asserted-Identity
Asserted-Type* PPI
SIP Privacy* Default
Inbound Calls
Significant Digits* 10
Connected Line ID Presentation* Allowed
Connected Name Presentation* Allowed
Calling Search Space Internal-CSS
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
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 Devic
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* Allowed


Trunk Configuration
Related
Save
Delete
Reset
Add New
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*
Allowed
Calling Name Presentation*
Allowed
Caller ID DN
Caller Name
Redirecting Diversion Header Delivery - Outbound
SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1*
172.20.110.151
5060
MTP Preferred Originating Codec*
711ulaw
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*
SME-Profile22
DTMF Signaling Method*
RFC 2833
Normalization Script
Normalization Script
< None >
Enable Trace
Parameter Name
Parameter Value
1

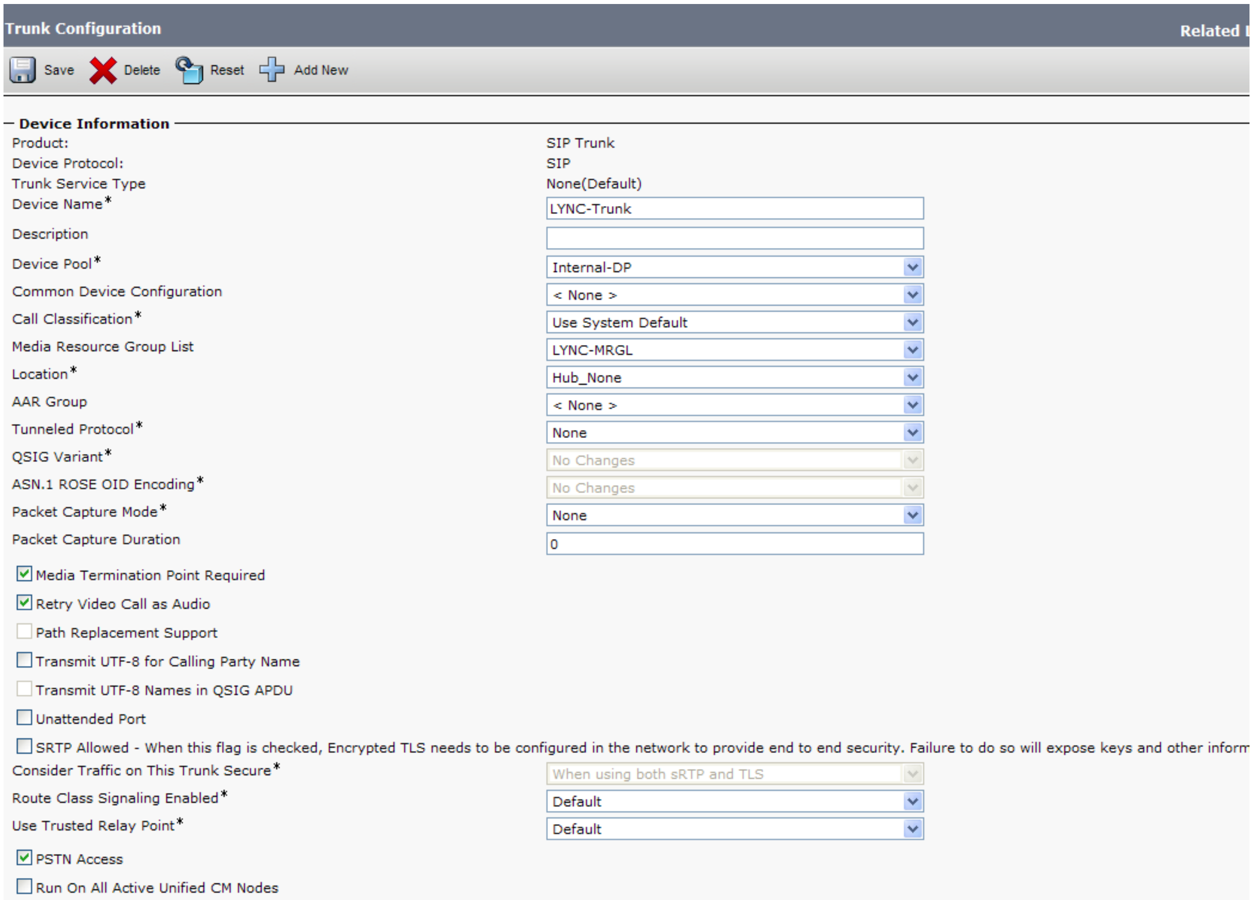
Cisco UCM-SME SIP trunk to Cisco UCM

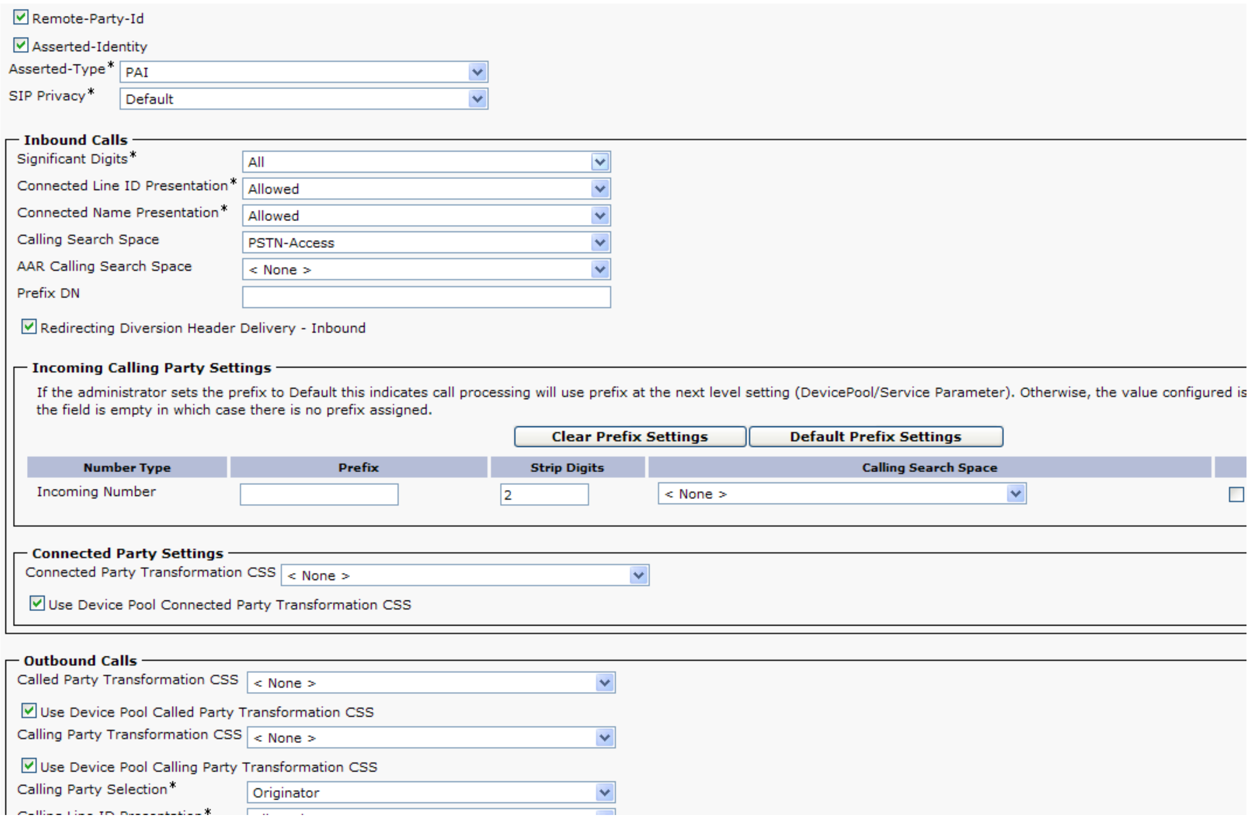


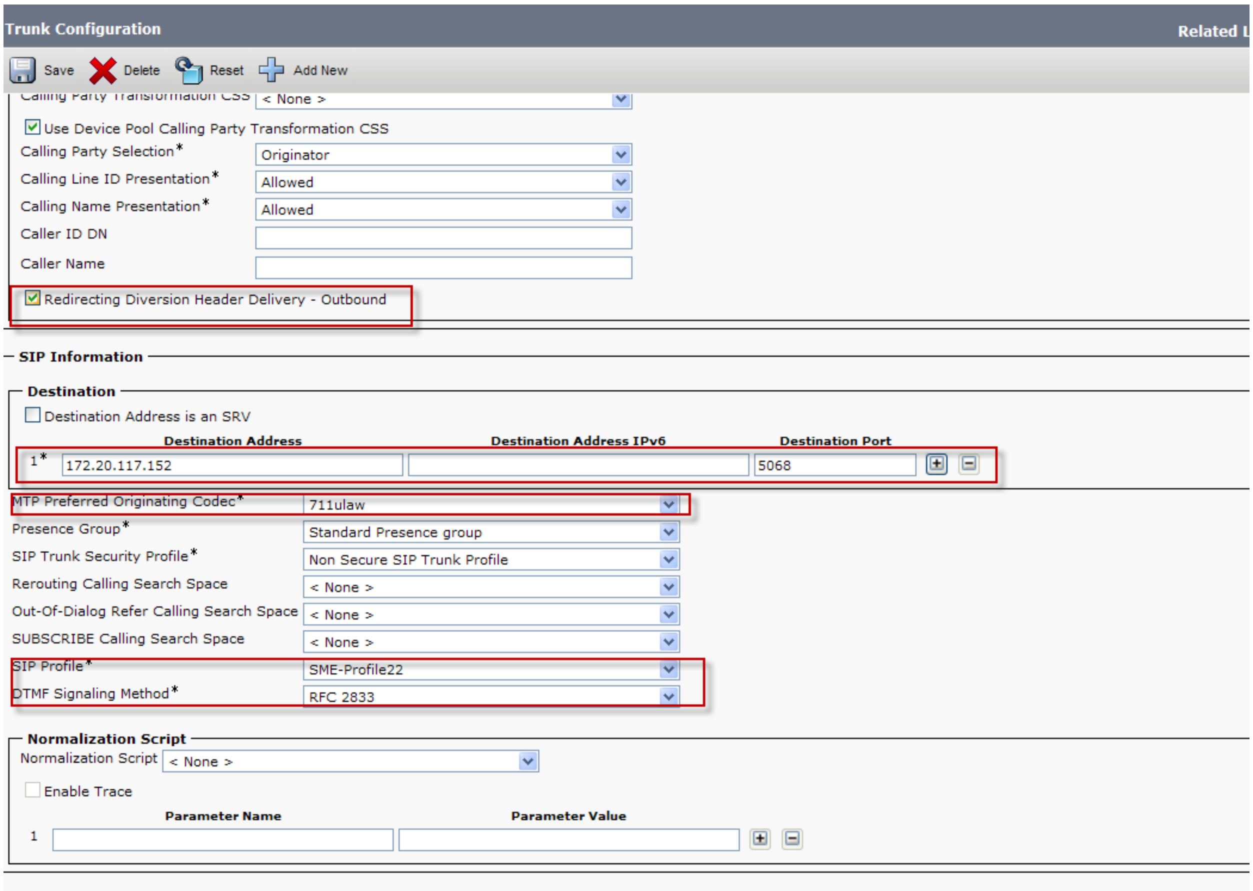

Trunk Configuration
Save
Delete
Reset
Add New
Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type*
PAI
SIP Privacy*
None
Inbound Calls
Significant Digits*
All
Connected Line ID Presentation*
Allowed
Connected Name Presentation*
Allowed
Calling Search Space
PSTN-Access
AAR Calling 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 configur
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
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


Trunk Configuration
Save
Delete
Reset
Add New
Calling Line ID Presentation*
Allowed
Calling Name Presentation*
Allowed
Caller ID DN
Caller Name
Redirecting Diversion Header Delivery - Outbound
SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1*
172.20.8.42
5060
MTP Preferred Originating Codec*
G729/G729a
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*
SME SIP Profile
DTMF Signaling Method*
RFC 2833
Normalization Script
Normalization Script
< None >
Enable Trace
Parameter Name
Parameter Value
1
Geolocation Configuration
Geolocation
< None >
Geolocation Filter
< None >
Send Geolocation Information

Cisco UCM-SME SIP trunk to Microsoft LYNC 2010 Server Standard Edition

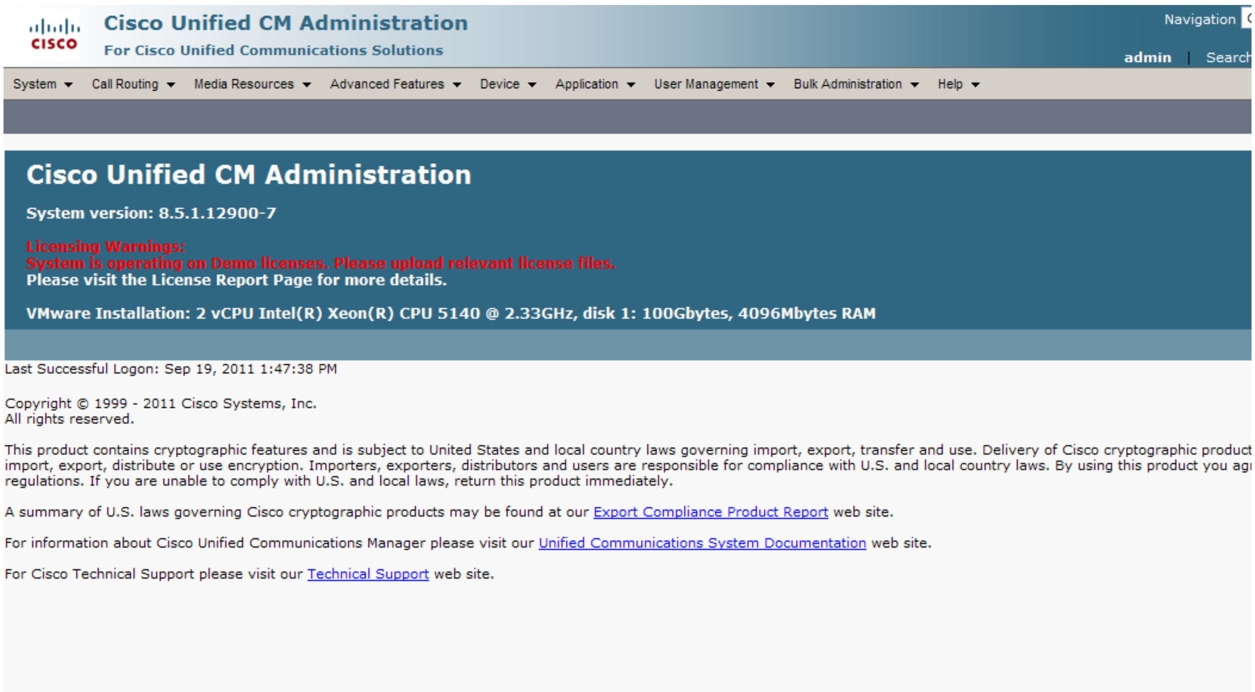



Remote-Party-Id
Asserted-Identity
Asserted-Type* PAI
SIP Privacy* Default
Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Allowed
Connected Name Presentation* Allowed
Calling Search Space PSTN-Access
AAR Calling Search Space < None >
Prefix DN
Redirecting Diversion Header Delivery - 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
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
Incoming Number
2
< 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

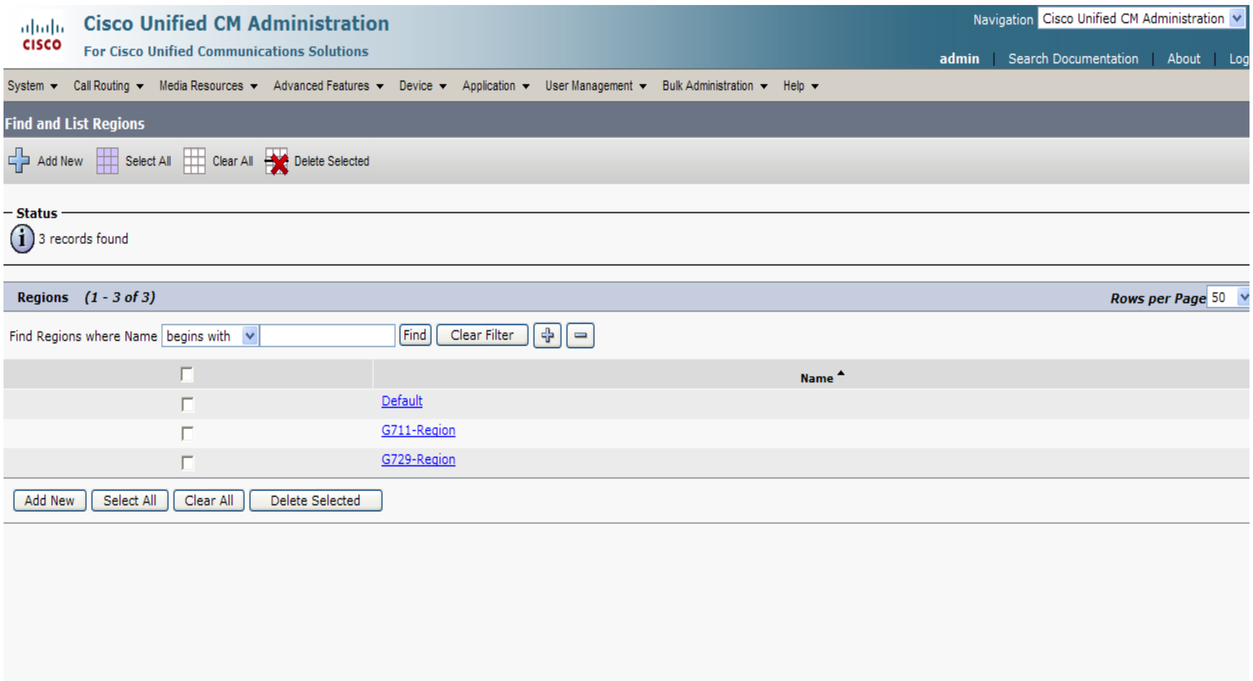

Trunk Configuration
Related L
Save
Delete
Reset
Add New
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation CSS
Calling Party Selection*
Originator
Calling Line ID Presentation*
Allowed
Calling Name Presentation*
Allowed
Caller ID DN
Caller Name
Redirecting Diversion Header Delivery - Outbound
SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1*
172.20.117.152
5068
MTP Preferred Originating Codec*
711ulaw
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*
SME-Profile22
DTMF Signaling Method*
RFC 2833
Normalization Script
Normalization Script
< None >
Enable Trace
Parameter Name
Parameter Value
1

Configuring the Cisco Unified Communications Manager

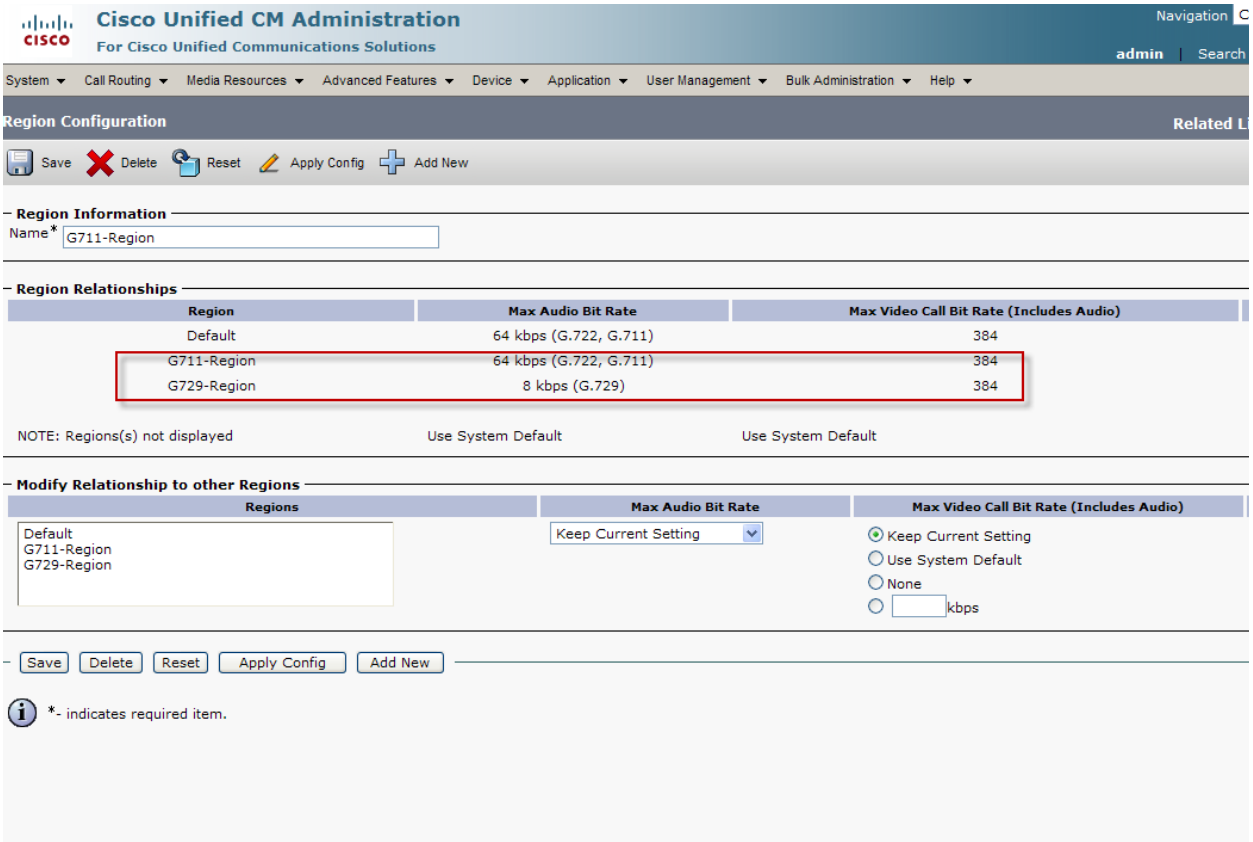
Cisco UCM Softw are release



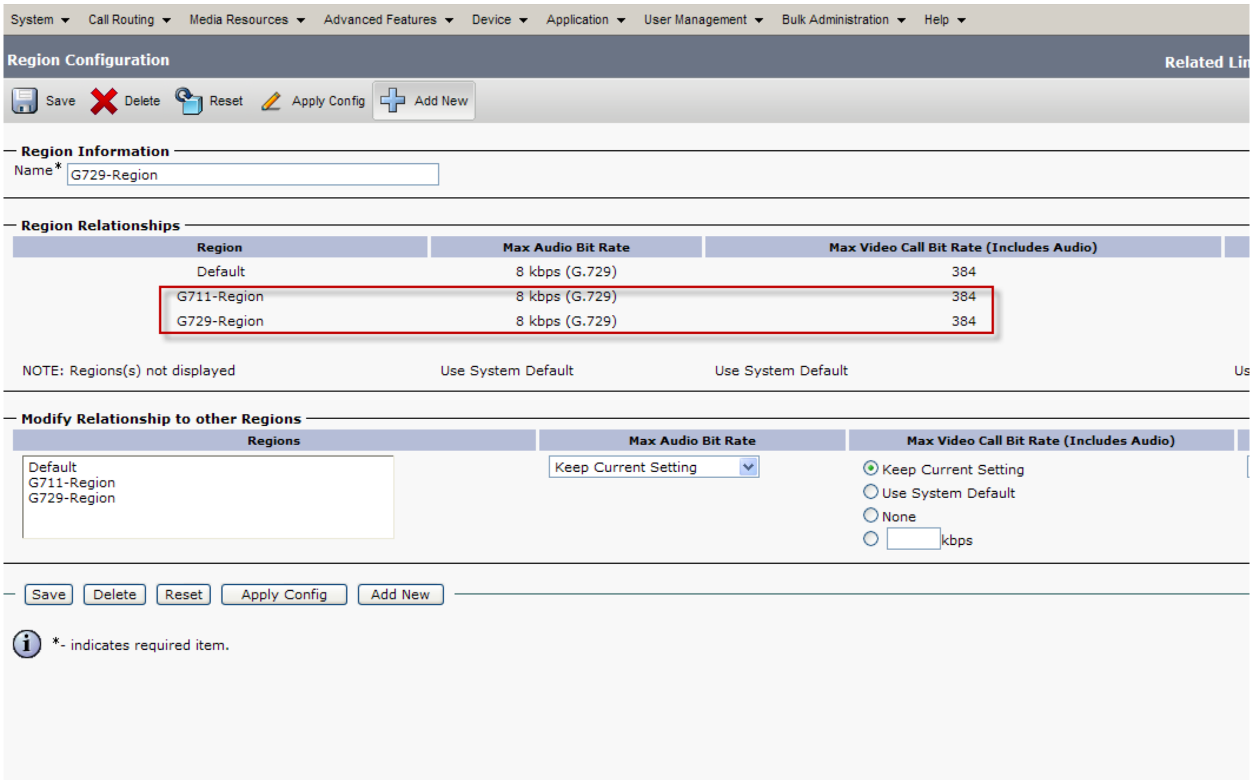
Cisco UCM Regions Configurations



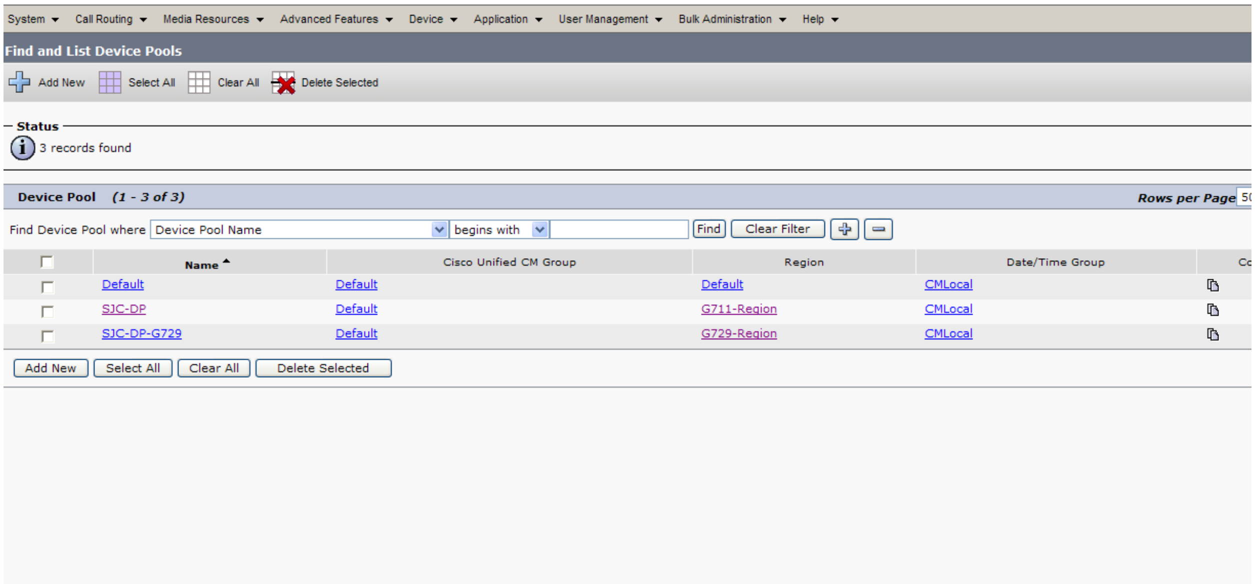
Cisco UCM Regions Configurations – G711 Region



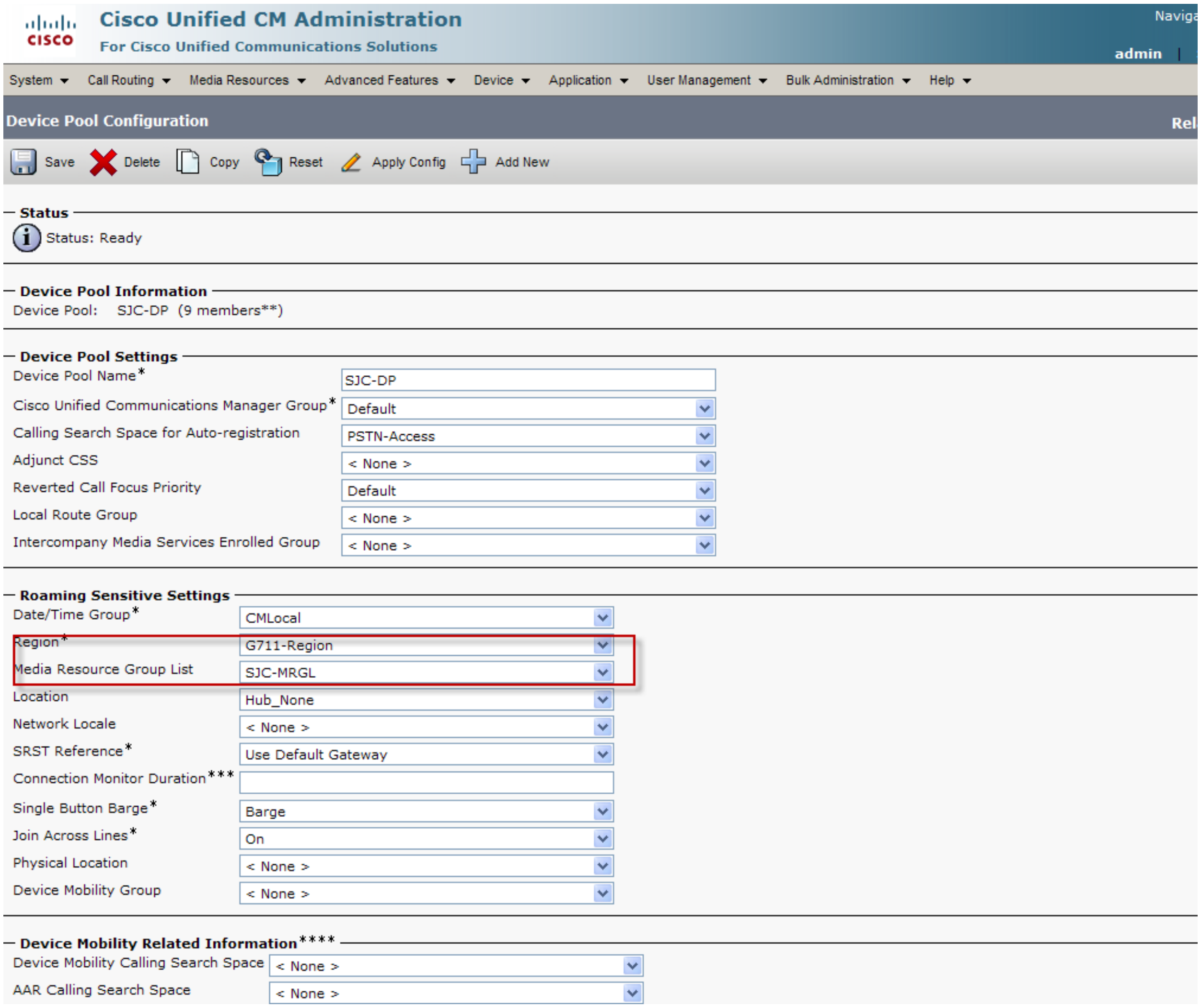
Cisco UCM Region Configurations – G729 Region

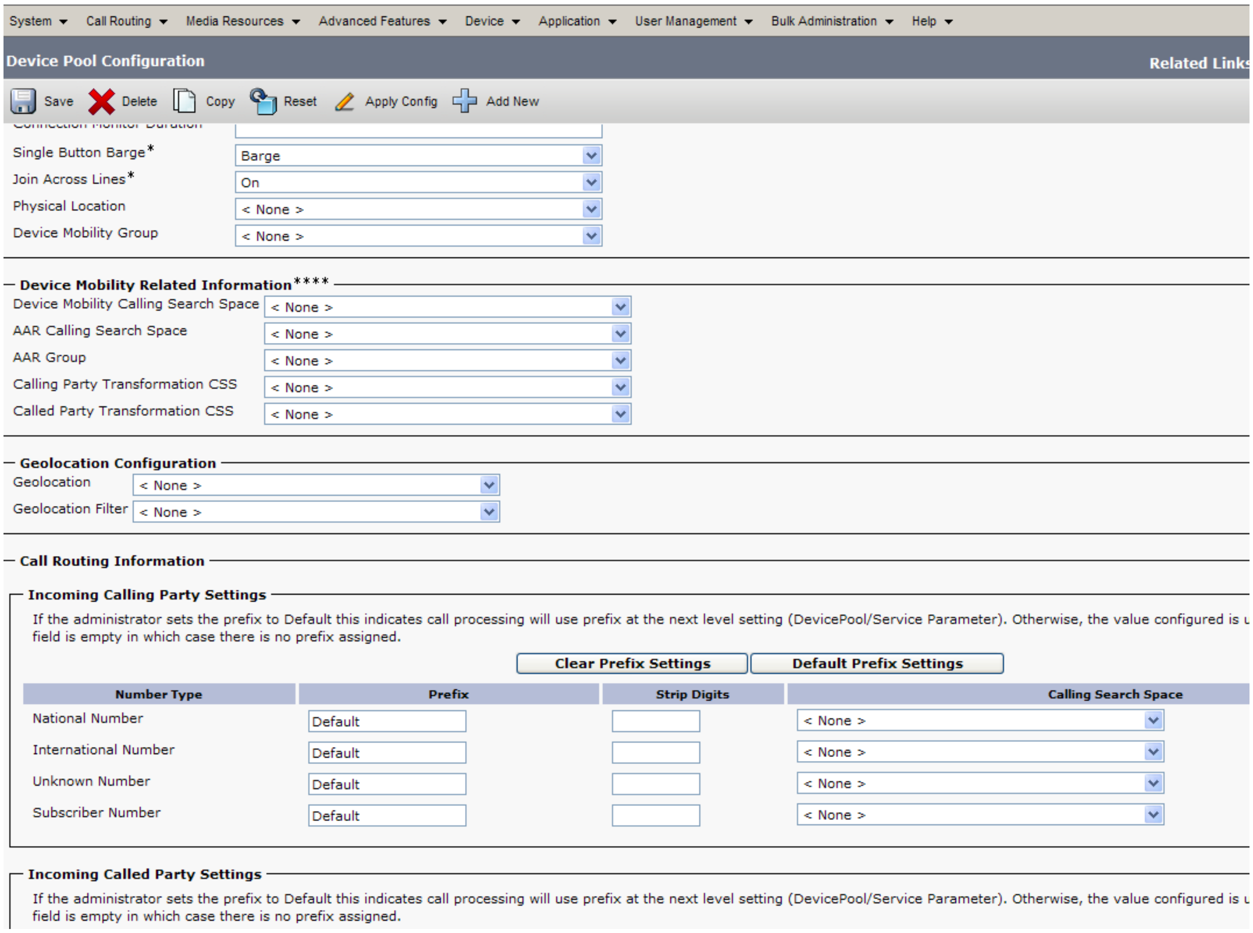


Cisco UCM Device Pool

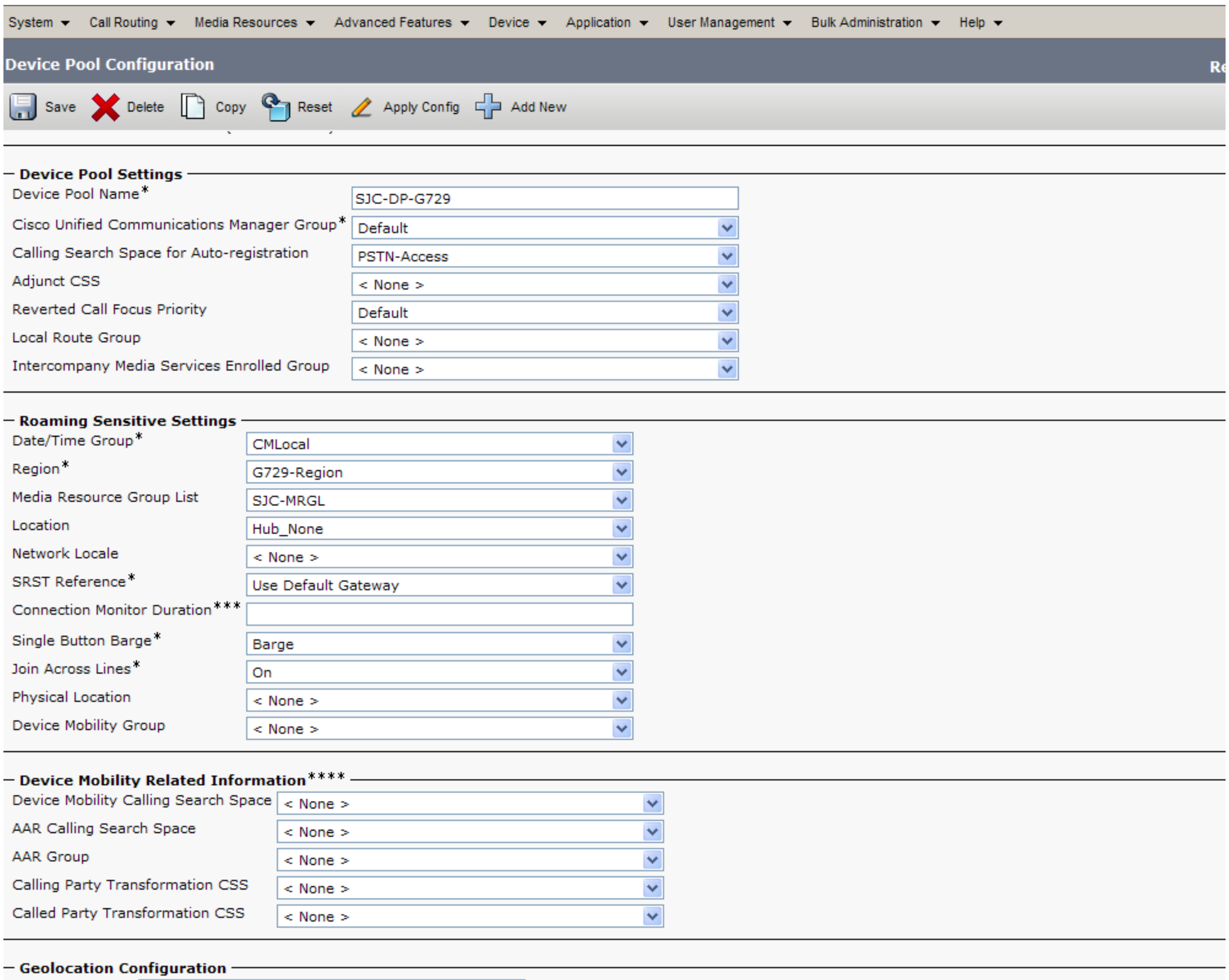


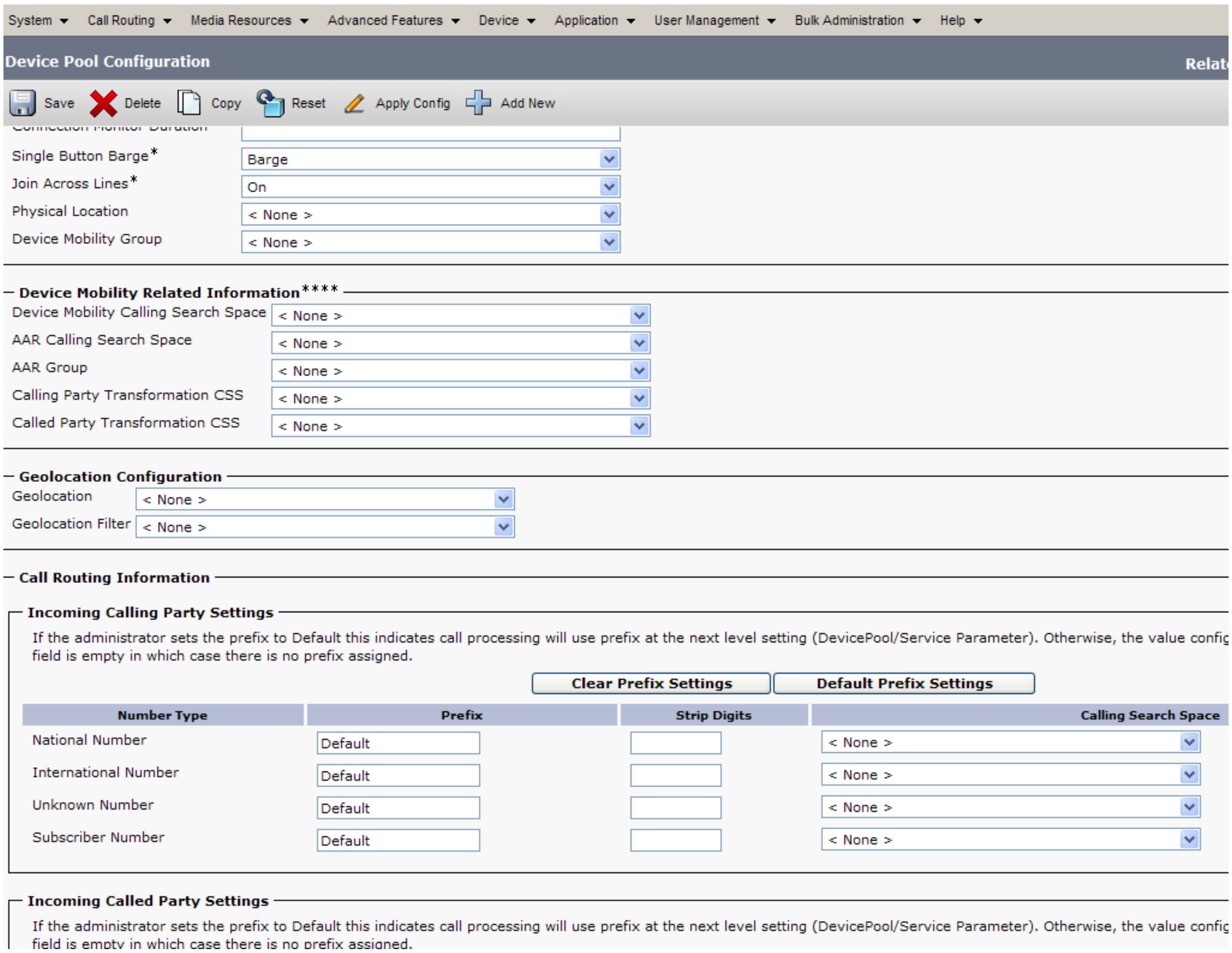
Cisco UCM Device Pool for G711



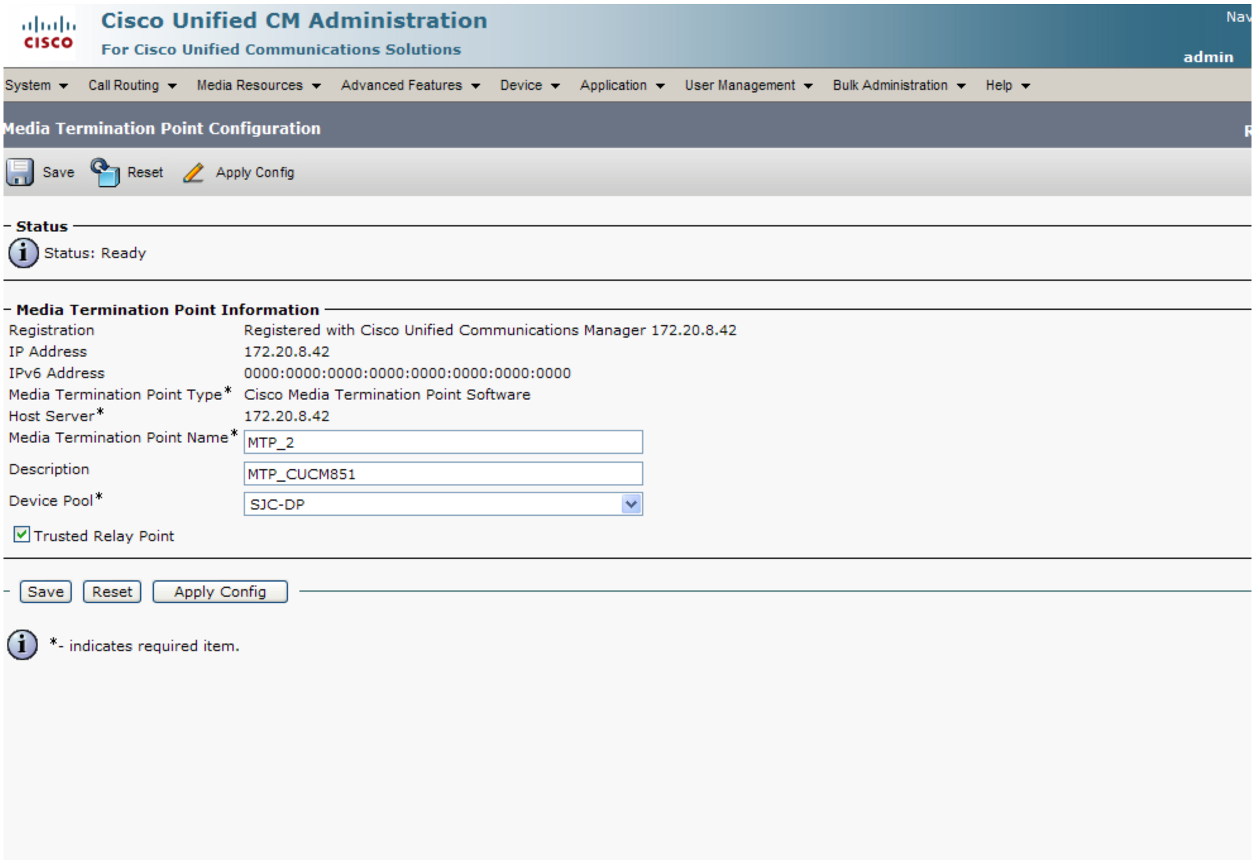

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Device Pool Configuration
Related Links
Save
Delete
Copy
Reset
Apply Config
Add New
Single Button Barge*
Barge
Join Across Lines*
On
Physical Location
< None >
Device Mobility Group
< None >
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 u
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 u
field is empty in which case there is no prefix assigned.

Cisco UCM Device Pool for G729

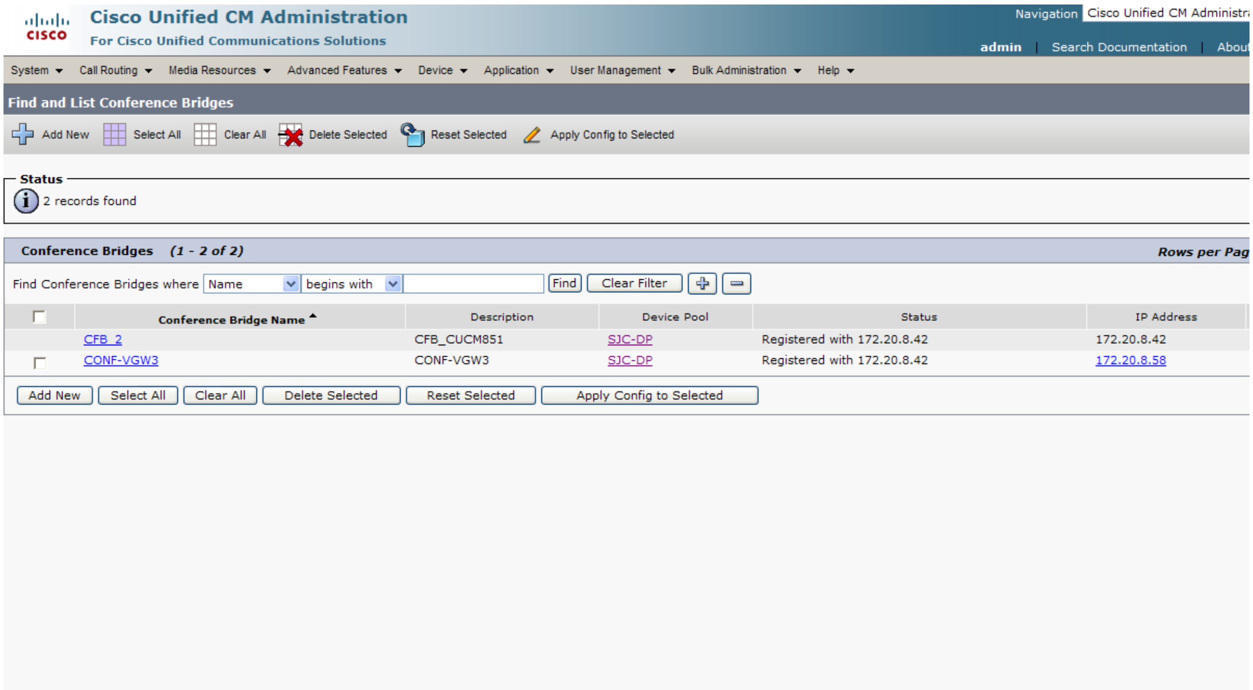



System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Device Pool Configuration
Save Delete Copy Reset Apply Config Add New
Single Button Barge*
Barge
Join Across Lines*
On
Physical Location
< None >
Device Mobility Group
< None >
Device Mobility 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 config
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 config
field is empty in which case there is no prefix assigned.

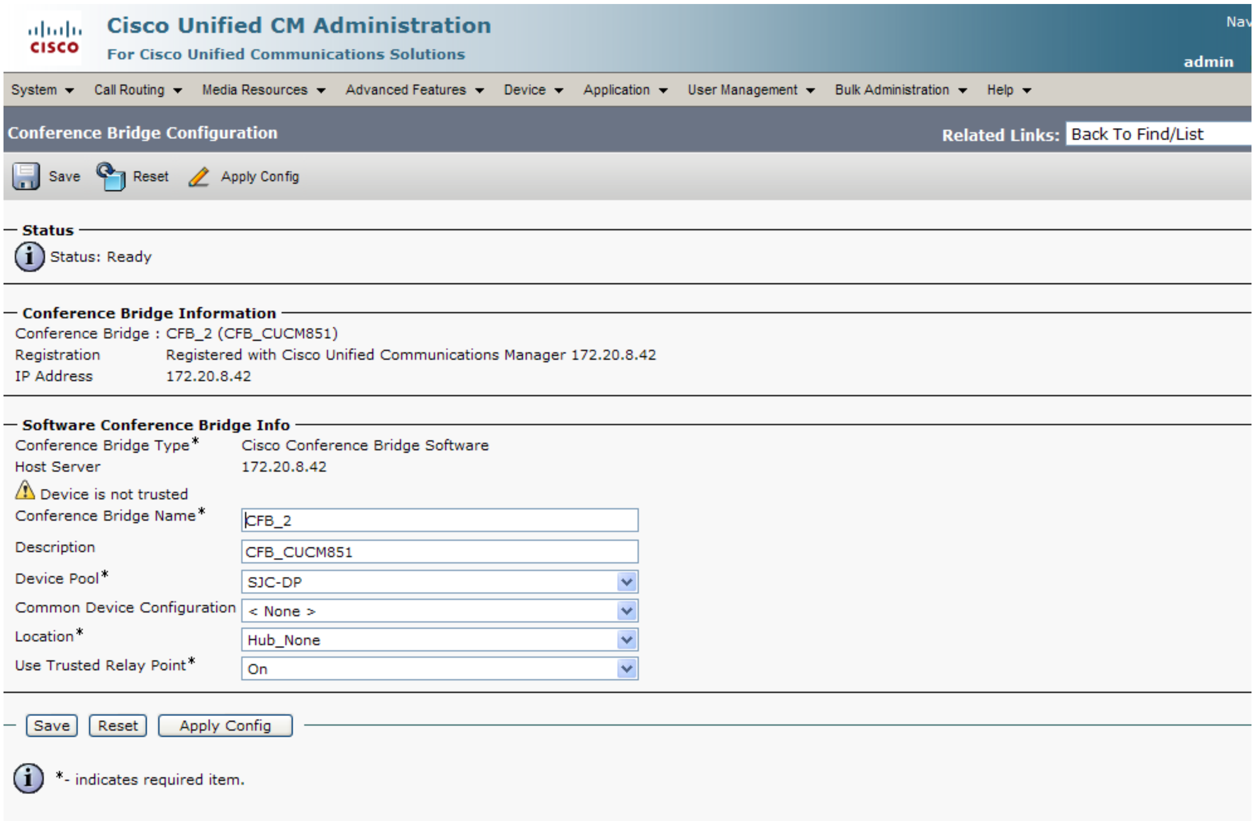
Cisco UCM Media Termination Point Configuration



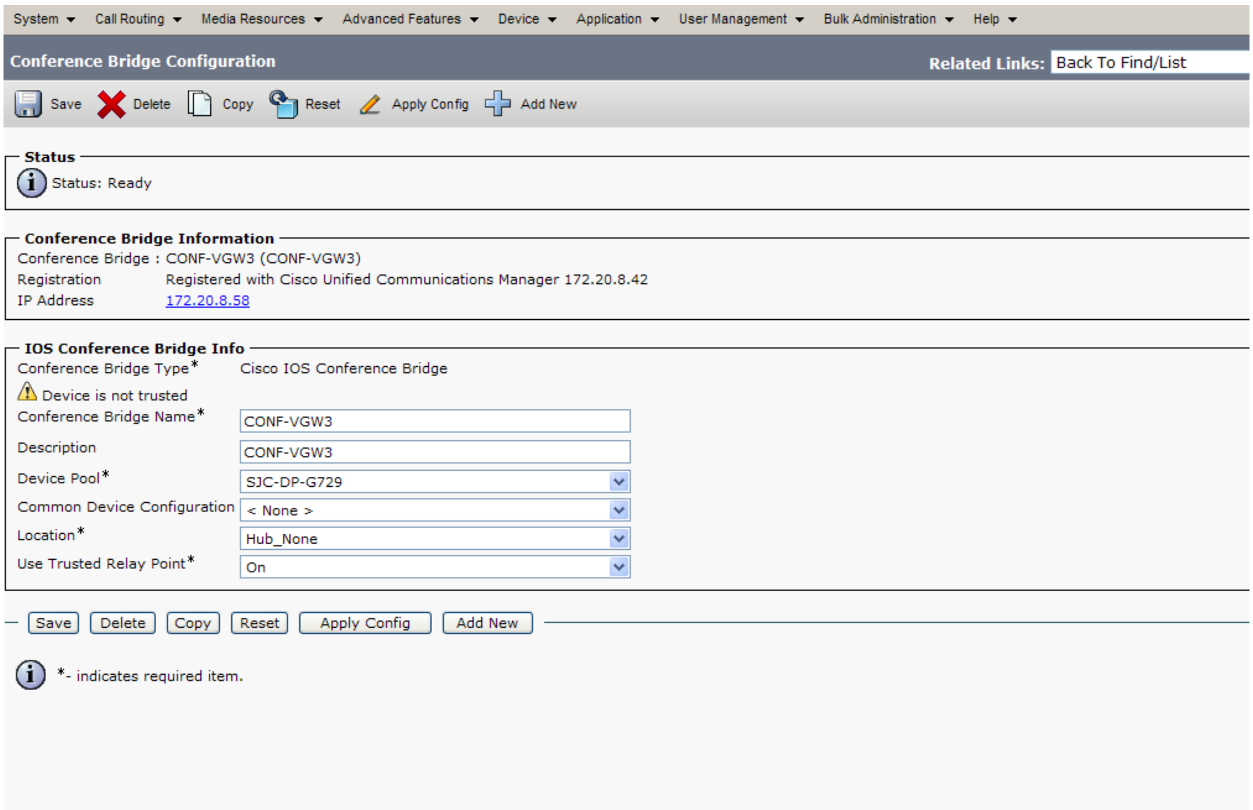
Cisco UCM Conference Bridge Configurations



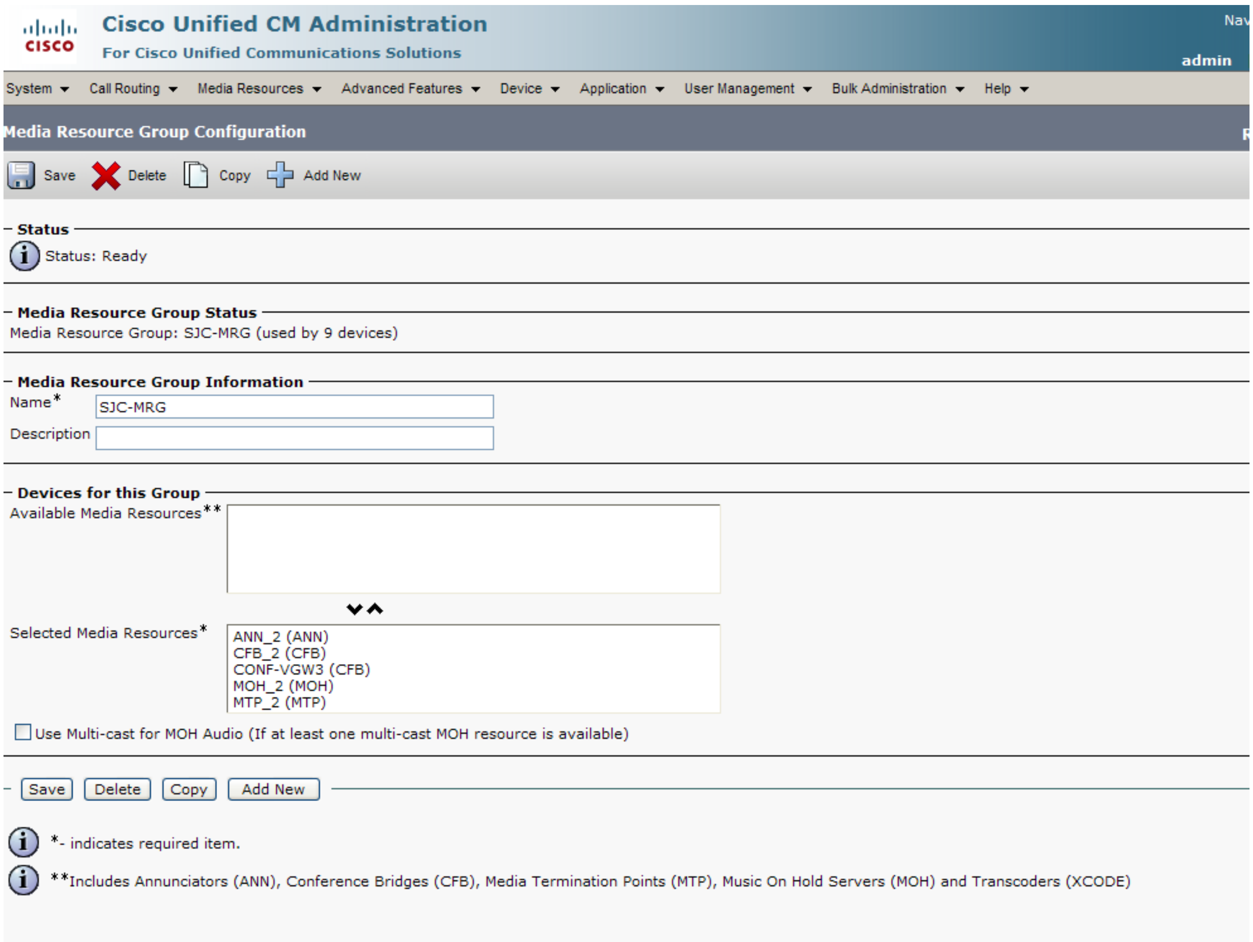
Software Conference Bridge



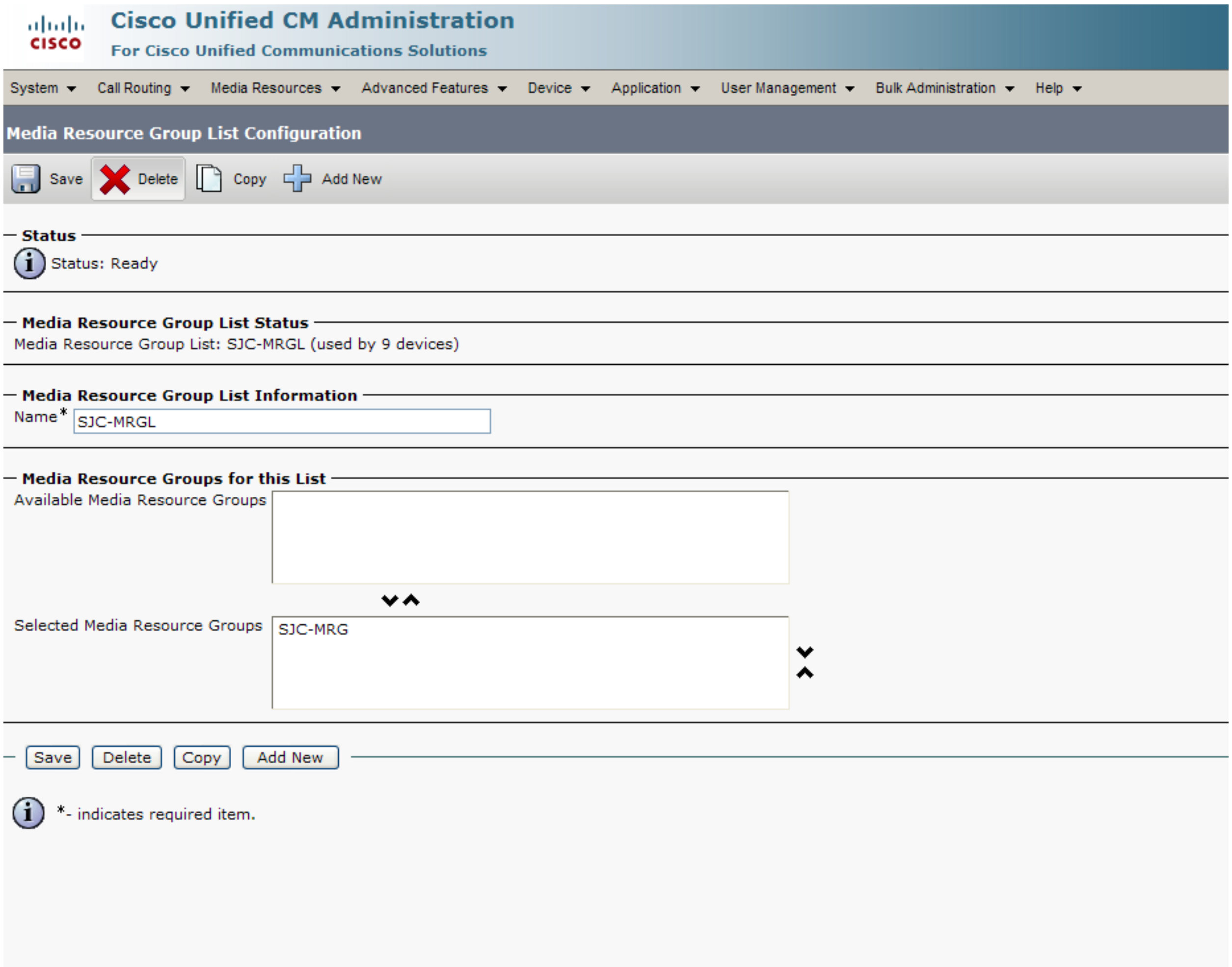
Hardware Conference Bridge



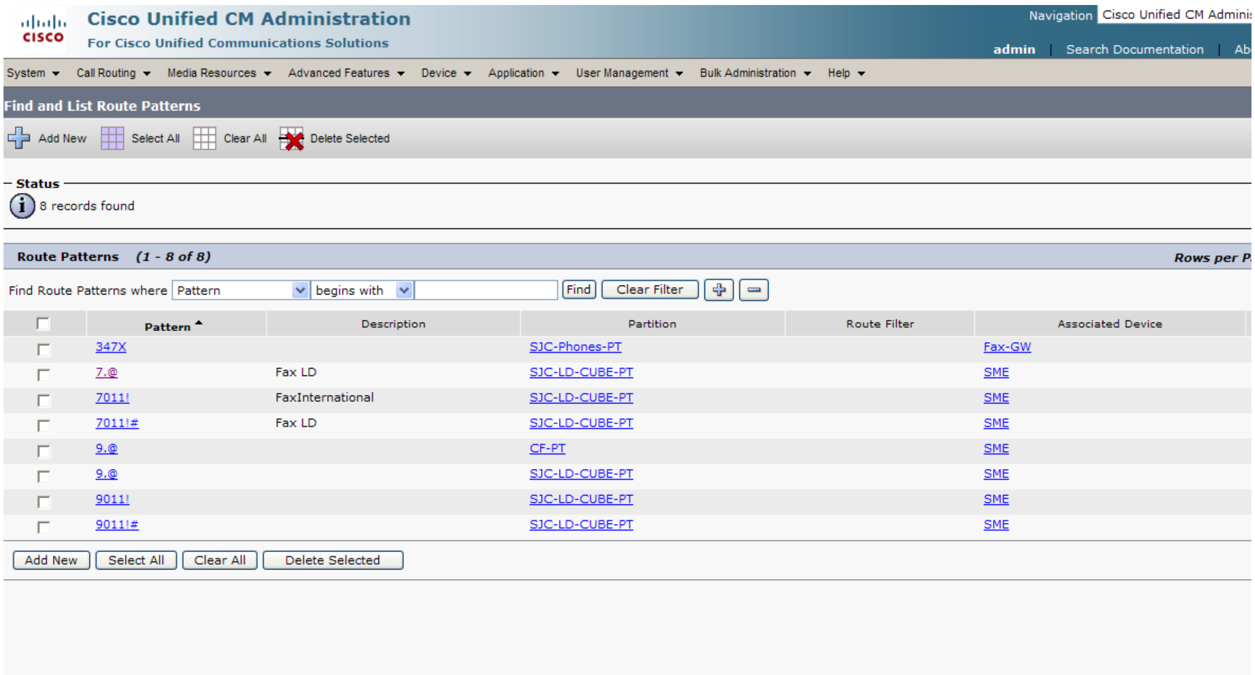
Cisco UCM Media Resource Group



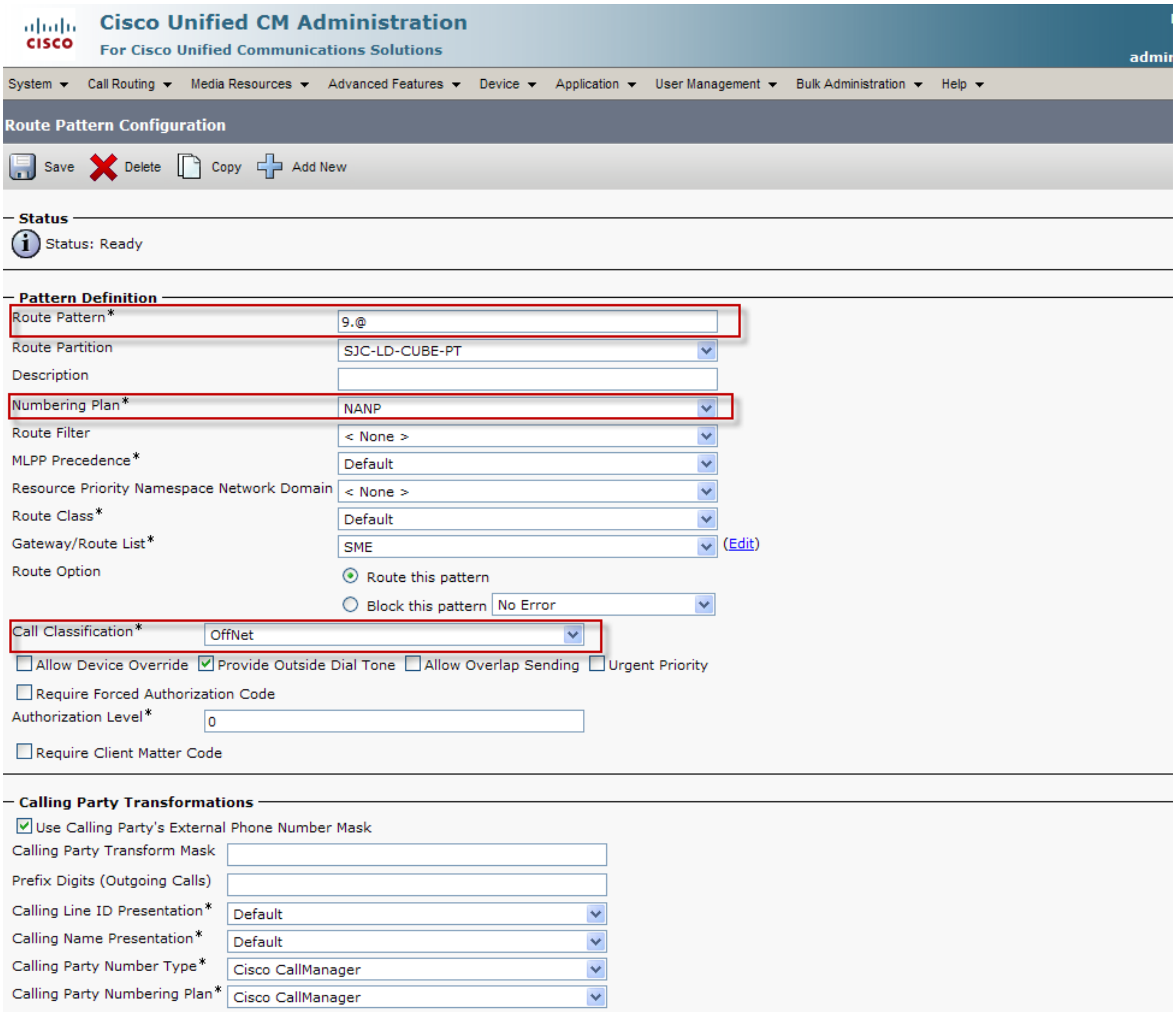
Cisco UCM Media Resource Group List

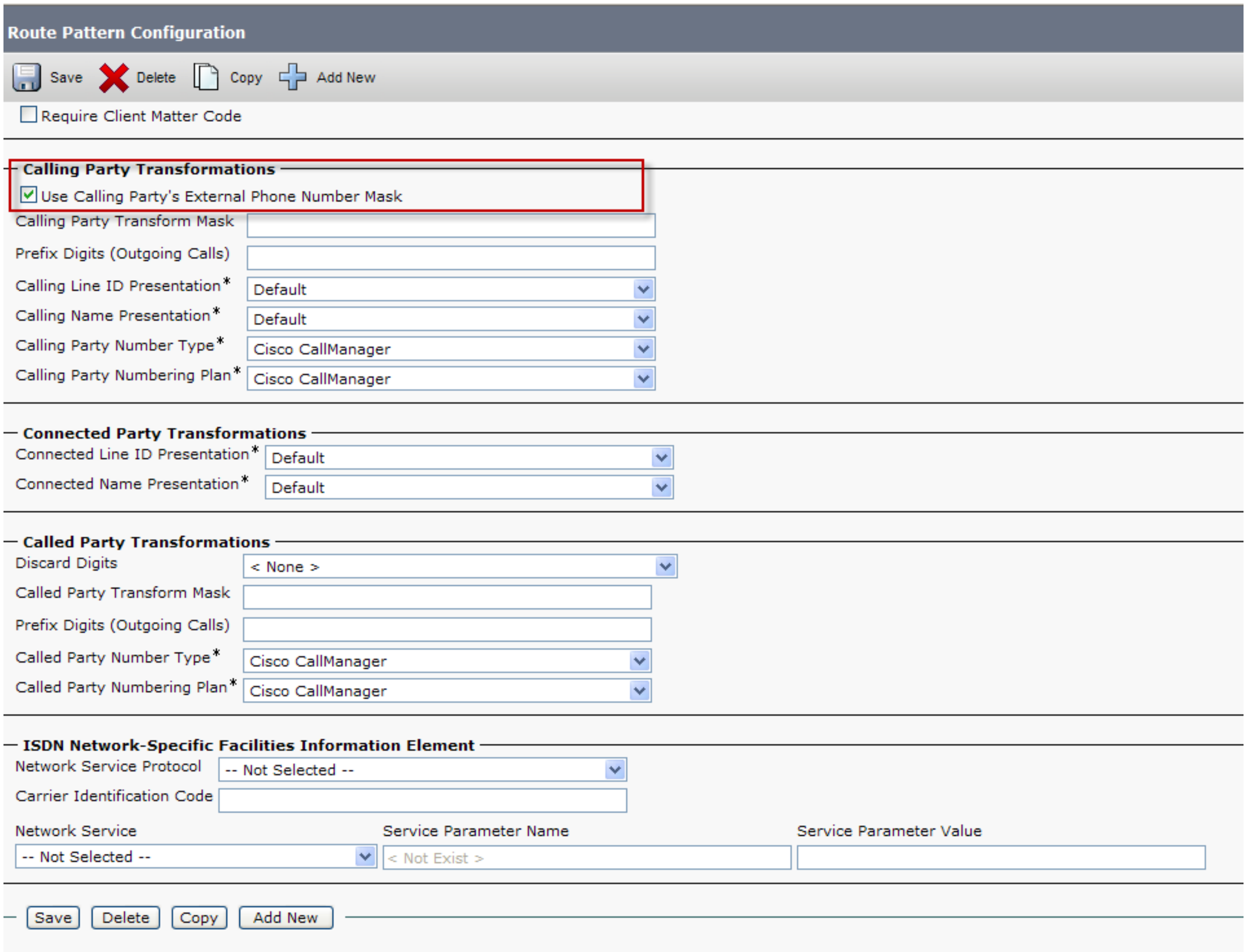


Cisco UCM Route Pattern Configuration



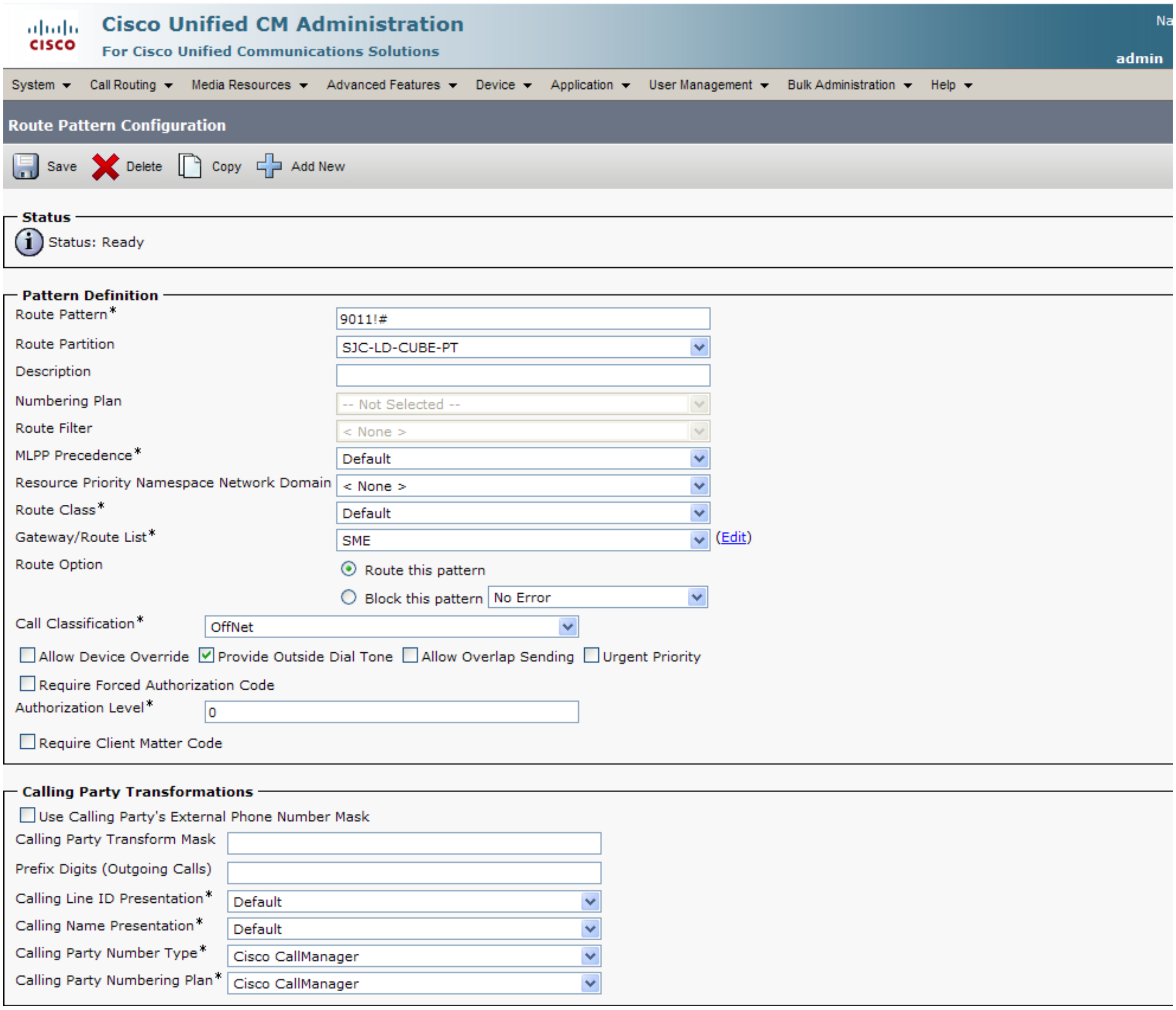
PSTN Route Pattern

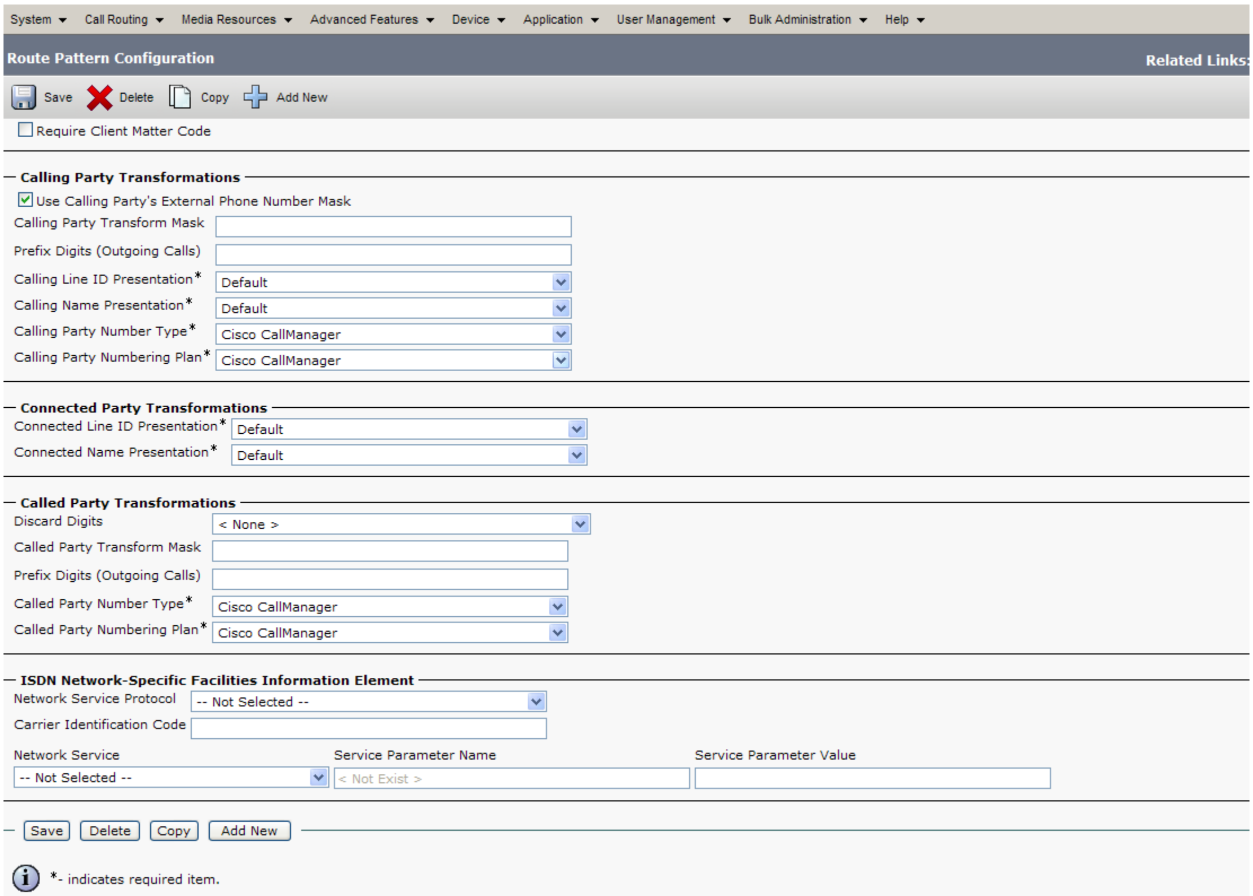




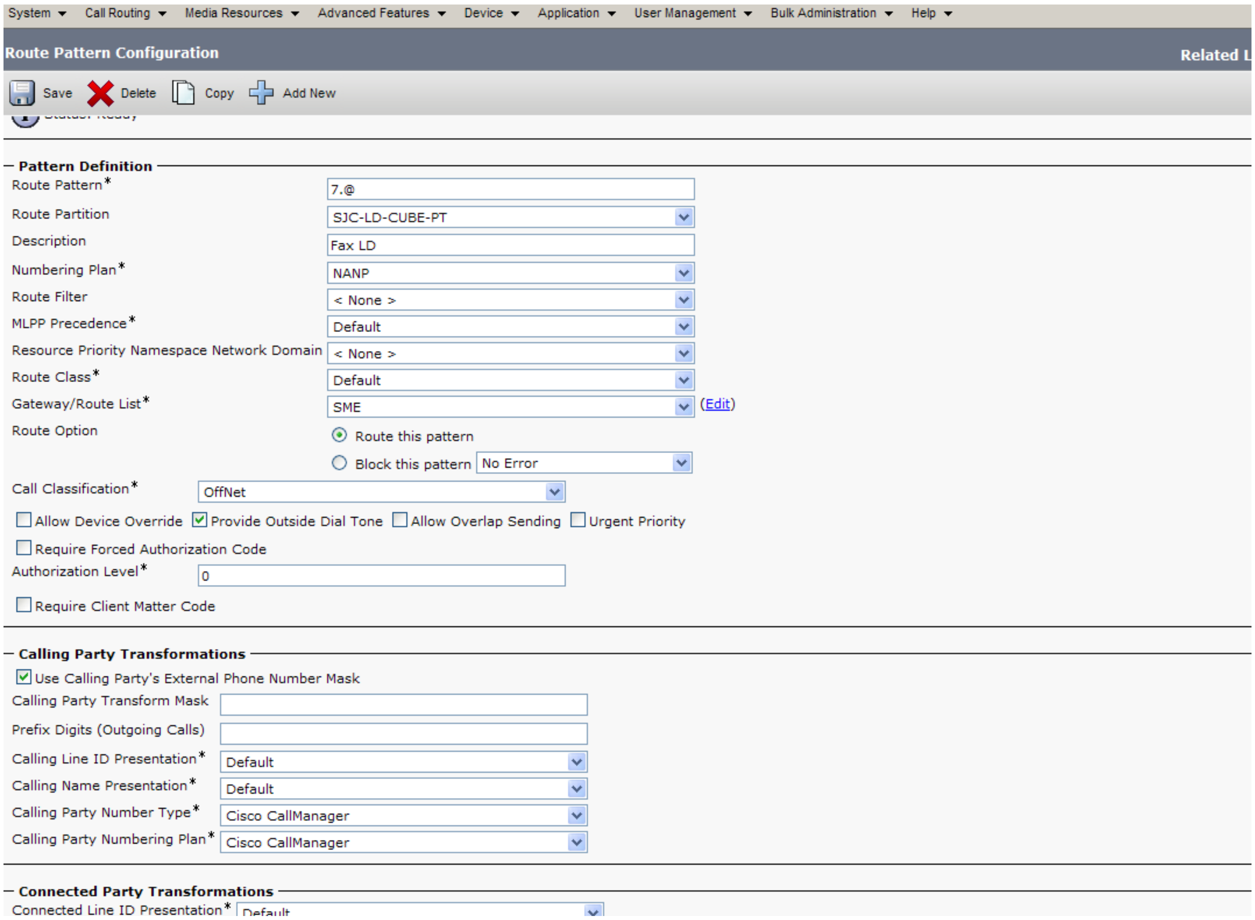
Route Pattern Configuration
Save
Delete
Copy
Add New
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling Party Number Type*
Cisco CallManager
Calling Party Numbering Plan*
Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Called Party Transformations
Discard Digits
< None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Cisco CallManager
Called Party Numbering Plan*
Cisco CallManager
ISDN Network-Specific Facilities Information Element
Network Service Protocol
-- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save
Delete
Copy
Add New

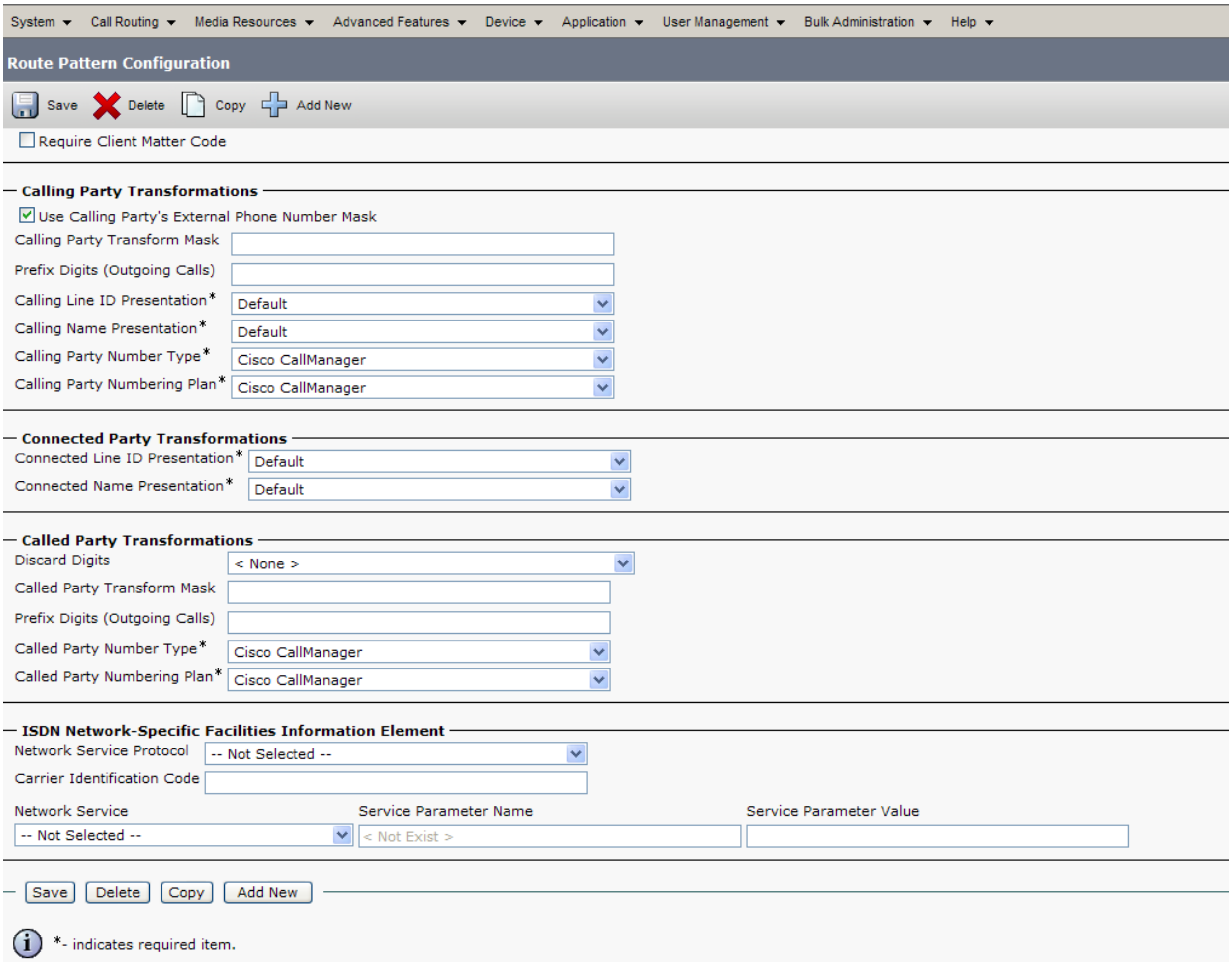
International Route Pattern



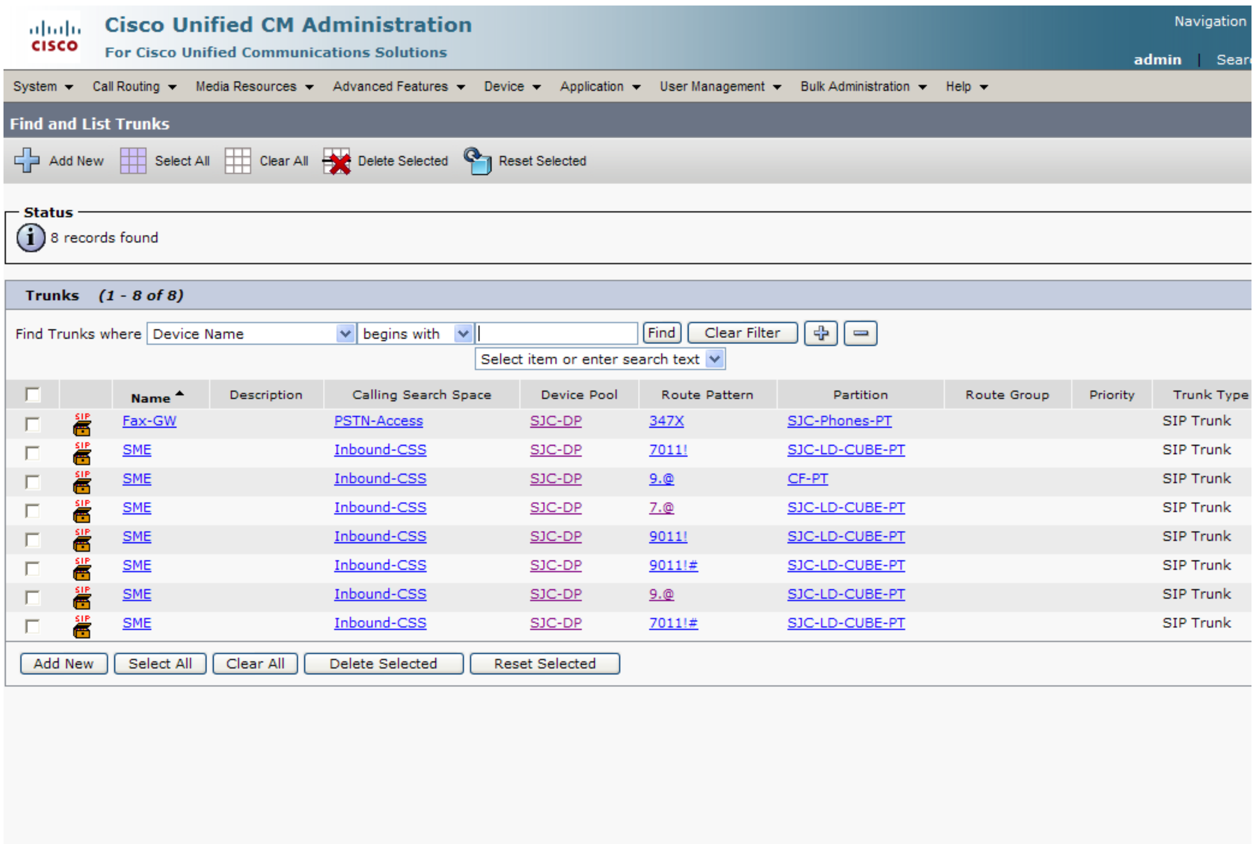

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Route Pattern Configuration
Related Links:
Save
Delete
Copy
Add New
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling Party Number Type*
Cisco CallManager
Calling Party Numbering Plan*
Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Called Party Transformations
Discard Digits
< None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Cisco CallManager
Called Party Numbering Plan*
Cisco CallManager
ISDN Network-Specific Facilities Information Element
Network Service Protocol
-- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save
Delete
Copy
Add New
*- indicates required item.

Fax Route Pattern

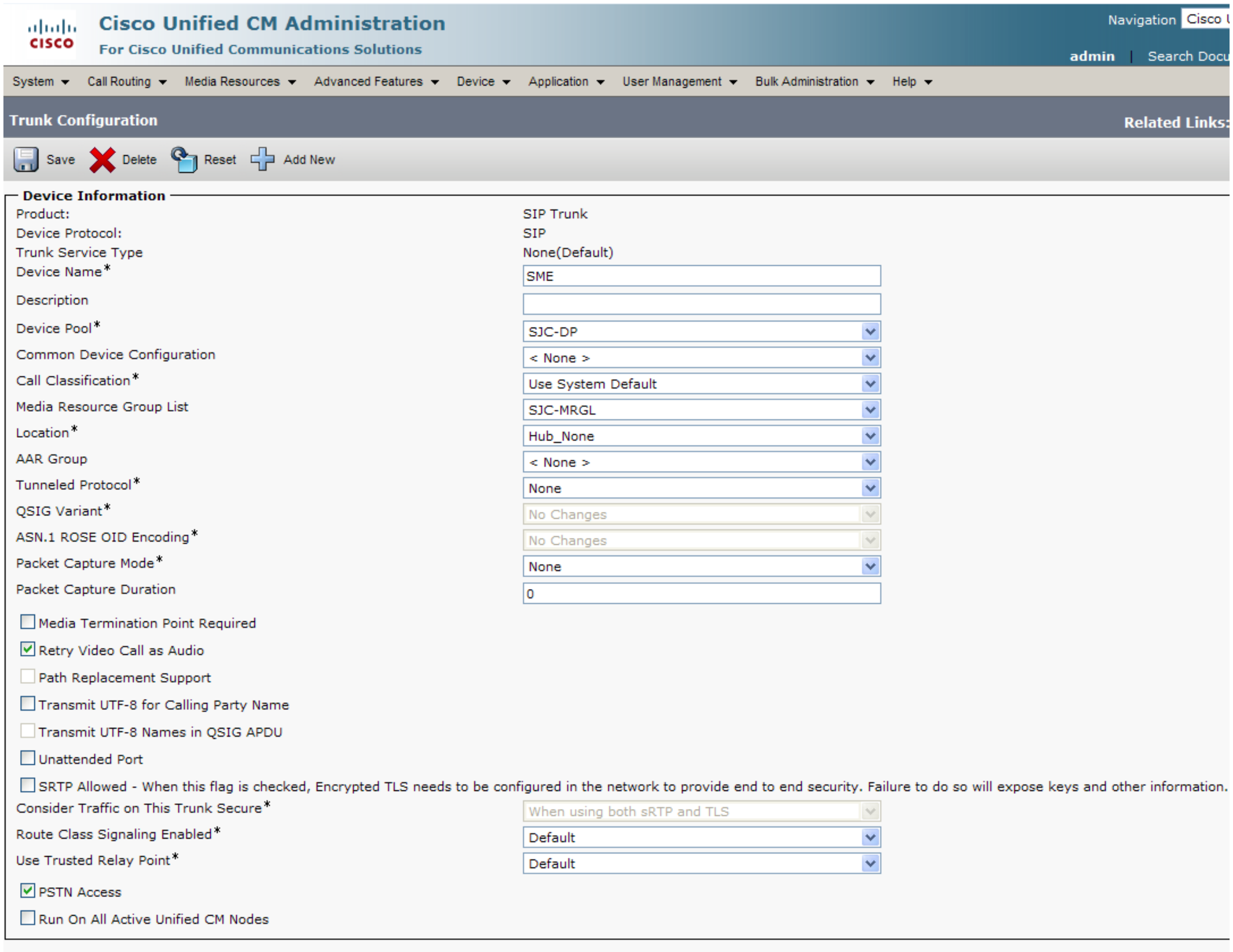


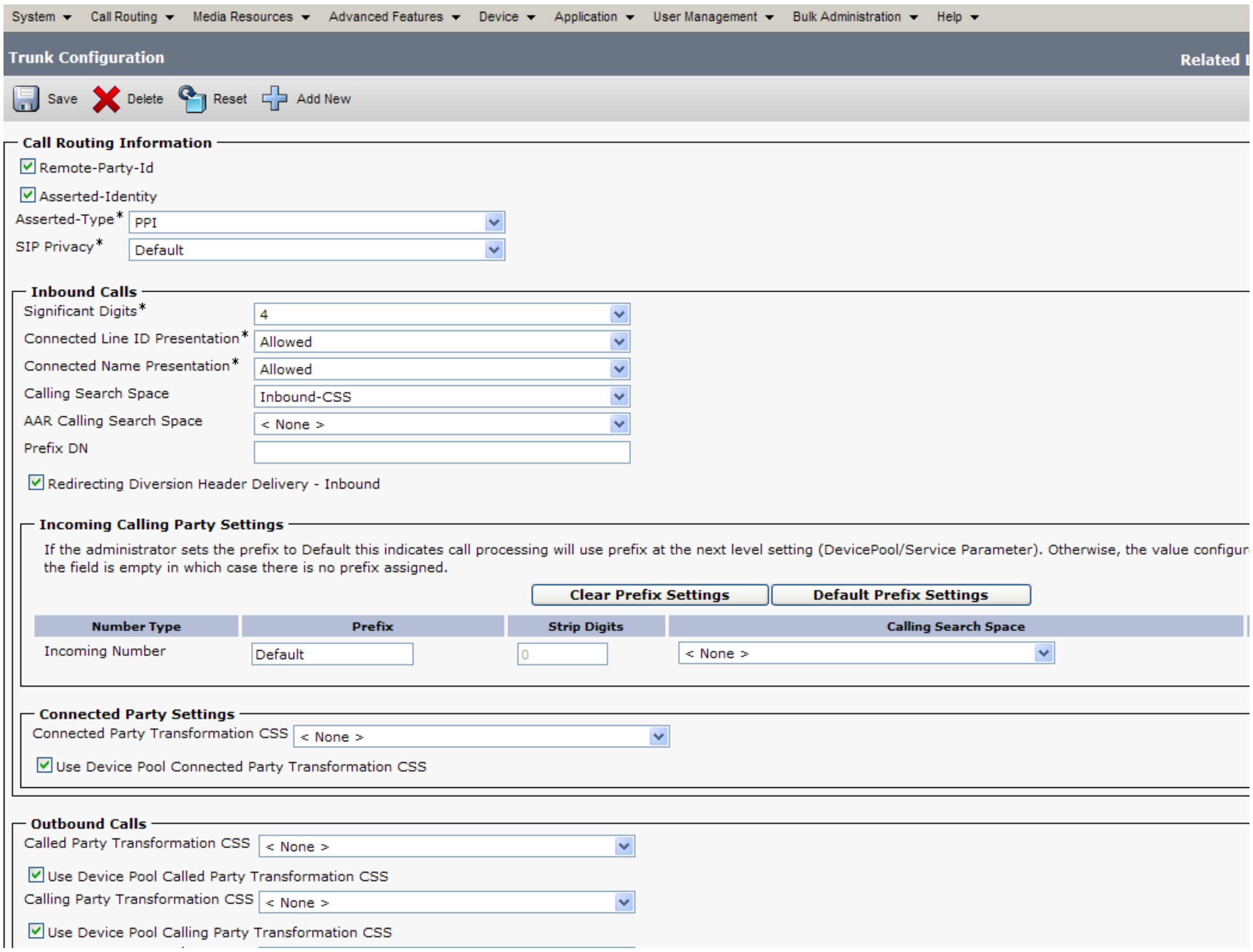

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Route Pattern Configuration
Save
Delete
Copy
Add New
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling Party Number Type*
Cisco CallManager
Calling Party Numbering Plan*
Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Called Party Transformations
Discard Digits
< None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Cisco CallManager
Called Party Numbering Plan*
Cisco CallManager
ISDN Network-Specific Facilities Information Element
Network Service Protocol
-- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save
Delete
Copy
Add New
*- indicates required item.

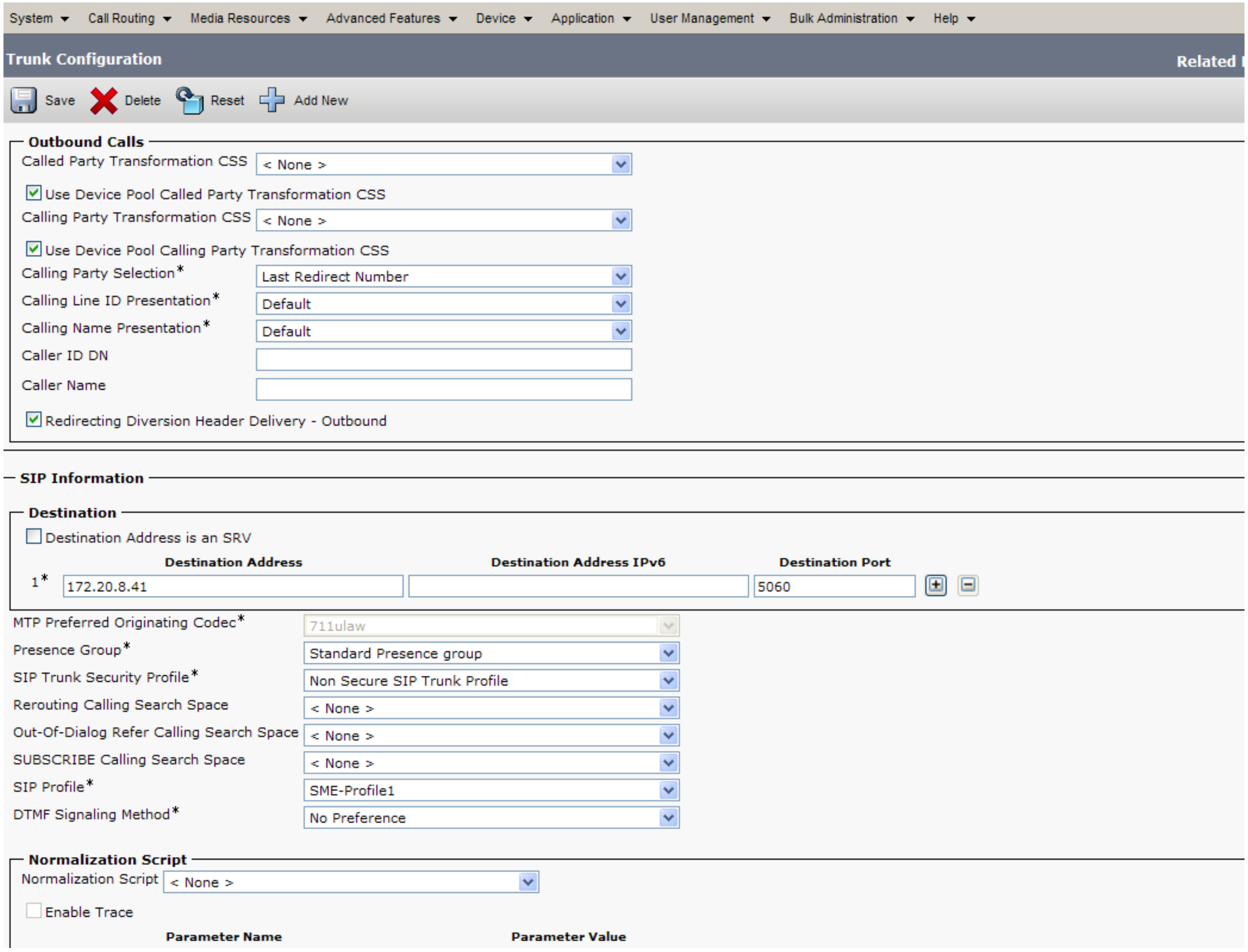
Cisco UCM SIP trunks configurations



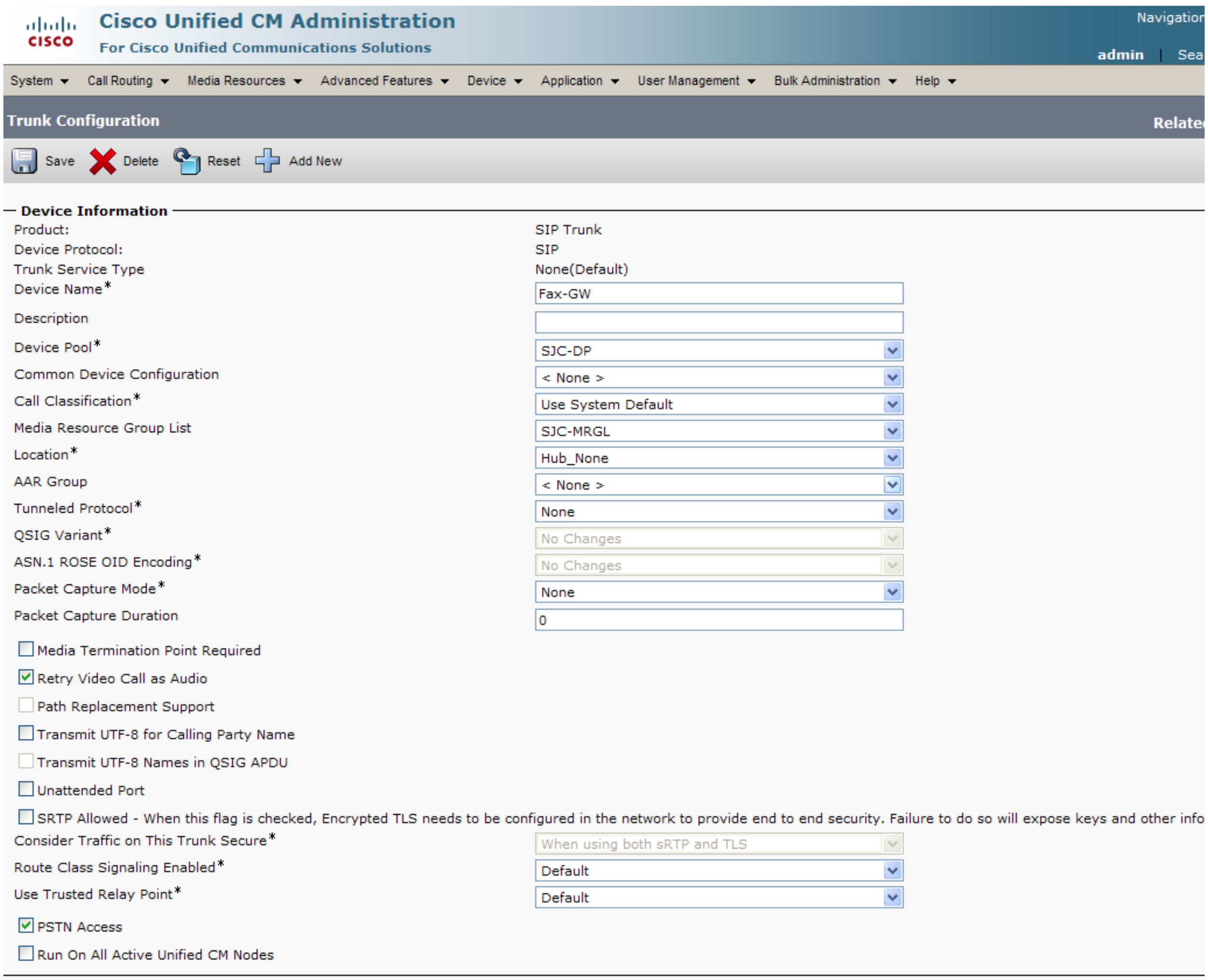
SIP Trunk Configuration for the Session Manager Edition

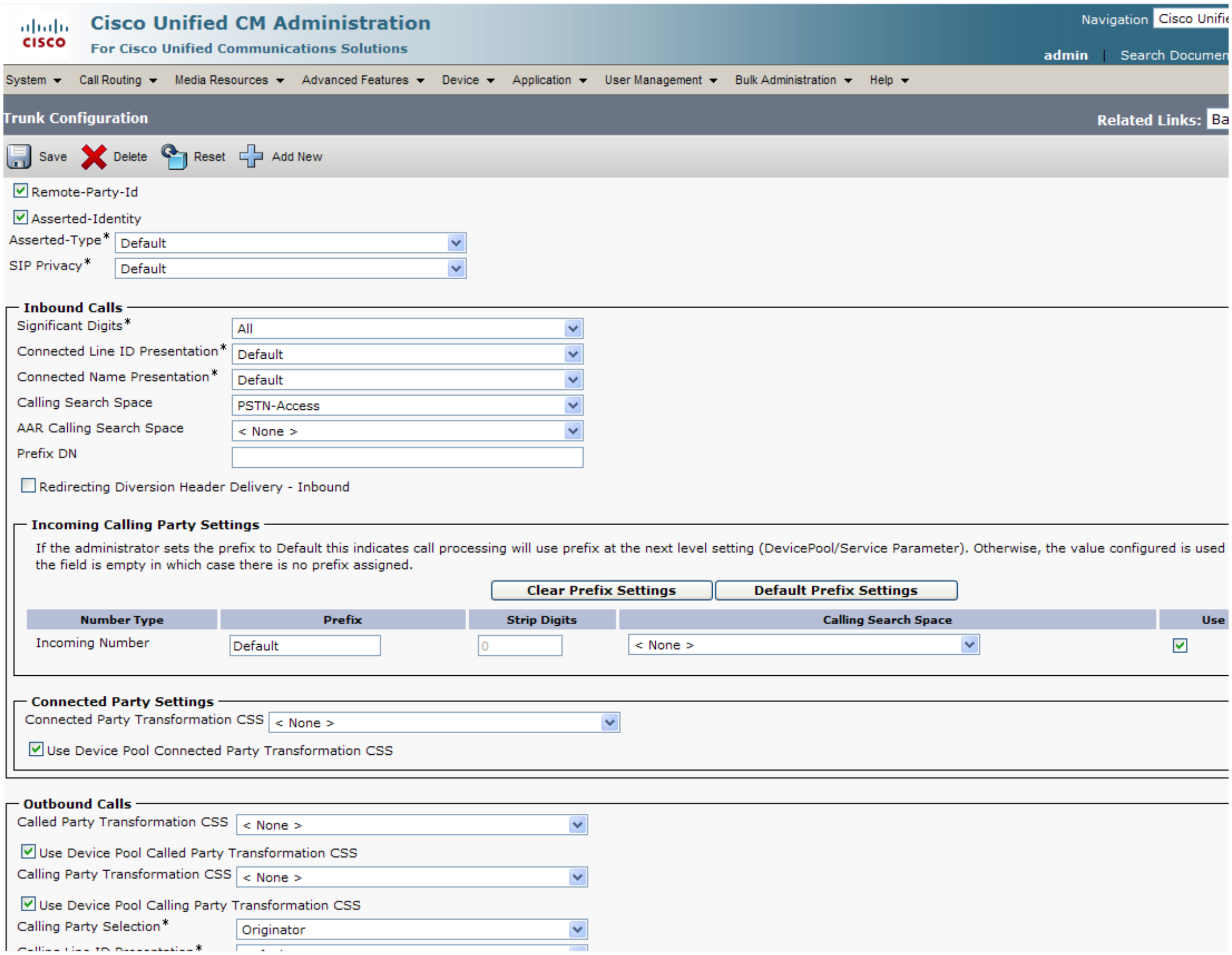



System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Trunk Configuration
Save
Delete
Reset
Add New
Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type*
PPI
SIP Privacy*
Default
Inbound Calls
Significant Digits*
4
Connected Line ID Presentation*
Allowed
Connected Name Presentation*
Allowed
Calling Search Space
Inbound-CSS
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 configur
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
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


System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Trunk Configuration
Related
Save Delete Reset Add New
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* Last Redirect Number
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Caller ID DN
Caller Name
Redirecting Diversion Header Delivery - Outbound
SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1* 172.20.8.41 5060
MTP Preferred Originating Codec* 711ulaw
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* SME-Profile1
DTMF Signaling Method* No Preference
Normalization Script
Normalization Script < None >
Enable Trace
Parameter Name
Parameter Value

SIP Trunk Configuration for the Fax Gateway





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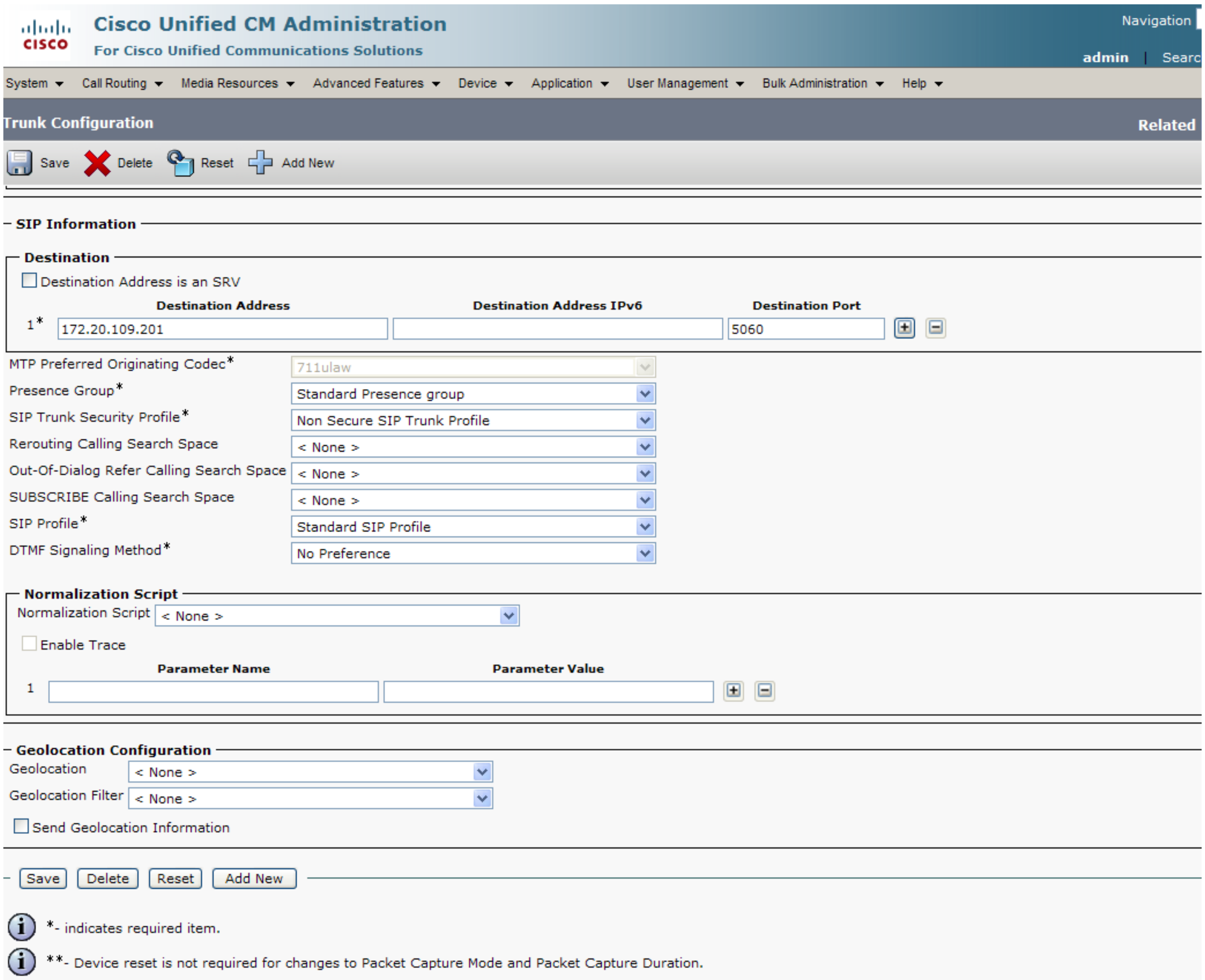
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Trunk Configuration
Related Links: Ba
Save Delete Reset Add New
Remote-Party-Id
Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default
Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space PSTN-Access
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 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
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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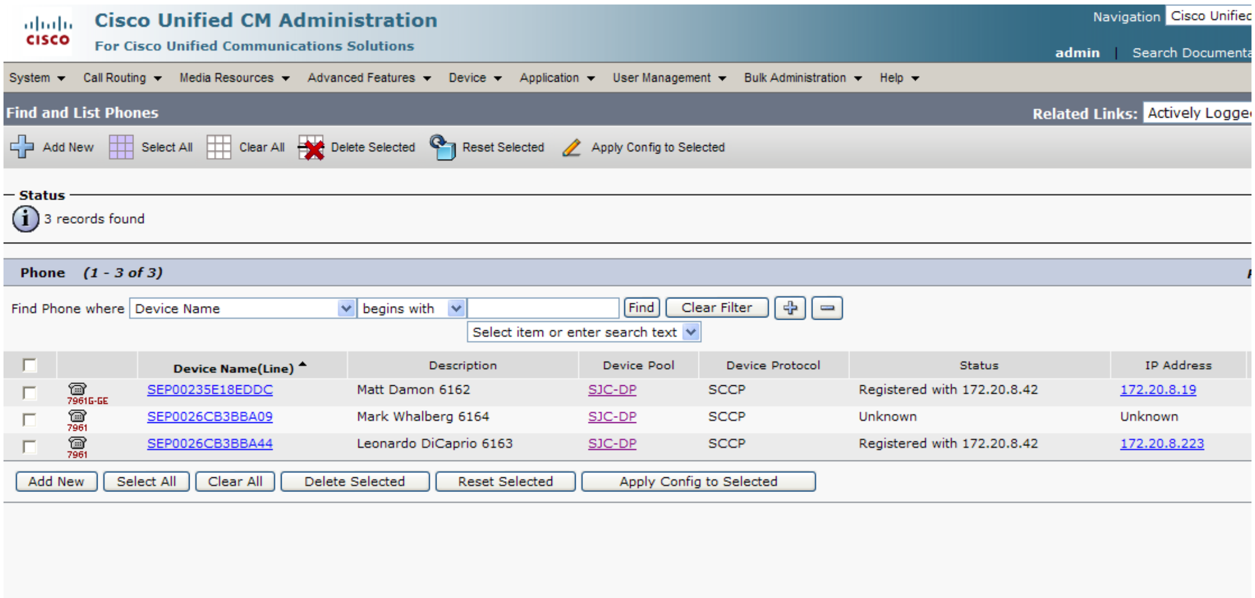
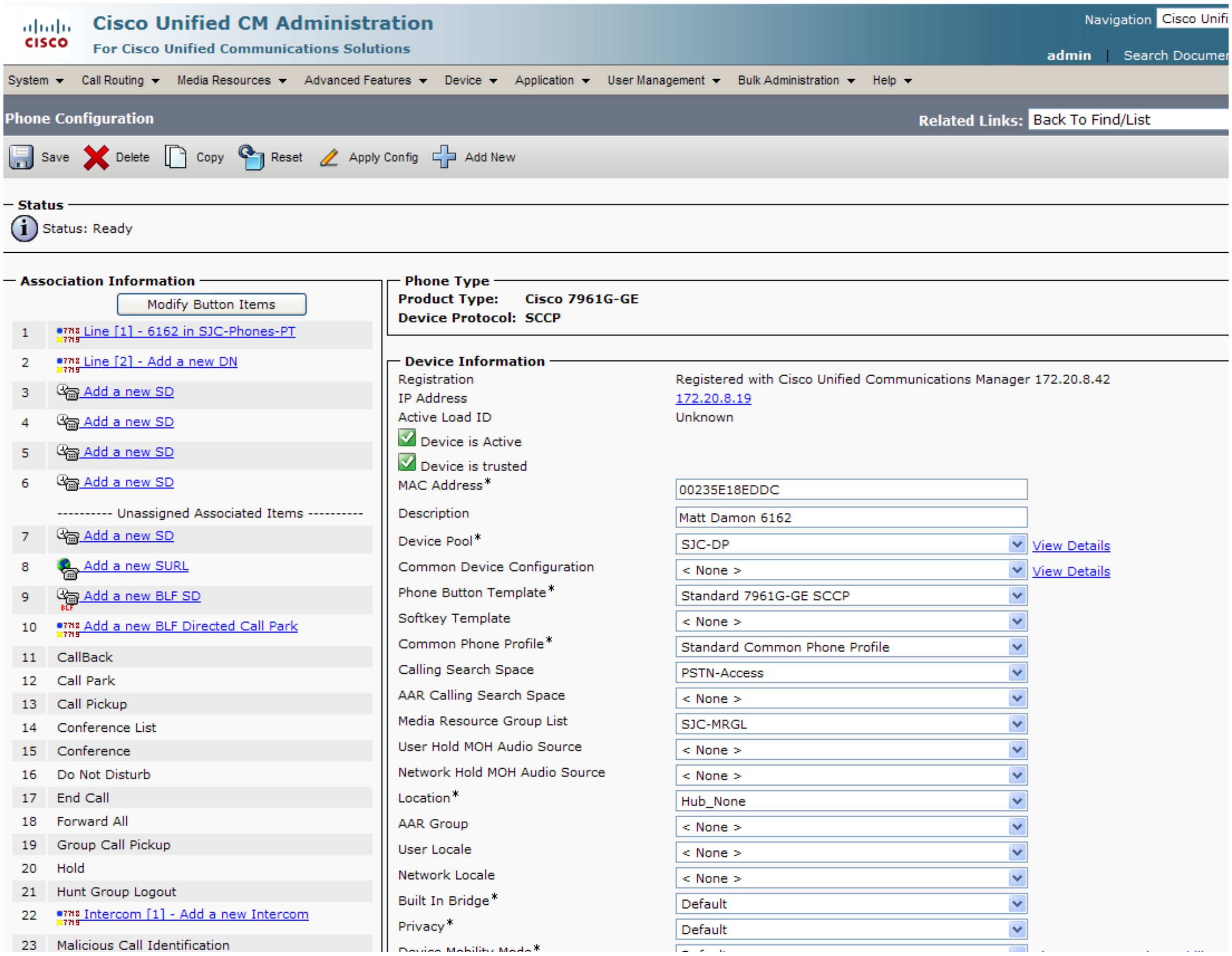
Trunk Configuration
Related
Save Delete Reset Add New
SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1* 172.20.109.201 5060
MTP Preferred Originating Codec* 711ulaw
Presence Group* Standard Presence group
SIP Trunk Security Profile* 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
DTMF Signaling Method* No Preference
Normalization Script
Normalization Script < None >
Enable Trace
Parameter Name
Parameter Value
1
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.

SCCP Phone configurations on the Cisco UCM

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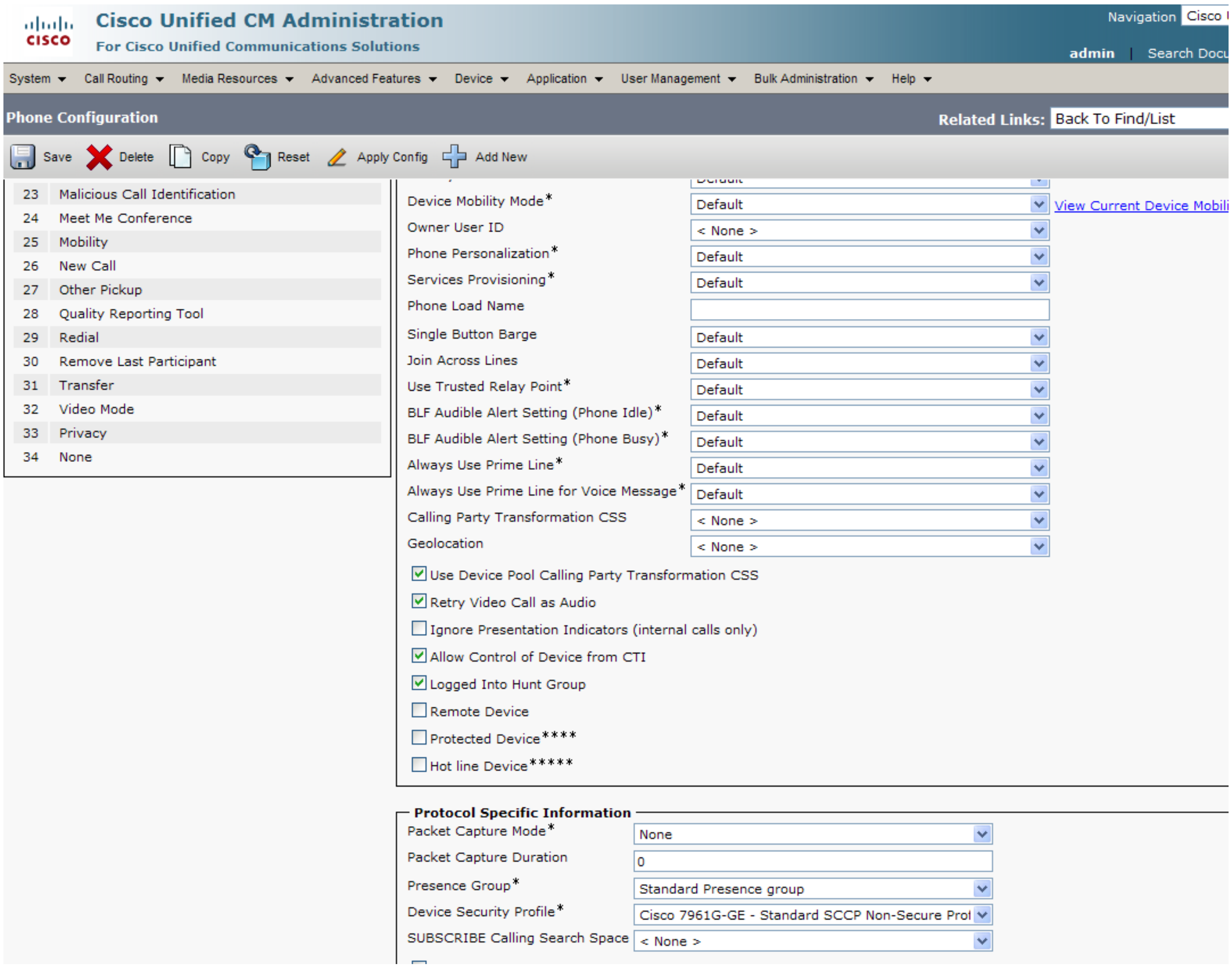
Add New
Status
Status: Ready
Association Information
Modify Button Items
1 Line [1] - 6162 in SJC-Phones-PT
2 Line [2] - Add a new DN
3 Add a new SD
4 Add a new SD
5 Add a new SD
6 Add a new SD
---------- Unassigned Associated Items ----------
7 Add a new SD
8 Add a new SURL
9 Add a new BLF SD
10 Add a new BLF Directed Call Park
11 CallBack
12 Call Park
13 Call Pickup
14 Conference List
15 Conference
16 Do Not Disturb
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
Phone Type
Product Type: Cisco 7961G-GE
Device Protocol: SCCP
Device Information
Registration Registered with Cisco Unified Communications Manager 172.20.8.42
IP Address 172.20.8.19
Active Load ID Unknown
Device is Active
Device is trusted
MAC Address* 00235E18EDDC
Description Matt Damon 6162
Device Pool* SJC-DP View Details
Common Device Configuration < None > View Details
Phone Button Template* Standard 7961G-GE SCCP
Softkey Template < None >
Common Phone Profile* Standard Common Phone Profile
Calling Search Space PSTN-Access
AAR Calling Search Space < None >
Media Resource Group List SJC-MRGL
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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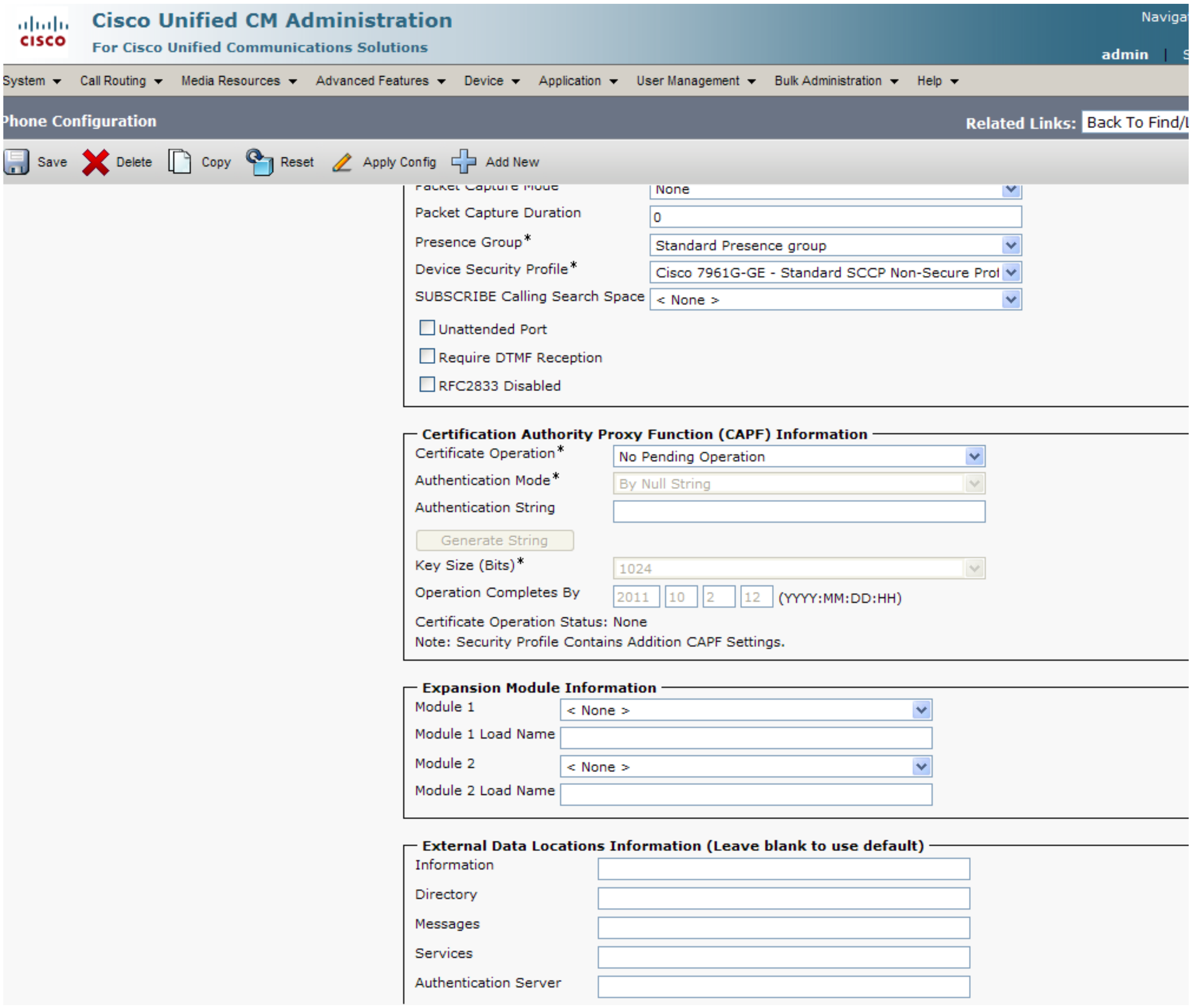
Add New
23 Malicious Call Identification
24 Meet Me Conference
25 Mobility
26 New Call
27 Other Pickup
28 Quality Reporting Tool
29 Redial
30 Remove Last Participant
31 Transfer
32 Video Mode
33 Privacy
34 None
Device Mobility Mode* Default
View Current Device Mobili
Owner User ID < None >
Phone Personalization* 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
Calling Party Transformation CSS < None >
Geolocation < None >
Use Device Pool Calling Party Transformation CSS
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*****
Protocol Specific Information
Packet Capture Mode* None
Packet Capture Duration 0
Presence Group* Standard Presence group
Device Security Profile* Cisco 7961G-GE - Standard SCCP Non-Secure Prol
SUBSCRIBE Calling Search Space < None >



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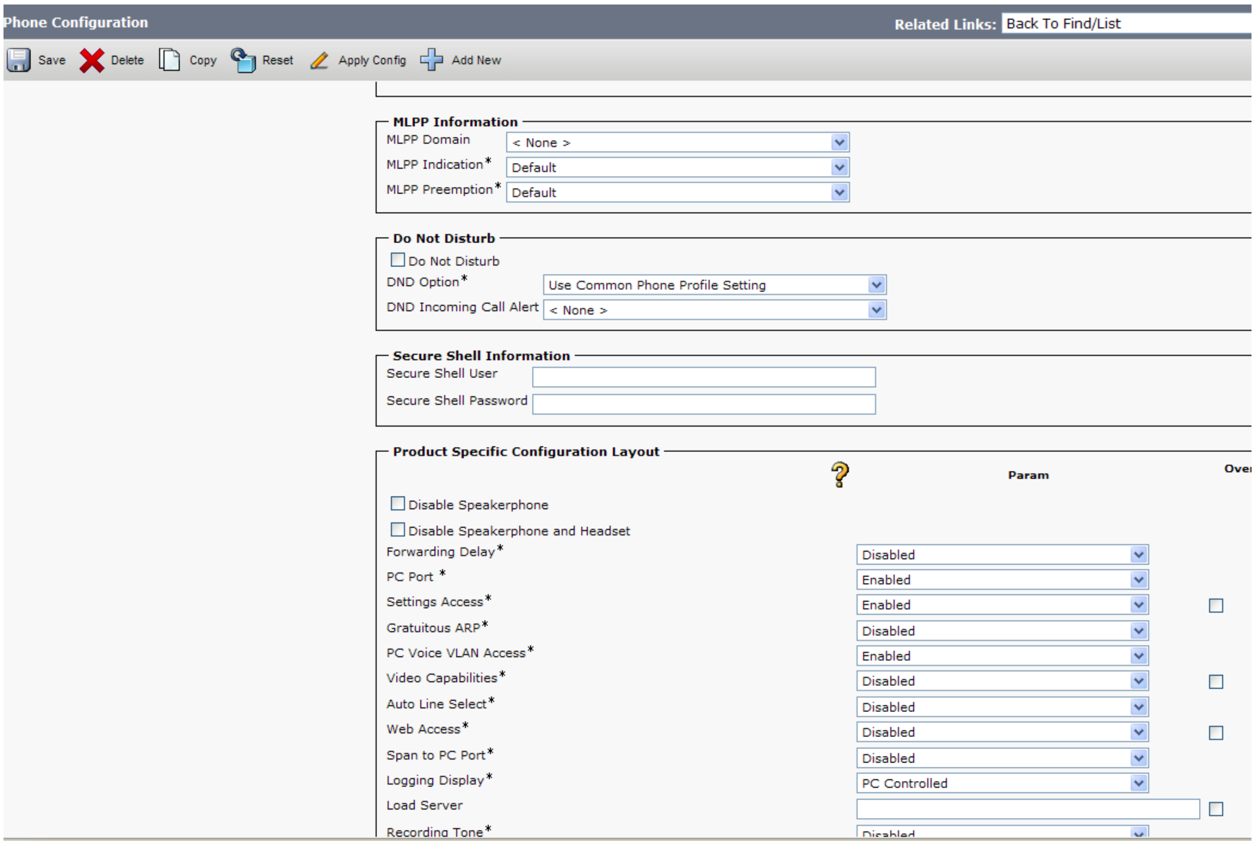
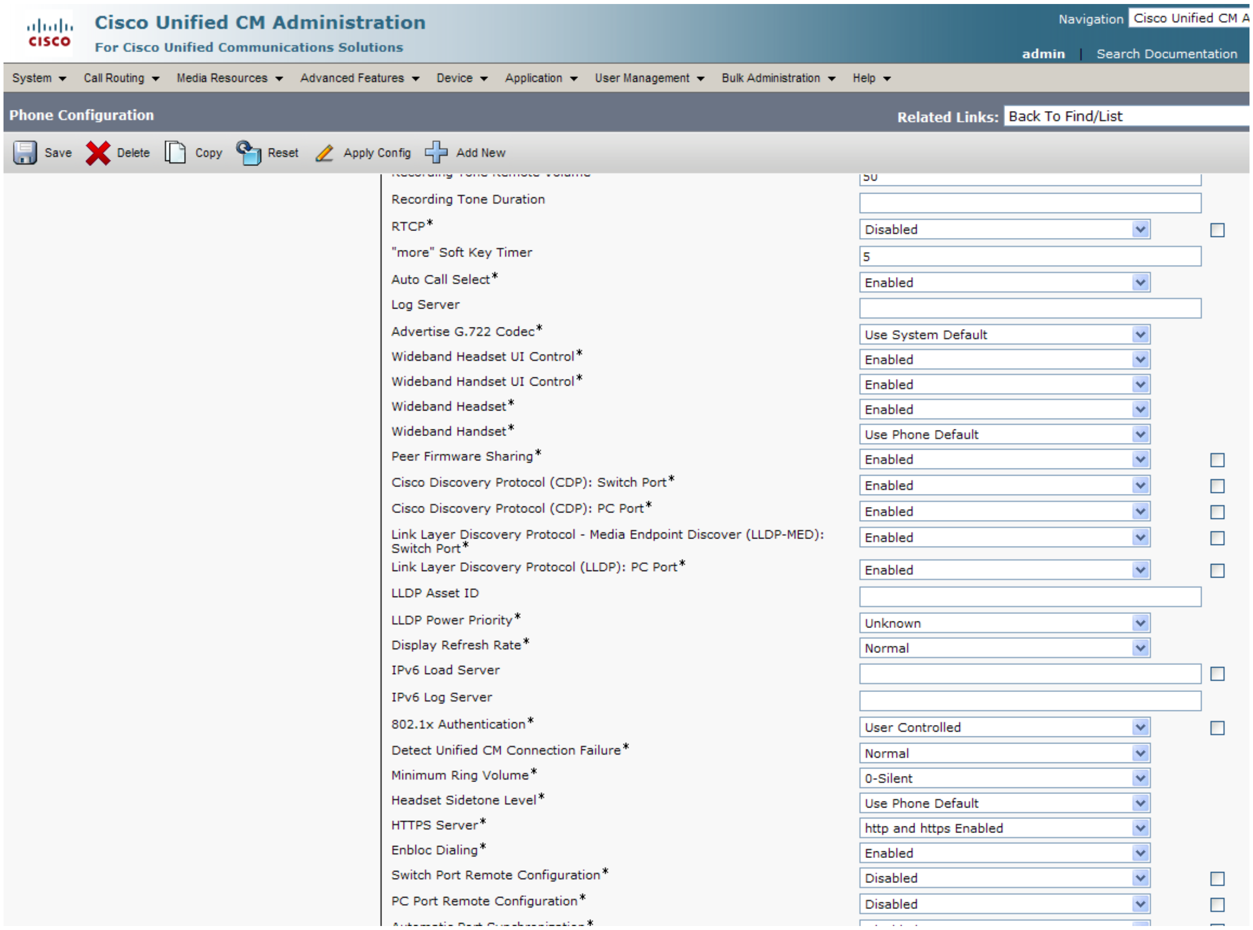
Save
Delete
Copy
Reset
Apply Config
Add New
Packet Capture Duration 0
Presence Group* Standard Presence group
Device Security Profile* Cisco 7961G-GE - Standard SCCP Non-Secure Prof
SUBSCRIBE Calling Search Space < None >
Unattended Port
Require DTMF Reception
RFC2833 Disabled
Certification Authority 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 2011 10 2 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


Phone Configuration
Related Links: Back To Find/List
Save
Delete
Copy
Reset
Apply Config
Add New
MLPP Information
MLPP Domain
< None >
MLPP Indication*
Default
MLPP Preemption*
Default
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
Param
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
Span to PC Port*
Disabled
Logging Display*
PC Controlled
Load Server
Recording Tone*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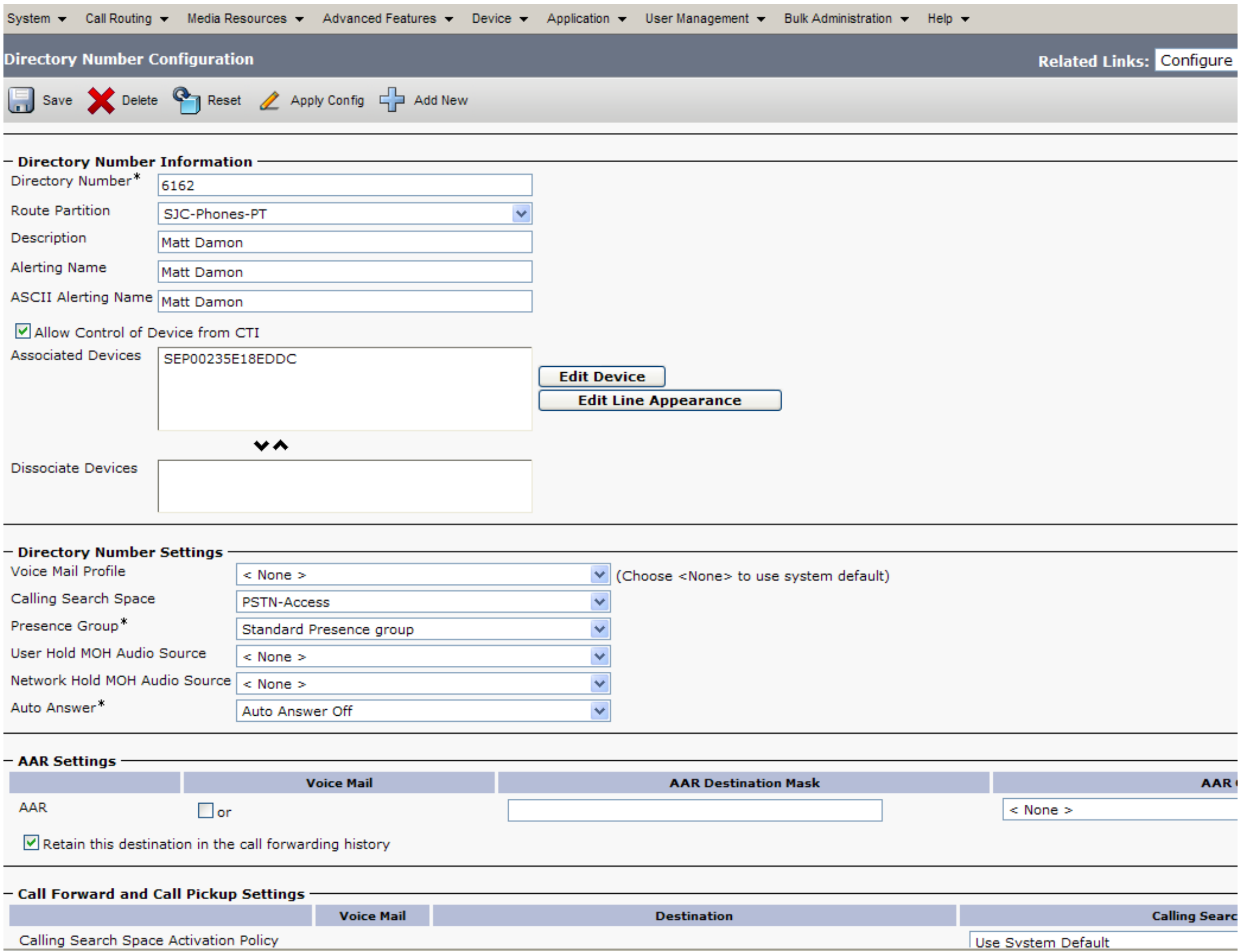
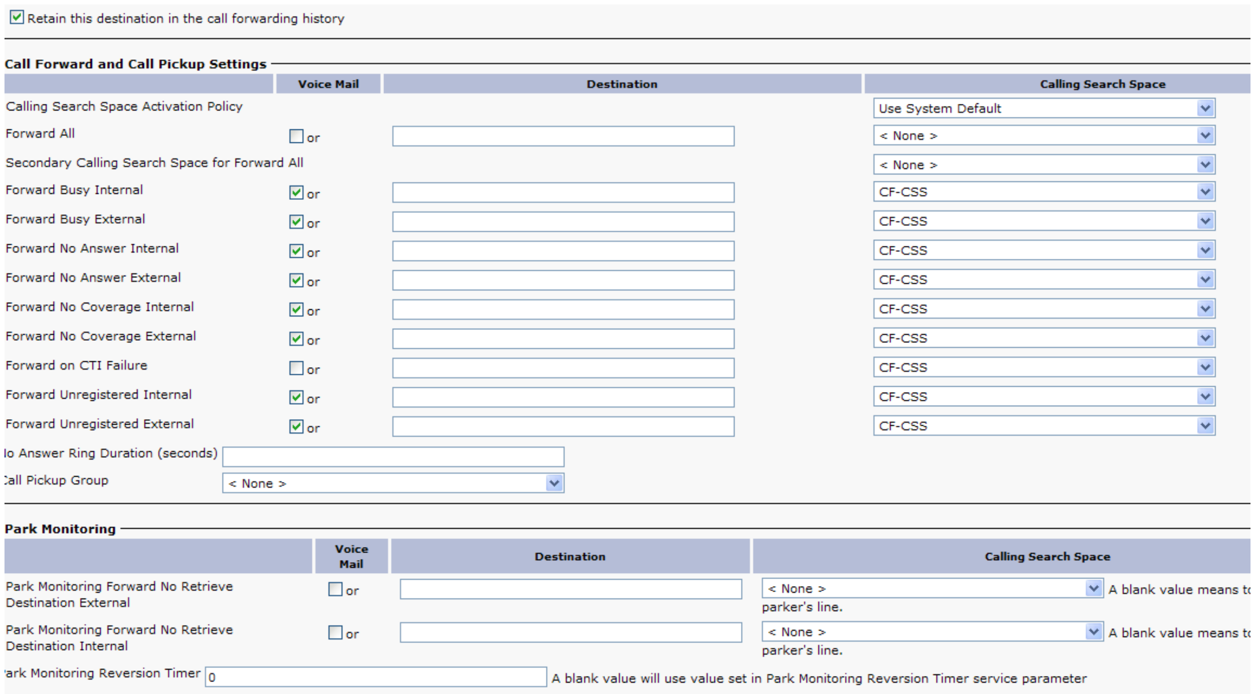
Related Links: Back To Find/List
Save Delete Copy Reset Apply Config Add New
Recording Tone Duration
RTCP* Disabled
"more" Soft Key Timer 5
Auto Call Select* Enabled
Log Server
Advertise G.722 Codec* Use System Default
Wideband Headset UI Control* Enabled
Wideband Handset UI Control* Enabled
Wideband Headset* Enabled
Wideband Handset* Use Phone Default
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
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* Use Phone Default
HTTPS Server* http and https Enabled
Enbloc Dialing* Enabled
Switch Port Remote Configuration* Disabled
PC Port Remote Configuration* Disabled

DN Configuration for SCCP Phone




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		CF-CSS
Forward Busy External	or		CF-CSS
Forward No Answer Internal	or		CF-CSS
Forward No Answer External	or		CF-CSS
Forward No Coverage Internal	or		CF-CSS
Forward No Coverage External	or		CF-CSS
Forward on CTI Failure	or		CF-CSS
Forward Unregistered Internal	or		CF-CSS
Forward Unregistered External	or		CF-CSS

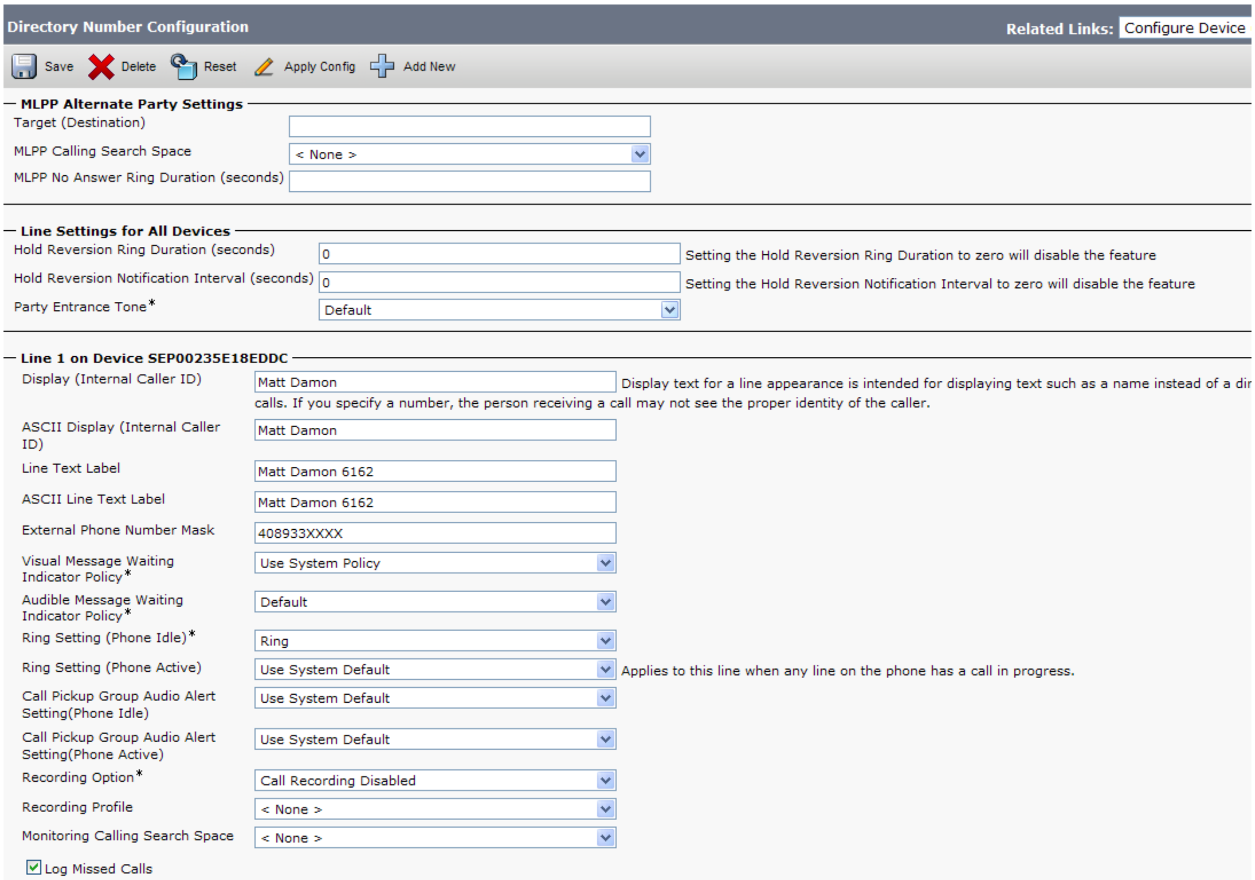
lo Answer Ring Duration (seconds)

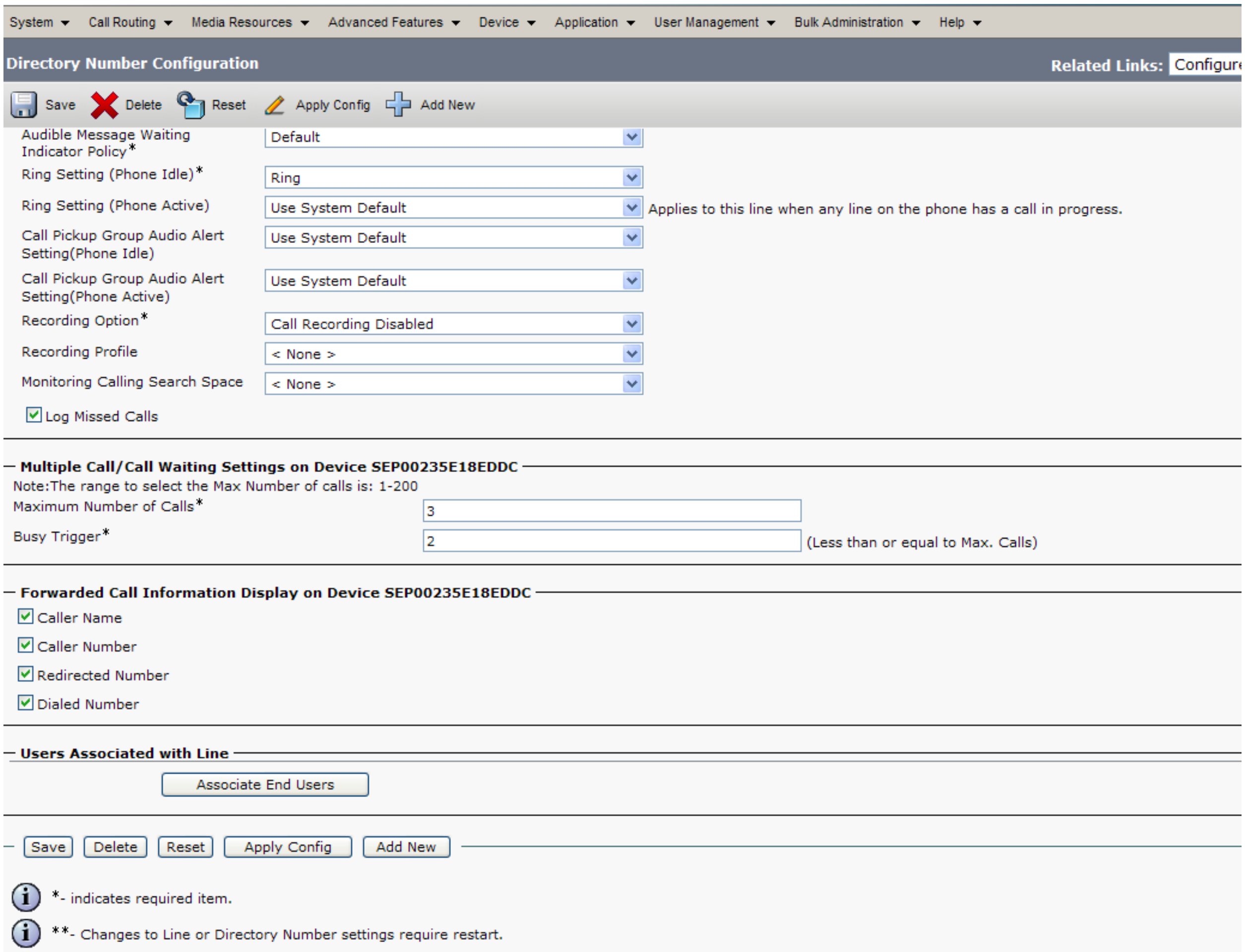
Call Pickup Group < None >

Park Monitoring

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

'ark Monitoring Reversion Timer 0 A blank value will use value set in Park Monitoring Reversion Timer service parameter


Directory Number Configuration
Related Links: Configure Device
Save
Delete
Reset
Apply Config
Add New
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space
< None >
MLPP No Answer Ring Duration (seconds)
Line Settings for All Devices
Hold Reversion Ring Duration (seconds)
0
Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)
0
Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*
Default
Line 1 on Device SEP00235E18EDDC
Display (Internal Caller ID)
Matt Damon
Display text for a line appearance is intended for displaying text such as a name instead of a dir
calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)
Matt Damon
Line Text Label
Matt Damon 6162
ASCII Line Text Label
Matt Damon 6162
External Phone Number Mask
408933XXXX
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 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 >
Monitoring Calling Search Space
< None >
Log Missed Calls



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Directory Number Configuration
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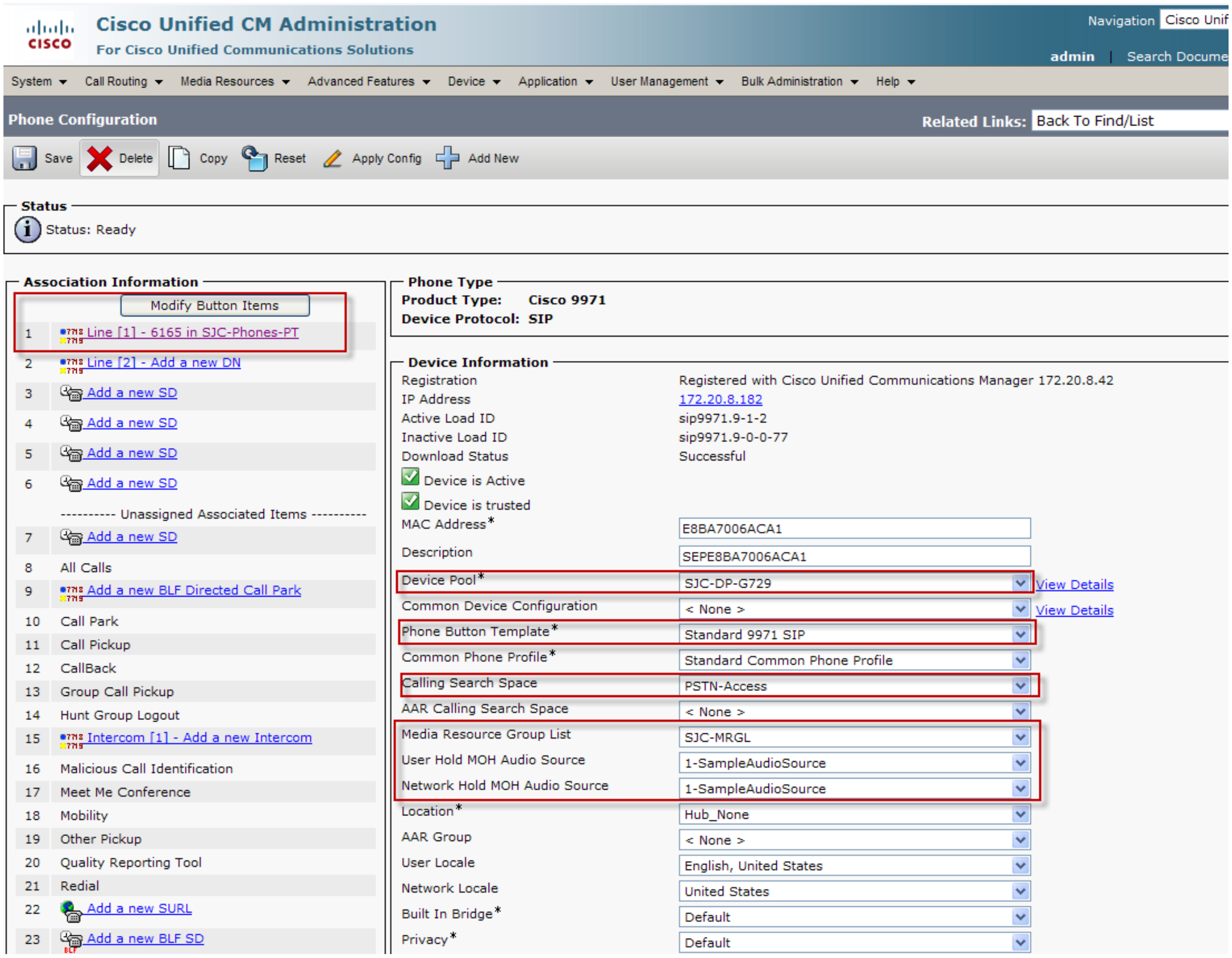
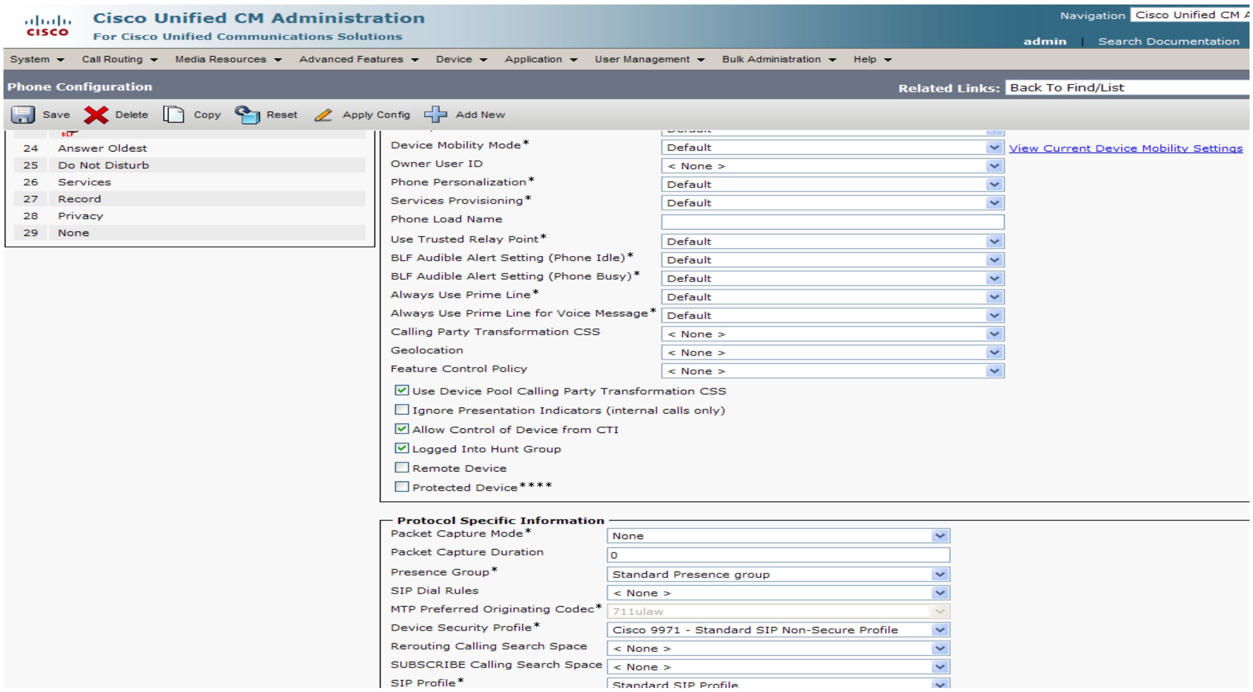
Delete
Reset
Apply Config
Add New
Audible Message Waiting Indicator Policy*
Default
Ring Setting (Phone Idle)*
Ring
Ring Setting (Phone Active)
Use System Default
Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)
Use System Default
Call Pickup Group Audio Alert Setting(Phone Active)
Use System Default
Recording Option*
Call Recording Disabled
Recording Profile
< None >
Monitoring Calling Search Space
< None >
Log Missed Calls
Multiple Call/Call Waiting Settings on Device SEP00235E18EDDC
Note:The range to select the Max Number of calls is: 1-200
Maximum Number of Calls*
3
Busy Trigger*
2
(Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP00235E18EDDC
Caller Name
Caller Number
Redirected Number
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.

SIP Phone configurations on the Cisco UCM

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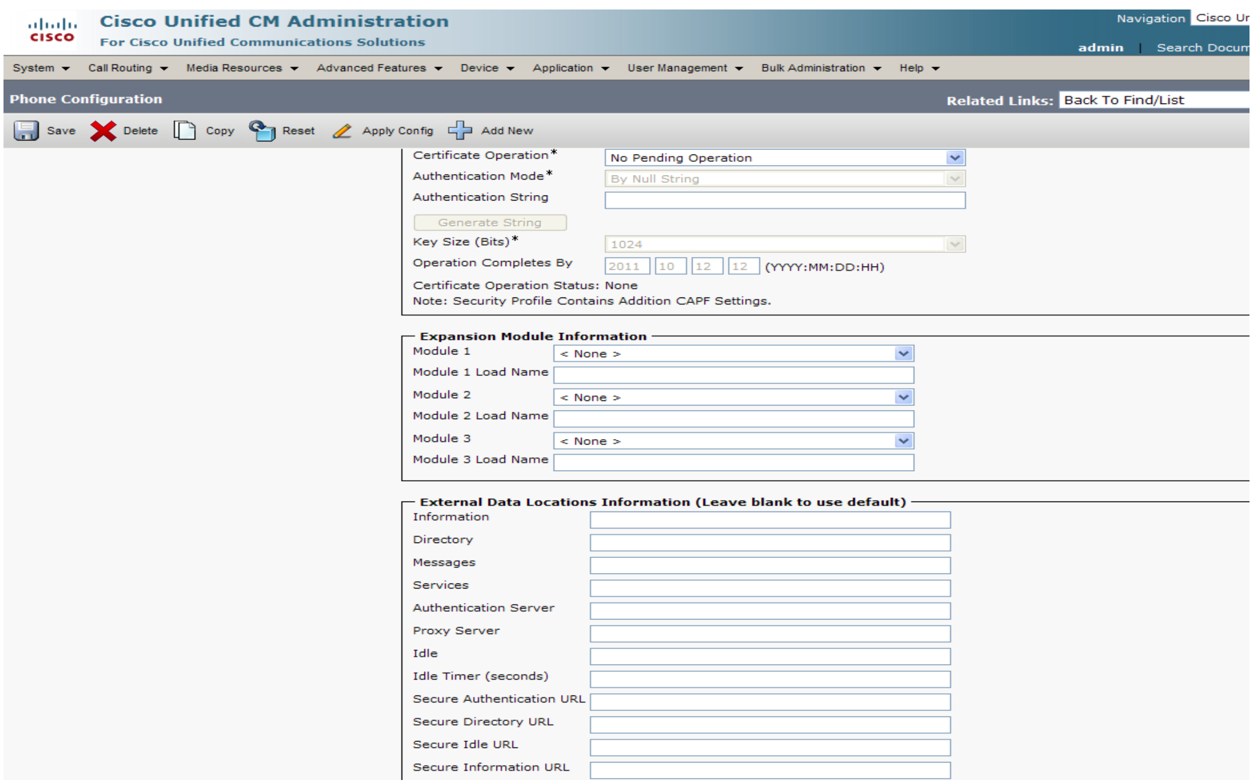
Phone Configuration
Related Links: Back To Find/List
Save Delete Copy Reset Apply Config Add New
24 Answer Oldest
25 Do Not Disturb
26 Services
27 Record
28 Privacy
29 None
Device Mobility Mode* Default
View Current Device Mobility Settings
Owner User ID < None >
Phone Personalization* Default
Services Provisioning* Default
Phone Load Name
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
Calling Party Transformation CSS < None >
Geolocation < None >
Feature Control Policy < None >
Use Device Pool Calling Party Transformation CSS
Ignore Presentation Indicators (internal calls only)
Allow Control of Device from CTI
Logged Into Hunt Group
Remote Device
Protected Device****
Protocol Specific Information
Packet Capture Mode* None
Packet Capture Duration 0
Presence Group* Standard Presence group
SIP Dial Rules < None >
MTP Preferred Originating Codec* 711ulaw
Device Security Profile* Cisco 9971 - Standard SIP Non-Secure Profile
Rerouting Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* Standard SIP Profile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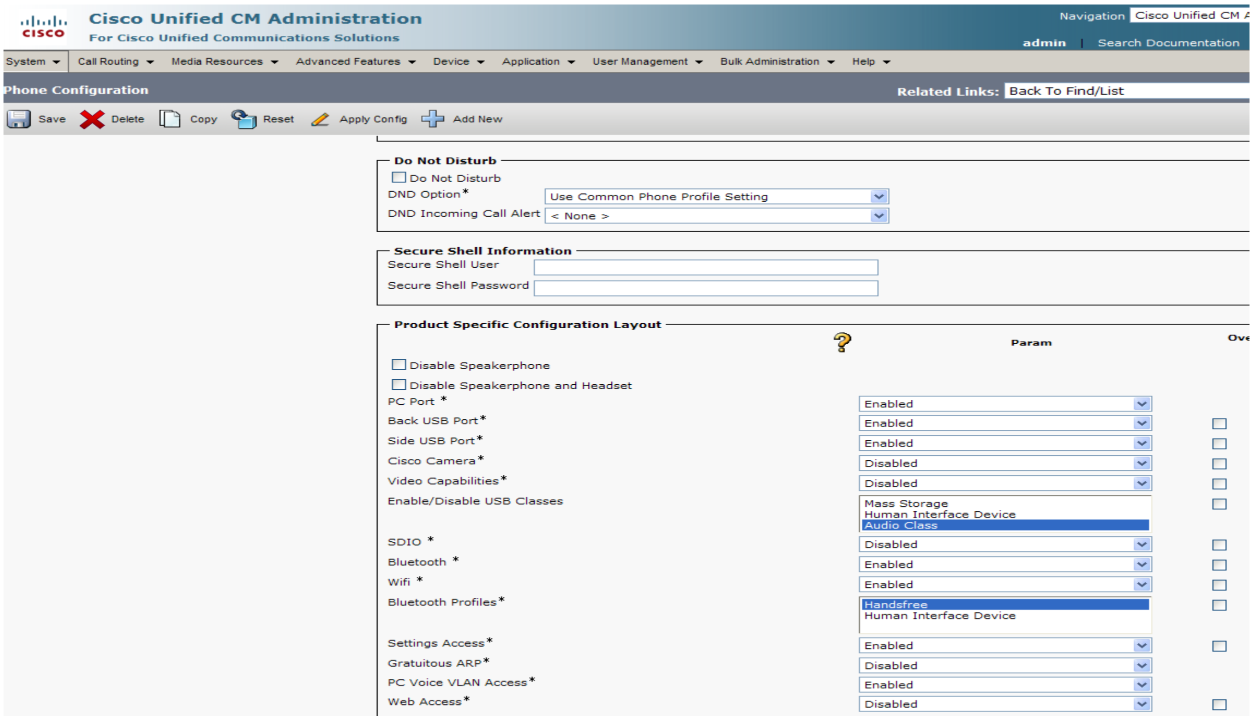
Phone Configuration
Related Links: Back To Find/List
Save Delete Copy Reset Apply Config Add New
Certificate Operation* No Pending Operation
Authentication Mode* By Null String
Authentication String
Generate String
Key Size (Bits)* 1024
Operation Completes By 2011 10 12 12 (YYYY:MM:DD:HH)
Certificate 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
Module 3 < None >
Module 3 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



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Save
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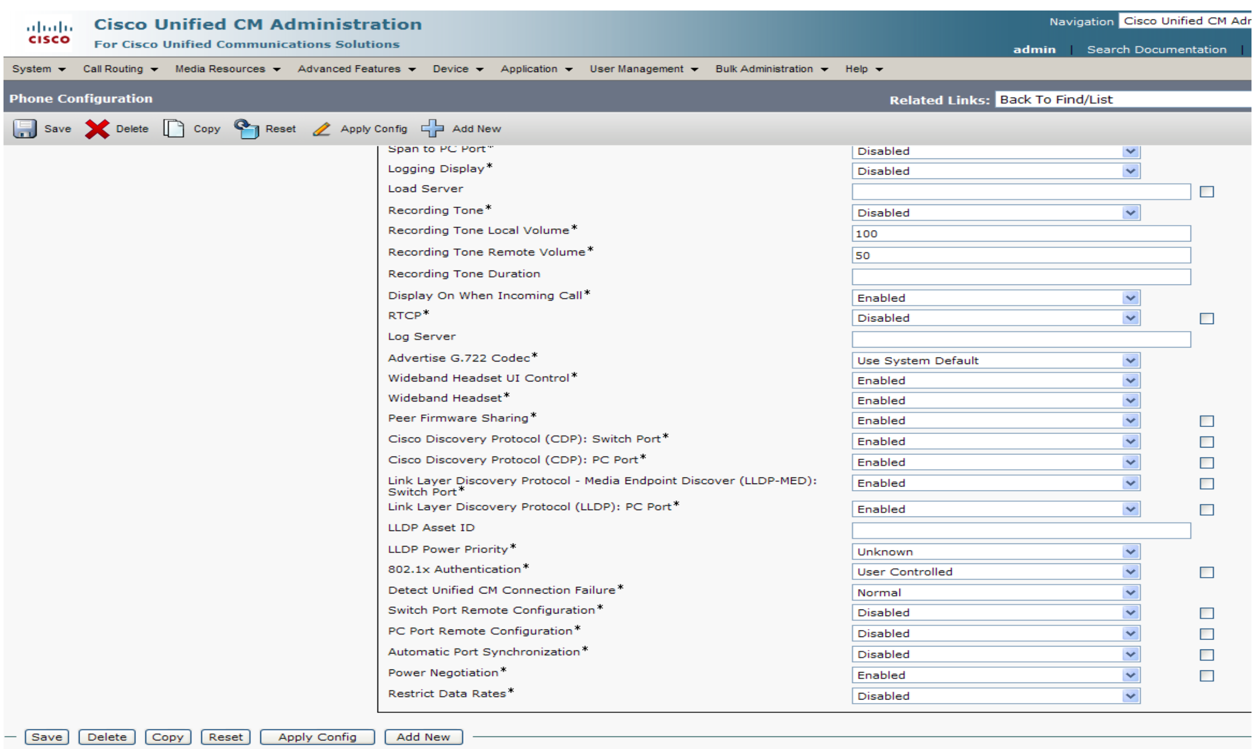
Copy
Reset
Apply Config
Add New
Do Not Disturb
Do Not Disturb
DND Option*
Use Common Phone Profile Setting
DND Incoming Call Alert
< None >
Secure Shell Information
Secure Shell User
Secure Shell Password
Product Specific Configuration Layout
Param
Disable Speakerphone
Disable Speakerphone and Headset
PC Port * Enabled
Back USB Port* Enabled
Side USB Port* Enabled
Cisco Camera* Disabled
Video Capabilities* Disabled
Enable/Disable USB Classes
Mass Storage
Human Interface Device
Audio Class
SDIO * Disabled
Bluetooth * Enabled
Wifi * Enabled
Bluetooth Profiles*
Handsfree
Human Interface Device
Settings Access* Enabled
Gratuitous ARP* Disabled
PC Voice VLAN Access* Enabled
Web Access* Disabled



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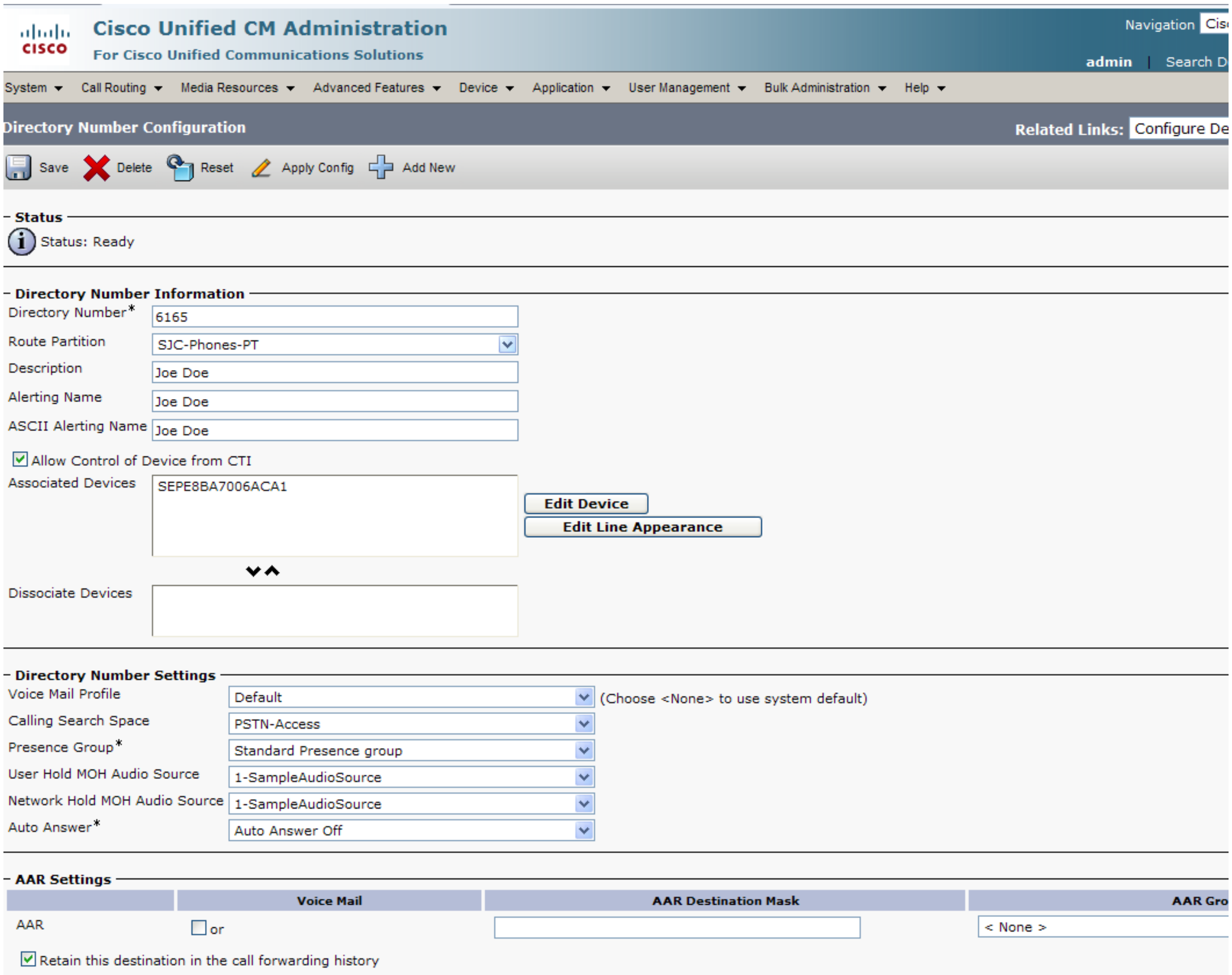
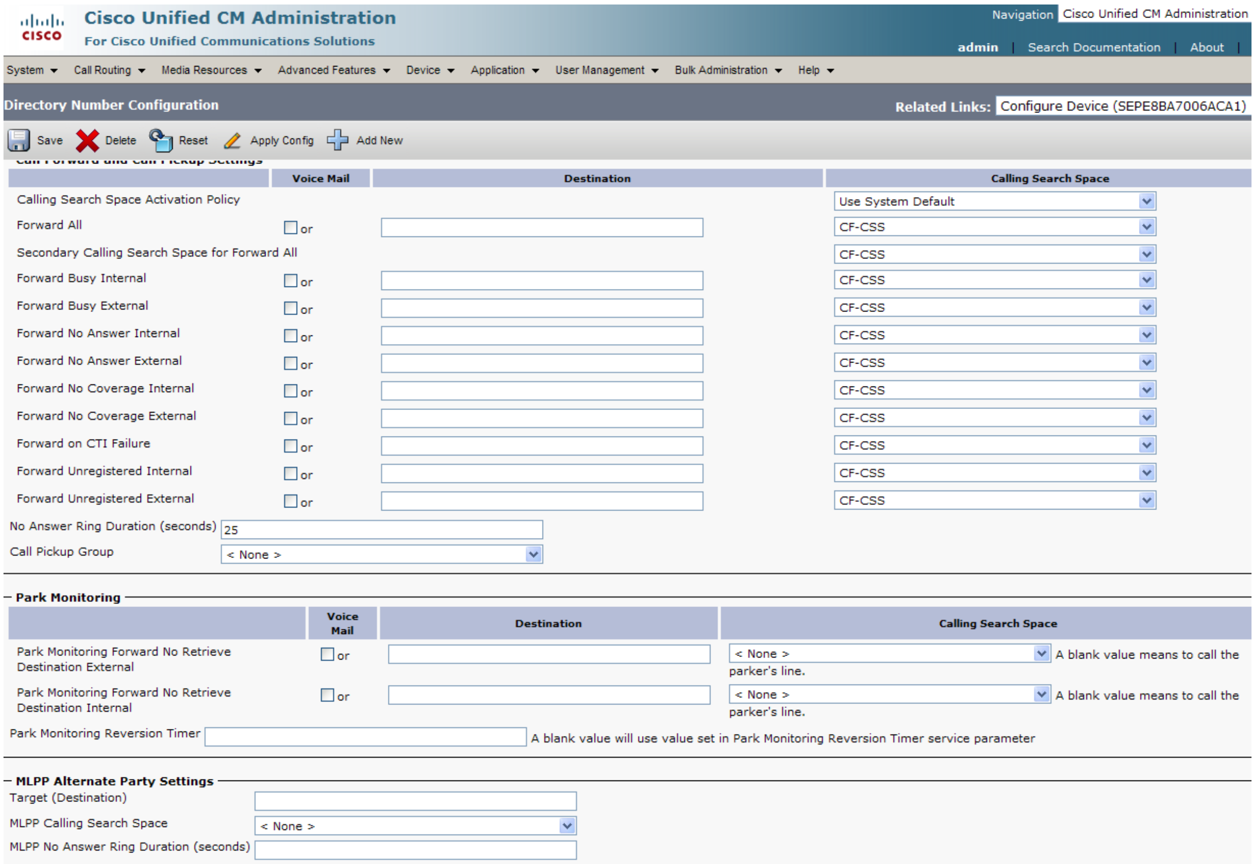
Delete
Copy
Reset
Apply Config
Add New
Span to PC Port* Disabled
Logging Display* Disabled
Load Server
Recording Tone* Disabled
Recording Tone Local Volume* 100
Recording Tone Remote Volume* 50
Recording Tone Duration
Display On When Incoming Call* Enabled
RTCP* Disabled
Log Server
Advertise G.722 Codec* 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
802.1x Authentication* User Controlled
Detect Unified CM Connection Failure* Normal
Switch Port Remote Configuration* Disabled
PC Port Remote Configuration* Disabled
Automatic Port Synchronization* Disabled
Power Negotiation* Enabled
Restrict Data Rates* Disabled
Save
Delete
Copy
Reset
Apply Config
Add New

DN Configuration for the SIP Phone

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Directory Number Configuration
Related Links: Configure Device (SEPE8BA7006ACA1)
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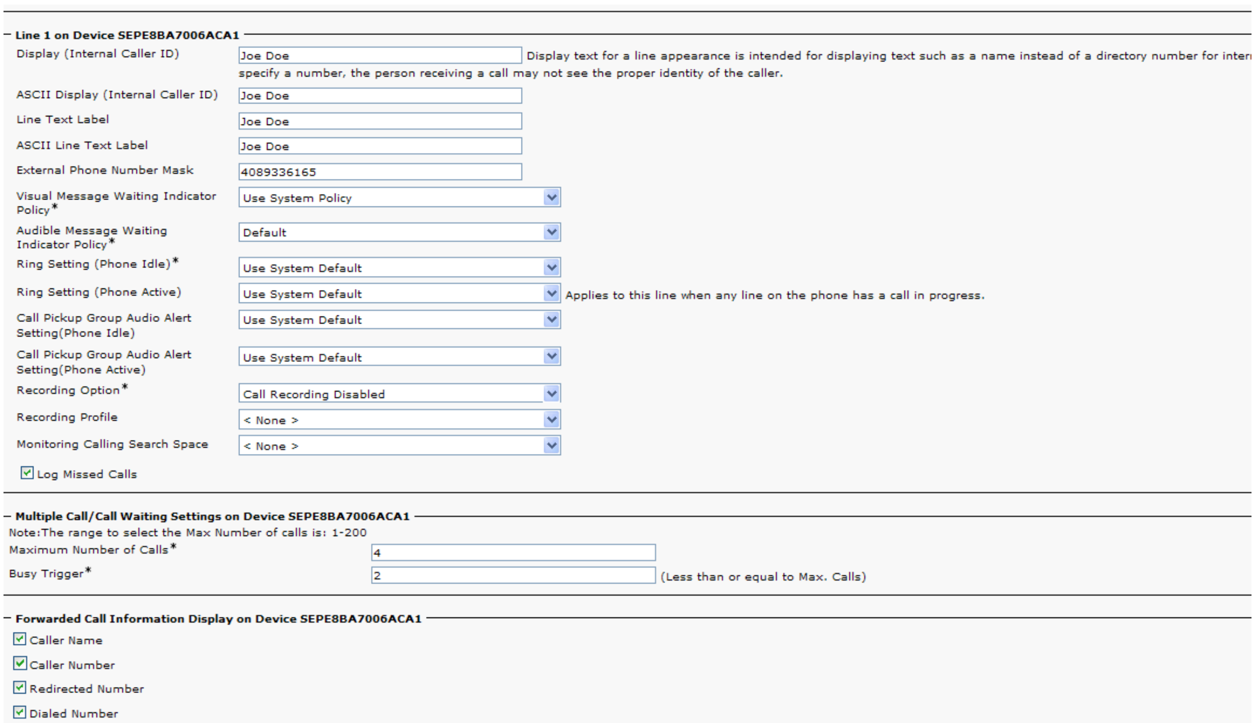
Voice Mail
Destination
Calling Search Space
Calling Search Space Activation Policy
Use System Default
Forward All
Secondary Calling Search Space for Forward All
Forward Busy Internal
Forward Busy External
Forward No Answer Internal
Forward No Answer External
Forward No Coverage Internal
Forward No Coverage External
Forward on CTI Failure
Forward Unregistered Internal
Forward Unregistered External
CF-CSS
or
No Answer Ring Duration (seconds) 25
Call Pickup Group < None >
Park Monitoring
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 Settings
Target (Destination)
MLPP Calling Search Space < None >
MLPP No Answer Ring Duration (seconds)


Line 1 on Device SEPE8BA7006ACA1
Display (Internal Caller ID)
Joe Doe
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for inter
specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)
Joe Doe
Line Text Label
Joe Doe
ASCII Line Text Label
Joe Doe
External Phone Number Mask
4089336165
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 >
Monitoring Calling Search Space
< None >
Log Missed Calls
Multiple Call/Call Waiting Settings on Device SEPE8BA7006ACA1
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 SEPE8BA7006ACA1
Caller Name
Caller Number
Redirected Number
Dialed Number

Cisco IOS Gateway Configurations

Cisco Unified Border Element Configuration on the ASR

```
CUBE-ASR1K_Vz_151#show ver
Cisco IOS Software, IOS-XE Software (PPC_LINUX_IOSD-ADVENTERPRISEK9-M), Version 15.1(2)S1, RELEASE SOFTWARE (fc2)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2011 by Cisco Systems, Inc.
Compiled Wed 01-Jun-11 04:09 by mcpre


Cisco IOS-XE software, Copyright (c) 2005-2011 by cisco Systems, Inc.
All rights reserved.  Certain components of Cisco IOS-XE software are
licensed under the GNU General Public License ("GPL") Version 2.0.  The
software code licensed under GPL Version 2.0 is free software that comes
with ABSOLUTELY NO WARRANTY.  You can redistribute and/or modify such
GPL code under the terms of GPL Version 2.0.  For more details, see the
documentation or "License Notice" file accompanying the IOS-XE software,
or the applicable URL provided on the flyer accompanying the IOS-XE
software.

ROM: IOS-XE ROMMON

CUBE-ASR1K_Vz_151 uptime is 5 weeks, 2 days, 19 hours, 48 minutes
Uptime for this control processor is 5 weeks, 2 days, 19 hours, 50 minutes
System returned to ROM by reload
System image file is "bootflash:asr1000rp1-adventerprisek9.03.03.01.S.151-2.S1.bin"
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 ASR1002 (2RU) processor with 1703423K/6147K bytes of memory.
4 Gigabit Ethernet interfaces
32768K bytes of non-volatile configuration memory.
4194304K bytes of physical memory.
7798783K bytes of eUSB flash at bootflash:.

Configuration register is 0x2102

CUBE-ASR1K_Vz_151#show run
Building configuration...

Current configuration : 8863 bytes
!
! Last configuration change at 12:34:31 UTC Fri Sep 9 2011
!
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CUBE-ASR1K_Vz_151
!
boot-start-marker
boot system bootflash:asr1000rp1-adventerprisek9.03.03.01.S.151-2.S1.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging buffered 20000000
logging console errors
enable password cisco
!
no aaa new-model
!
ipc zone default
 association 1
  no shutdown
!
ip source-route
!
!
!
ip domain name ciscolab.globalipcom.com
ip name-server 172.30.218.36
!
!
!
```

```
!
!
multilink bundle-name authenticated
!
!
!
voice service voip
 allow-connections sip to sip
 no supplementary-service sip refer
 redirect ip2ip
 sip
  header-passing error-passthru
  asserted-id pai
  localhost dns:gw1.ciscolab.globalipcom.com
   early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
  g729 annexb-all
  no call service stop
!
voice class sip-profiles 1
 response ANY sip-header Allow-Header modify "invite, options, bye, cancel, ack, prack, update, refer, subscribe, notify, info, register" "invite, bye, cancel, ack, prack, subscribe, notify, info, register"
 response 183 sip-header Allow-Header modify "ACK, BYE, CACEL, INFO, INVITE, OPTIONS, PRACK, REFER, NOTIFY" "ACK, BYE, CANCEL, INFO, INVITE, PRACK, REFER, NOTIFY"
 response ANY sip-header Allow-Header modify "INVITE, OPTIONS, BYE, CACEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER" "INVITE, BYE, CACEL, ACK, PRACK, SUBSCRIBE, NOTIFY, INFO, REGISTER"
!
voice class sip-profiles 2
 request REINVITE sdp-header Audio-Attribute modify "inactive" "sendrecv"
 request REINVITE sdp-header Audio-Attribute modify "sendonly" "sendrecv"
 request ACK sdp-header Audio-Attribute modify "sendonly" "sendrecv"
 response 200 sdp-header Audio-Attribute modify "sendonly" "sendrecv"
!
!
!
!
voice translation-rule 1
 rule 1 /71/ /1\1/
!
voice translation-rule 2
 rule 1 /^84089550468/ /14089550468/
!
voice translation-rule 7
 rule 1 /71/ /1\1/
!
!
voice translation-profile Outgoing-Fax-G711
 translate called 1
!
voice translation-profile RICUBcallingnum
 translate calling 2
 translate called 2
```

```
!
!
redundancy
 mode none
!
!
!
no ip ftp passive
ip ftp username cisco
ip ftp password cisco
!
!
interface GigabitEthernet0/0/0
 ip address 172.20.110.151 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 no ip address
 standby delay minimum 30 reload 60
 standby version 2
 standby 1 priority 50
 standby 1 track 1 decrement 10
 shutdown
 negotiation auto
 bfd interval 500 min_rx 500 multiplier 3
!
interface GigabitEthernet0/0/2
 no ip address
 shutdown
 negotiation auto
!
interface GigabitEthernet0/0/3
 no ip address
 shutdown
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
ip default-gateway 172.20.110.1
ip forward-protocol nd
!
no ip http server
no ip http secure-server
ip route 172.20.0.0 255.255.0.0 172.20.110.1
ip route 172.30.218.0 255.255.255.0 172.20.110.150
!
logging esm config
!
!
tftp-server flash:
```

```
!
control-plane
!
!
!
dial-peer voice 9000 voip[1]
 description to SP facing
 destination-pattern ..........
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class sip asserted-id pai
 dtmf-relay rtp-nte
 fax protocol pass-through g711ulaw
!
dial-peer voice 9001 voip[2]
 description Incoming to SME - SME facing
 destination-pattern 408933....
 session protocol sipv2
 session target ipv4:172.20.8.41
 session transport tcp
 dtmf-relay rtp-nte
 codec g711ulaw
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 577 voip
 description Outbound to Fax Machines
 destination-pattern 408577....
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 408577....
 codec g711ulaw
 fax protocol pass-through g711ulaw
!
dial-peer voice 3470 voip[3]
 description To Fax Machines towards SME
 shutdown
 destination-pattern 40893334..
 session protocol sipv2
 session target ipv4:172.20.8.41
 session transport udp
 incoming called-number 40893334..
 codec g711ulaw
 fax protocol pass-through g711ulaw
!
```

[1] Outbound dialpeer towards service provider
[2] Outbound dialpeer towards SME. CUBE sends all the 10 digits to SME for centralized dialplan resolution
[3] Outbound dialpeer towards fax machine gateway behind the SME (refer topology diagram for details)

```
dial-peer voice 408 voip[4]
 description To SP 7-digit dialing - SP facing
 destination-pattern .......
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class sip asserted-id pai
 dtmf-relay rtp-nte
 codec g711ulaw
 fax protocol pass-through g711ulaw
!
dial-peer voice 9003 voip[5]
 description Fax calls using G.711u - SP facing
 translation-profile outgoing Outgoing-Fax-G711
 destination-pattern 71..........
 session protocol sipv2
 session target sip-server
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 codec g711ulaw
 fax protocol pass-through g711ulaw
!
dial-peer voice 9900 voip[6]
 description to SP - SME facing
 session protocol sipv2
 session transport udp
 incoming called-number ..........
 voice-class sip asserted-id pai
 dtmf-relay rtp-nte
 codec g711ulaw
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 9901 voip[7]
 description Incoming to SME - SP facing
 session protocol sipv2
 session transport tcp
 incoming called-number 408933....
 dtmf-relay rtp-nte
 codec g711ulaw
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 4408 voip
 description To SP  7-digit dialing - SME facing
 session protocol sipv2
 session transport udp
 incoming called-number .......
```

[4] Dialpeer for 7 digit dialing towards the SP
[5] Dialpeer for Fax calls using G.711u - SP facing
[6] Incoming dialpeer SME facing
[7] Incoming dialpeer SP facing

```
 dtmf-relay rtp-nte
 fax protocol pass-through g711ulaw
!
dial-peer voice 9004 voip[8]
 description To SP - International calls - SP facing
 destination-pattern 011T
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 codec g711ulaw
 fax protocol pass-through g711ulaw
!
dial-peer voice 9904 voip[9]
 description To SP  International calls - SME facing
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 011T
 dtmf-relay rtp-nte
 codec g711ulaw
 fax protocol pass-through g711ulaw
!
dial-peer voice 1800 voip[10]
 destination-pattern 1800.......
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class sip asserted-id pai
 dtmf-relay rtp-nte
 codec g711ulaw
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 7 voip[11]
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 71………
 voice-class sip asserted-id pai
 dtmf-relay rtp-nte
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
!
sip-ua
 set pstn-cause 1 sip-status 503
```

[8] International dialing dialpeer towards the SP
[9] Incoming dialpeer for international calls from the SME
[10] Toll free dialing dialpeer
[11] Incoming fax dialpeer SME facing

```
 set pstn-cause 102 sip-status 503
 retry invite 2
 retry bye 2
 retry cancel 2
 timers trying 1000
 sip-server ipv4:172.30.218.49:5147
 g729-annexb override
!
!
line con 0
 password cisco
 login
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 password cisco
 login
!
end
```

```
MSFT-GW2#show ver
Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.1(3)T1, RELEASE SOFTWARE (fc2)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2011 by Cisco Systems, Inc.
Compiled Sun 27-Mar-11 07:36 by prod_rel_team

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

MSFT-GW2 uptime is 3 weeks, 6 days, 1 hour, 49 minutes
System returned to ROM by reload at 18:30:00 UTC Tue Aug 30 2011
System image file is "flash0:c2900-universalk9-mz.SPA.151-3.T1.bin"
Last reload type: Normal Reload
Last reload reason: Reload Command

Cisco CISCO2921/K9 (revision 1.0) with 999392K/49152K bytes of memory.
Processor board ID FTX1448AH5X
3 Gigabit Ethernet interfaces
2 Channelized E1/PRI ports
2 Channelized (E1 or T1)/PRI ports
DRAM configuration is 64 bits wide with parity enabled.
255K bytes of non-volatile configuration memory.
4099032K bytes of ATA System CompactFlash 0 (Read/Write)


License Info:

License UDI:

-------------------------------------------------
Device#   PID                   SN
-------------------------------------------------
*0        CISCO2921/K9          FTX1448AH5X



Technology Package License Information for Module:'c2900'

----------------------------------------------------------------
Technology    Technology-package          Technology-package
              Current       Type          Next reboot
-----------------------------------------------------------------
ipbase        ipbasek9      Permanent     ipbasek9
security      None          None          None
uc            uck9          Permanent     uck9
data          None          None          None

Configuration register is 0x2102

MSFT-GW2#show run
```

```
Building configuration...



Current configuration : 8733 bytes
!
! Last configuration change at 20:55:39 UTC Thu Sep 1 2011
!
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname MSFT-GW2
!
boot-start-marker
boot system flash0:c2900-universalk9-mz.SPA.151-3.T1.bin
boot-end-marker
!
!
card type e1 0 0
! card type command needed for slot/vwic-slot 1/1
logging buffered 999999
enable secret 5 $1$400b$0Kzn2HUoop0OL64b91spK.
enable password cisco
!
no aaa new-model
network-clock-participate slot 1
network-clock-participate wic 0
!
no ipv6 cef
ip source-route
ip cef
!

ip dhcp pool 3
   dns-server 172.20.2.181
   default-router 172.20.8.1
   lease 2
!
!
no ip domain lookup
multilink bundle-name authenticated
!
isdn switch-type primary-4ess
!
crypto pki token default removal timeout 0
!
crypto pki trustpoint TP-self-signed-3443052460
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-3443052460
 revocation-check none
!
```

```
!
crypto pki certificate chain TP-self-signed-3443052460
 certificate self-signed 01
  3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 33343433 30353234 3630301E 170D3131 30383330 31383332
  32345A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D33 34343330
  35323436 3030819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
  8100B84F A83B31D6 4CCB9507 199B0675 21042C87 3AC072A1 A9B87146 B33A81B9
  71449C3A 8640E523 F2E51F5C 53DA8844 25CD60F4 173A27B2 9FE5D4D0 FC6B2086
  CB52FBE7 5B75DBBA 6B0A522C 3FD88D1A 5ABE950D A0E334B8 5DC5C7E4 874CD72B
  51A60F19 F935309C DE7146C1 F5EAAFC2 A0B0DC7D 7FF195DD 6105967F 8EB1ABE8
  79510203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
  551D2304 18301680 14519620 E52CD148 41A6D69B 8EFEEA88 00ED3CA7 E7301D06
  03551D0E 04160414 519620E5 2CD14841 A6D69B8E FEEA8800 ED3CA7E7 300D0609
  2A864886 F70D0101 04050003 81810076 D12A4112 963CACAA 59657B51 EA051F2F
  4555E416 2FDF32DF 98D031A0 17C88DBB 8ADBD8D9 0770062B 2295D315 7E38C18F
  B4C0CCF6 2555CA47 20D09D2B CED701F9 445AD9B3 322783C8 736F1CEE F2AC2F59
  823B4C75 5985F137 2A1B77AB 8B6007CE 4FC2C29C 228A3BB2 F0B841B8 5CFA5201
  EAEDB235 E35340FD B79542E7 8D754F
        Quit
!
voice-card 0
 dspfarm
 dsp services dspfarm[12]
!
!
voice service pots
 map q850-cause 47 release-source all tone 1
 supplementary-service qsig call-forward
 supplementary-service qsig mwi release-call
 fax rate disable
!
voice service voip
 ip address trusted list
  ipv4 172.20.8.38
  ipv4 172.20.8.30
  ipv4 172.20.8.59
  ipv4 172.20.8.85
  ipv4 172.20.8.98
 ip address trusted call-block cause dest-unroutable
 sip
!
voice class h323 2
  h225 timeout connect 60
  h225 timeout call-proceeding 5
  h225 timeout ntf 5000
 h225 display-ie ccm-compatible
  call start fast
  call preserve limit-media-detection
  h245 caps suppress nte
```

[12] Enabling DSP sharing and DSp farm services

```
  telephony-service ccm-compatible
icense udi pid CISCO2921/K9 sn FTX1448AH5X
hw-module pvdm 0/0
!
!
redundancy
!
!
controller E1 0/0/0
 shutdown
!
controller E1 0/0/1
!
interface GigabitEthernet0/0
 description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
 bandwidth 1000000
 ip address 172.20.8.58 255.255.255.0
 duplex auto
 speed auto
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface GigabitEthernet0/2
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip default-gateway 172.20.8.1
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 255.255.255.0 172.20.8.1
ip route 0.0.0.0 255.255.255.255 GigabitEthernet0/0
ip route 172.20.8.1 255.255.255.255 GigabitEthernet0/0
!
access-list 23 permit 10.10.10.0 0.0.0.7
!
!
!
control-plane
!
!
ccm-manager fallback-mgcp
ccm-manager mgcp
```

```
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
!
mgcp
mgcp call-agent 172.20.8.38 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
mgcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
!
mgcp profile default
!
sccp local GigabitEthernet0/0
sccp ccm 172.20.8.42 identifier 11 version 7.0 [13]
sccp ccm 172.20.8.41 identifier 1 priority 1 version 7.0 [14]
sccp
!
sccp ccm group 11
 associate ccm 11 priority 1
 associate profile 11 register CONF-VGW3
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register XCODE-SJC
 associate profile 2 register IOS-MTP-SJC
 associate profile 3 register IOS-MTP-SJ-G711
 associate profile 7 register G729-Soft
!
dspfarm profile 1 transcode [15]
 codec g729r8
 codec g711ulaw
 codec g711alaw
 maximum sessions 6
 associate application SCCP
!
dspfarm profile 4 transcode
 description CME XCODER
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 maximum sessions 2
 associate application SCCP
 shutdown
```

[13] For resources registering to the CUCM
[14] For resources registering to the SME
[15] Transcoder resources

```
!
dspfarm profile 11 conference [16]
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 codec g722-64
 maximum conference-participants 16
 maximum sessions 1
 associate application SCCP
!
dspfarm profile 2 mtp [17]
 codec g711ulaw
 maximum sessions software 200
 associate application SCCP
!
dspfarm profile 3 mtp
 description SJC G711 MTP
 codec g711ulaw
 maximum sessions software 200
 shutdown
!
dspfarm profile 7 mtp
 codec g711ulaw
 maximum sessions software 10
 associate application SCCP
!
dspfarm profile 8 mtp
 codec g711ulaw
 shutdown
!
dial-peer voice 778 voip
 mailbox-selection orig-called-num
 session protocol sipv2
!
!
!
!
gatekeeper
 shutdown
!
!
telephony-service
 srst mode auto-provision none
 srst ephone description SRST..Welcome.. : May 19 2011 19:02:06
 srst dn line-mode octo
 max-ephones 20
 max-dn 20 no-reg both
 ip source-address 172.20.8.58 port 2000
```

[16] Conference bridge
[17] Software MTP

```
 max-conferences 8 gain -6
 transfer-system full-consult
!
!
ephone-dn-template 1
 call-waiting ring
 caller-id block
 huntstop channel 2
!
!
ephone-template 1
 max-calls-per-button 2
 speed-dial 1 2003 label "SIP-CME-2003"
 speed-dial 2 2004 label "SIP-CME-2004"
!
!
line con 0
 password cisco
 login
line aux 0
```

Fax Gateway Configuration

```
show ver
Cisco IOS Software, 2800 Software (C2800NM-IPVOICEK9-M), Version 15.0(1)M4, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2010 by Cisco Systems, Inc.
Compiled Thu 28-Oct-10 17:09 by prod_rel_team

ROM: System Bootstrap, Version 12.4(1r) [hqluong 1r], RELEASE SOFTWARE (fc1)

Cisco 2811 (revision 53.50) with 249856K/12288K bytes of memory.
Processor board ID FTX1040A1LY
2 FastEthernet interfaces
2 Voice FXS interfaces
DRAM configuration is 64 bits wide with parity enabled.
239K bytes of non-volatile configuration memory.
125440K bytes of ATA CompactFlash (Read/Write)


License Info:

License UDI:

-------------------------------------------------
Device#   PID                   SN
-------------------------------------------------
*0        CISCO2811             FTX1040A1LY



Configuration register is 0x2102

IOSGW#show run
Building configuration...


Current configuration : 5053 bytes
!
! Last configuration change at 00:50:39 UTC Thu Sep 8 2011
!
version 15.0
service timestamps debug uptime
service timestamps log datetime msec
no service password-encryption
service linenumber
!
hostname IOSGW
!
boot-start-marker
boot-end-marker
```

```
!
logging buffered 900000
enable password cisco
!
no aaa new-model
!
dot11 syslog
ip source-route
!
!
ip cef
!
!
no ip domain lookup
no ipv6 cef
multilink bundle-name authenticated
!
!
voice service voip
 h323
 sip
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
!
!
voice translation-rule 1
 rule 1 /^9/ /8/
!
voice translation-rule 2
 rule 2 /^.*\(....\)/ /732216\1/
!
!
voice translation-profile NPA
 translate calling 2
!
!
voice-card 0
 dspfarm
 dsp services dspfarm
!
license udi pid CISCO2811 sn FTX1040A1LY
archive
 log config
  hidekeys
username cisco privilege 15 secret 5 $1$1IWY$fwla1ooHsE/ORF20GsQFz.
!
interface FastEthernet0/0
 ip address 172.20.109.201 255.255.255.0
 duplex auto
 speed auto
!
```

```
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 172.16.100.1
ip route 0.0.0.0 0.0.0.0 172.20.109.1
!
!
!
!
control-plane
!
!
voice-port 0/0/0
 timeouts ringing infinity
 station-id name FAX-GW1
 station-id number 4089333473
!
voice-port 0/0/1
 timeouts ringing infinity
 station-id name 4089336163
 station-id number 4089333474
!
!
no mgcp package-capability res-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
!
dial-peer voice 2715 pots
 destination-pattern 4089336164
 port 0/0/0
 forward-digits 0
!
dial-peer voice 2717 pots
 destination-pattern 4089336163
 port 0/0/1
 forward-digits 0
!
dial-peer voice 711 voip[18]
 description outgoing call to International
 destination-pattern 7011T
 translate-outgoing calling 2
 session protocol sipv2
 session target ipv4:172.20.8.42
 incoming called-number 27..
```

[18] International fax calling

```
 dtmf-relay rtp-nte
 playout-delay nominal 80
 playout-delay mode fixed
 codec g711ulaw
 no vad
!
dial-peer voice 7 voip[19]
 destination-pattern 71……...
 session protocol sipv2
 session target ipv4:172.20.8.42
 incoming called-number 347.
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 3473 pots
 destination-pattern 3473
 port 0/0/0
!
dial-peer voice 3474 pots
 destination-pattern 3474
 port 0/0/1
!
!
sip-ua
 protocol mode ipv4
!
!
line con 0
 exec-timeout 610 0
 password cisco
 login
line aux 0
line vty 0 4
 exec-timeout 0 0
 password cisco
 login
!
scheduler allocate 20000 1000
end

IOSGW#
```

[19] Longdistance and local fax calling

Acronyms

Acronym	Definition
Lync	Lync 2010 Server
DTMF	Dual Tone Multi Frequency
SIP	Session Initiation Protocol
SDP	Session Description Protocol
B2BUA	Business to business User Agent
OITT	Open Interoperability Test Tool
UC-OIP	Microsoft Unified Communications Open Interoperability Program
UC	Unified Communications; also referenced in this document as to a user who is enabled for voice on Lync Server 2010 (i.e. "UC enabled")
LyncIT	Lync server 2010 Interoperability Test Tool
TCP	Transmission Control Protocol
TLS	Transport Layer Security
GW	Gateway
S/W	Software
DB	Database

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