Application Note

Microsoft Lync 2010 using SIP via Cisco Unified Communications Manager – Session Manager Edition 9.0 to Cisco Unified Communications Manager 9.0 and Cisco Unified Border Element (Enterprise Edition) 1.4 on ASR to Service Provider

October 19, 2012 – 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) 9.0 via Direct SIP to Microsoft Lync 2010 server using Microsoft Mediation Server, and via SIP connection to a Cisco Unified Communications Manager (Cisco UCM) 9.0, 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 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 1001 Aggregated Services Router
  Cisco ASR 1004 Aggregated Services Router with Route Processor-2
  Cisco ASR 1006 Aggregated Services Router with Route Processor-2
  Cisco 3900 Series Integrated Services Routers
  Cisco 2900 Series Integrated Services Routers
  Cisco AS5350XM Universal Gateway
  Cisco AS5400XM Universal Gateway
  Cisco 1861 Integrated Services Router
  Cisco 881 Integrated Services Router
  Cisco 888 Integrated Services Router
  Cisco IAD880 Series Integrated Access Devices
  Cisco IAD2430 Integrated Access Device

- 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

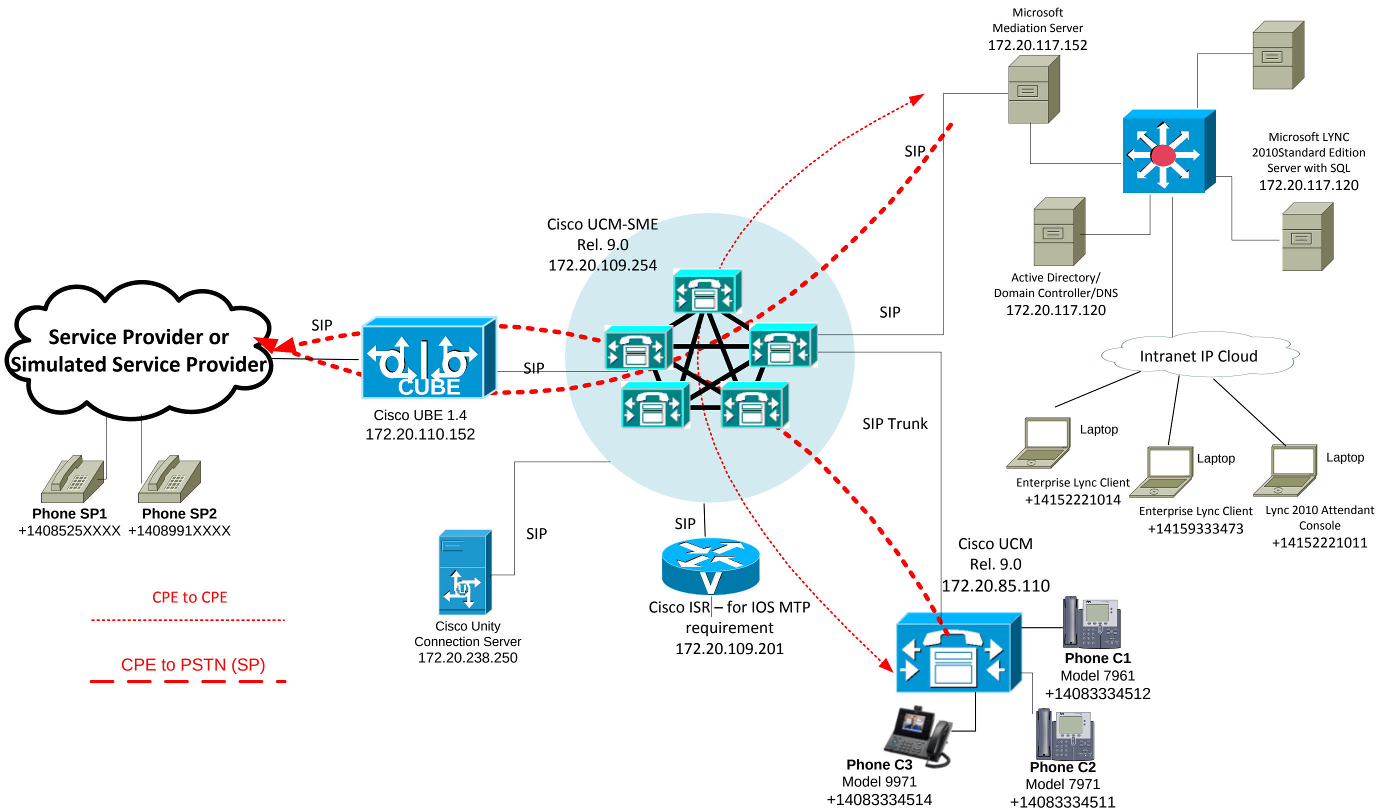


Figure 1. Basic Call Setup

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 and later, the Media Termination Point (MTP) checkbox under the SIP Trunk configuration page no longer needs to be checked to implement early offer function.

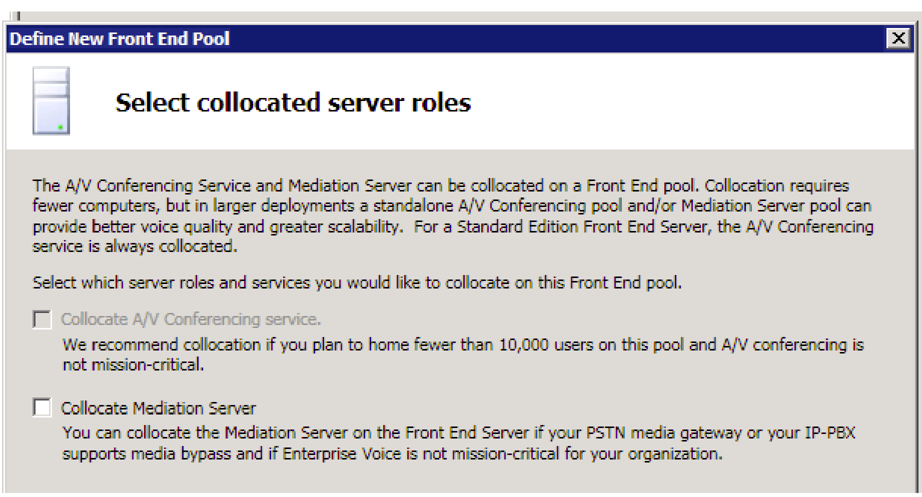
Capabilities

- 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.
- Cisco UCM now supports the use of the Audio Codec Preference and it uses them to select codec or prioritize codec's in the list. System now allows configuring customized Codec Preference List changing the order of codec's of any existing lists (Lossy and Low-Loss) as needed. System also allows the Admin to control if the received preference order of codec for SIP device can be honored by configuring "Accept Audio Codec Preferences in Received Offer" in SIP Profile or Service Parameter.
- Cisco UCM supports URI dialing but with certain new model of IP Phones and other endpoints only.

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, so the Fax is not tested on Lync.
- 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
- URI Dialing

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.
- G.711 fax pass-through was not tested as Microsoft Lync does not support it.
- 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

The following hardware is required:

- Cisco MCS 7835 Unified Communications Manager
- Cisco MCS 7835 Session Manager Edition
- Cisco 3825 voice gateway
- Cisco ASR1K router
- One Cisco Unified IP phone 7961 configured as SCCP phone
- One Cisco Unified IP phone 7971 configured as SCCP phone
- One Cisco Unified IP phone 9971 configured as SIP phone (Supports URI Dialing)
- Two DELL notebooks running LYNC clients
- One DELL notebook running LYNC 2010 Attendant Console

Software Requirements

The following software is required:

- Cisco Unified Communications Manager Release 9.0
- Cisco UCM-Session Manager Edition Release 9.0
- Microsoft Lync Server 2010 Standard Edition, Windows Server 2008 R2 SP1 x64 Enterprise Edition OS
- Windows Active Directory/DNS/Cert Server, Windows Server 2008 R2 SP1 x64 Enterprise Edition OS
- Backend Database: Windows SQL Server 2008 Enterprise Edition, Windows Server 2008 R2 SP1 x64 Enterprise Edition OS
- Microsoft Lync 2010 (build 4.0.7577.4)
- Cisco ASR1K router Release 3.4.2.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, data sheet, compatibility matrix, deployment, installation guides etc, go to:

http://www.cisco.com/en/US/prod/collateral/voicesw/ps6788/vcallcon/ps556/ps12515/data_sheet_c78-710875.html

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/docguide/9_0_1/CUCM_BK_CEDBAA6F_00_cucm-documentation-guide-90.html

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/compat/ccmcompmatr1.pdf

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/9_0_1/CUCM_BK_RF912712_00_cucm-release-notes-90.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:

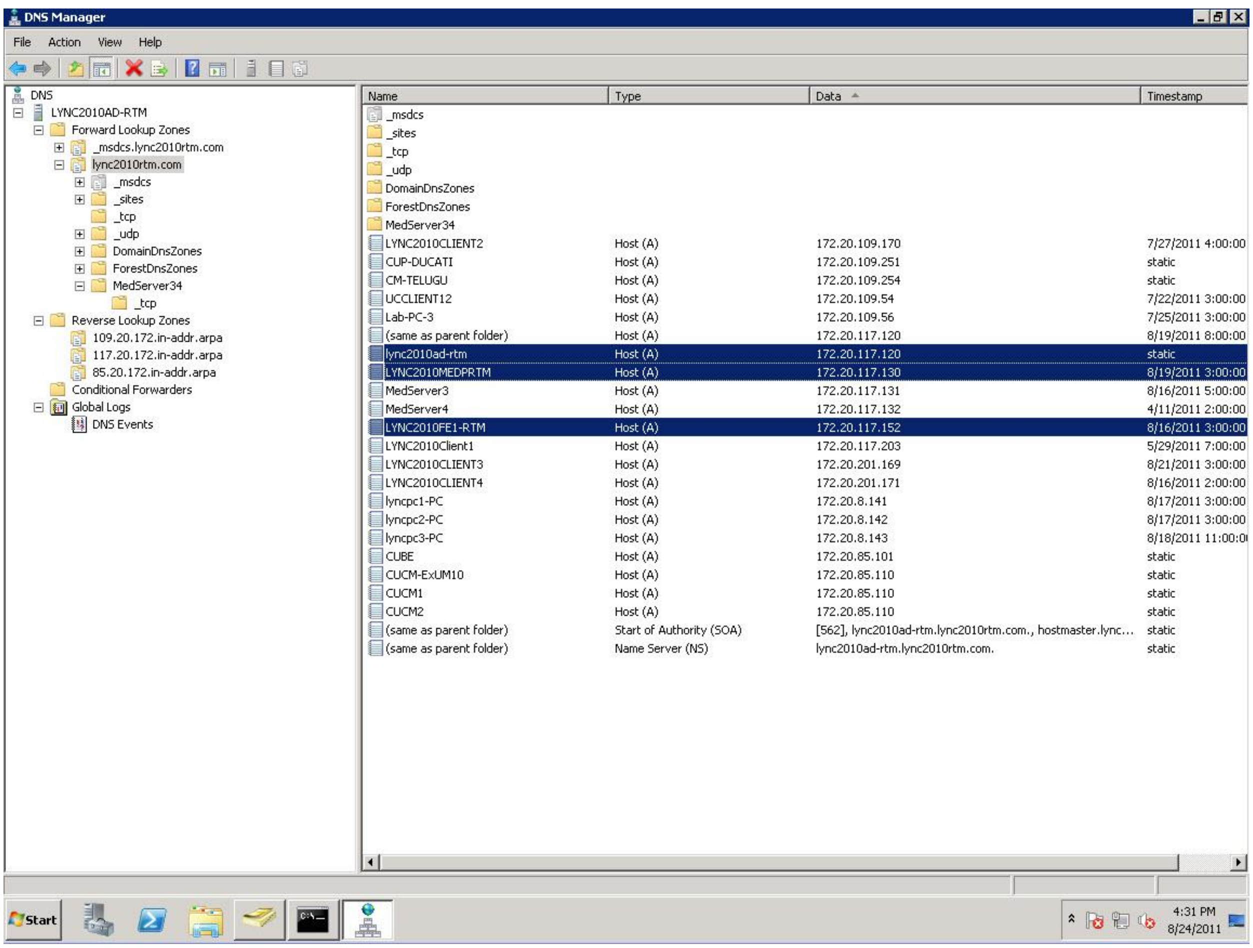
1. Domain Name System Configuration
2. Front End Server/Pool Configuration
3. Mediation Server Configuration
4. User Configuration
5. Microsoft Lync 2010 Configuration

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

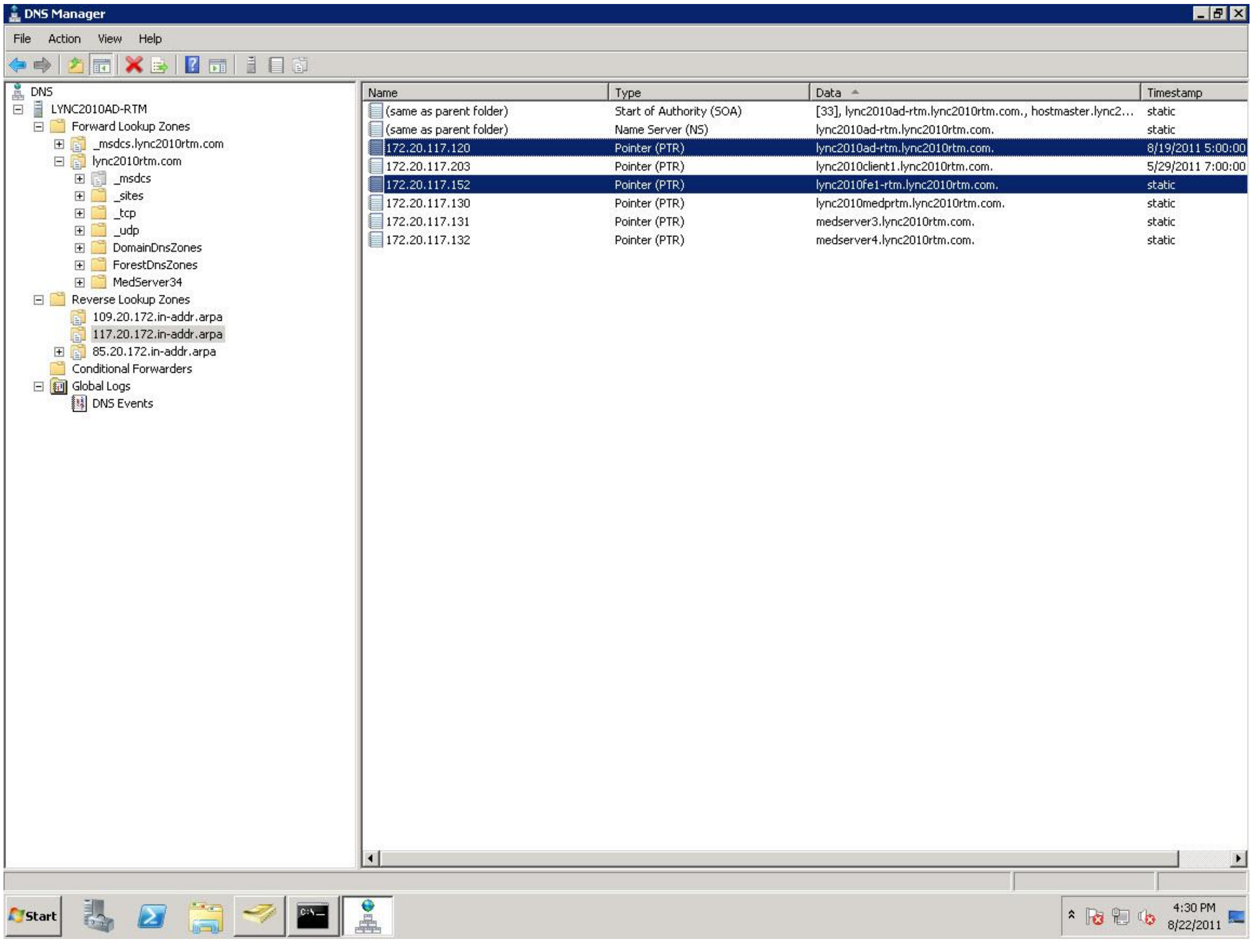
Start > Administrative Tools > DNS

Forward Lookup Zone



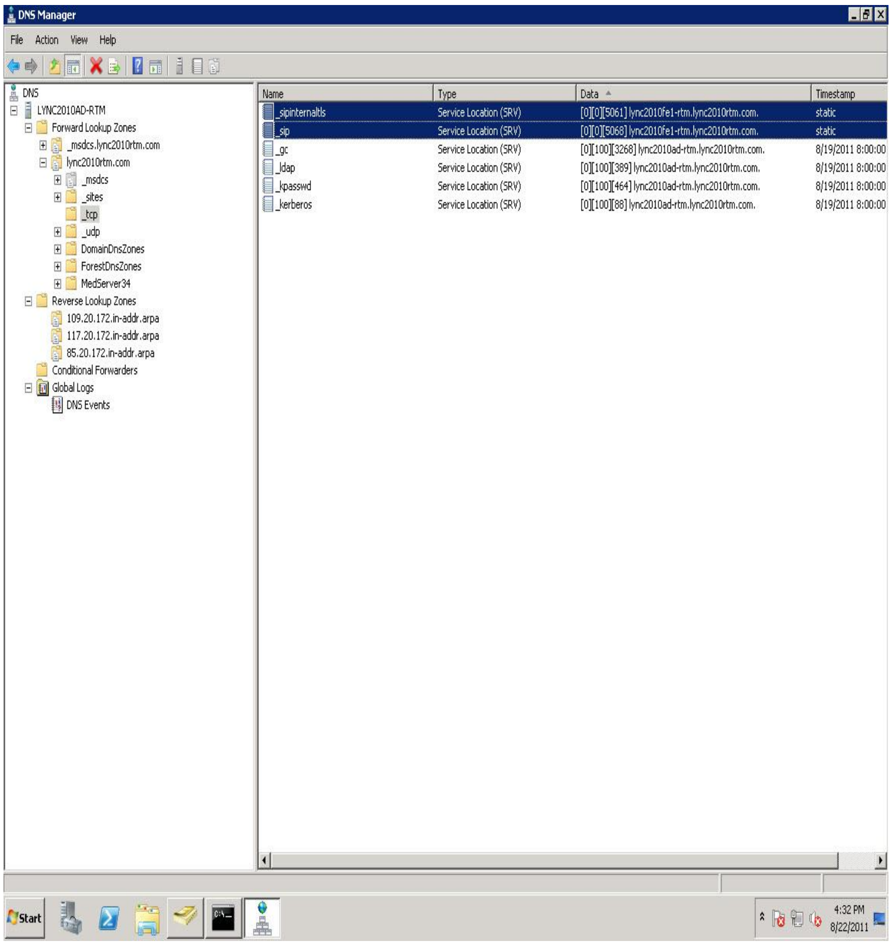
Host A records added for Lync Front End server, Active Directory/DNS and Mediation Server.

Reverse Lookup Zones Configuration

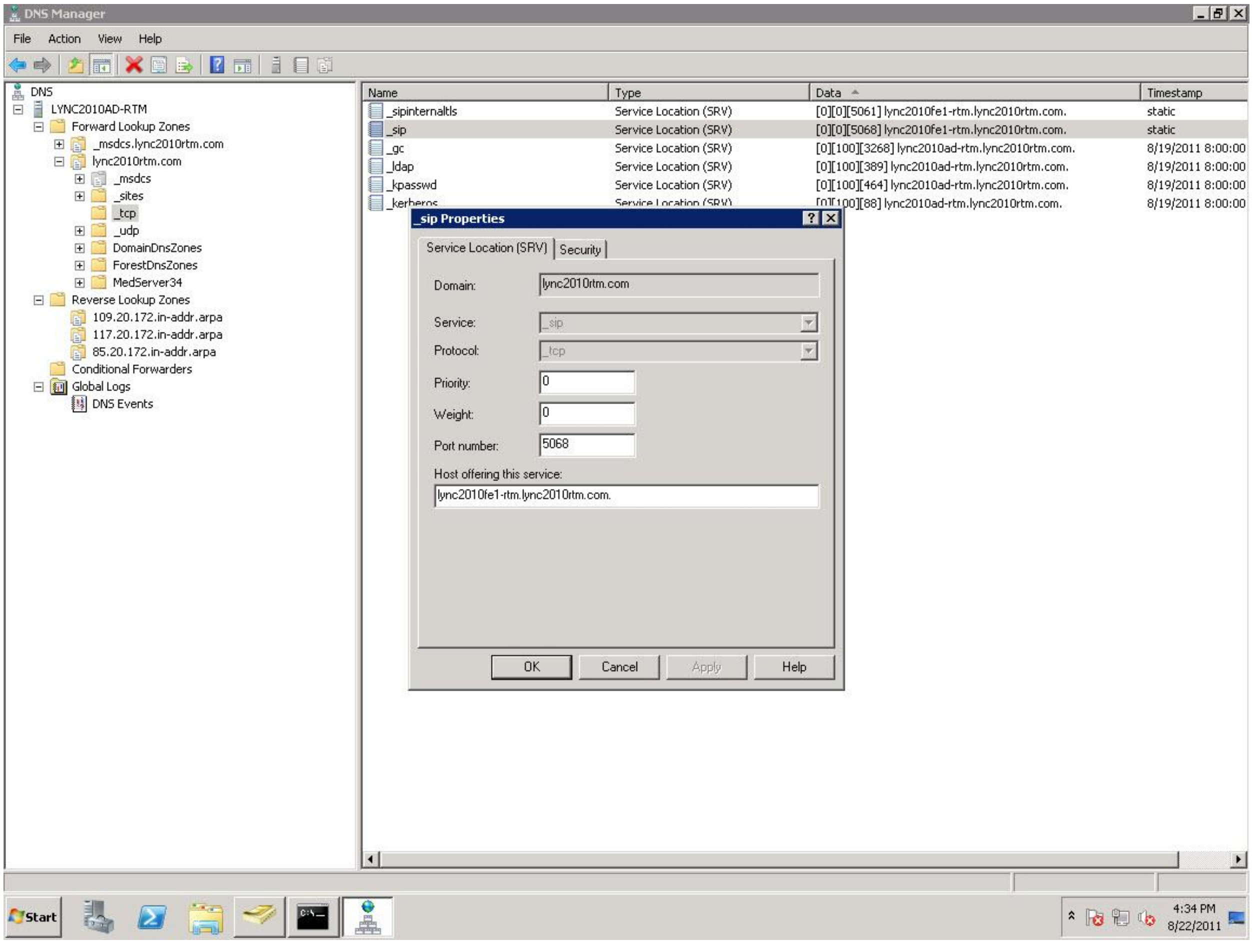


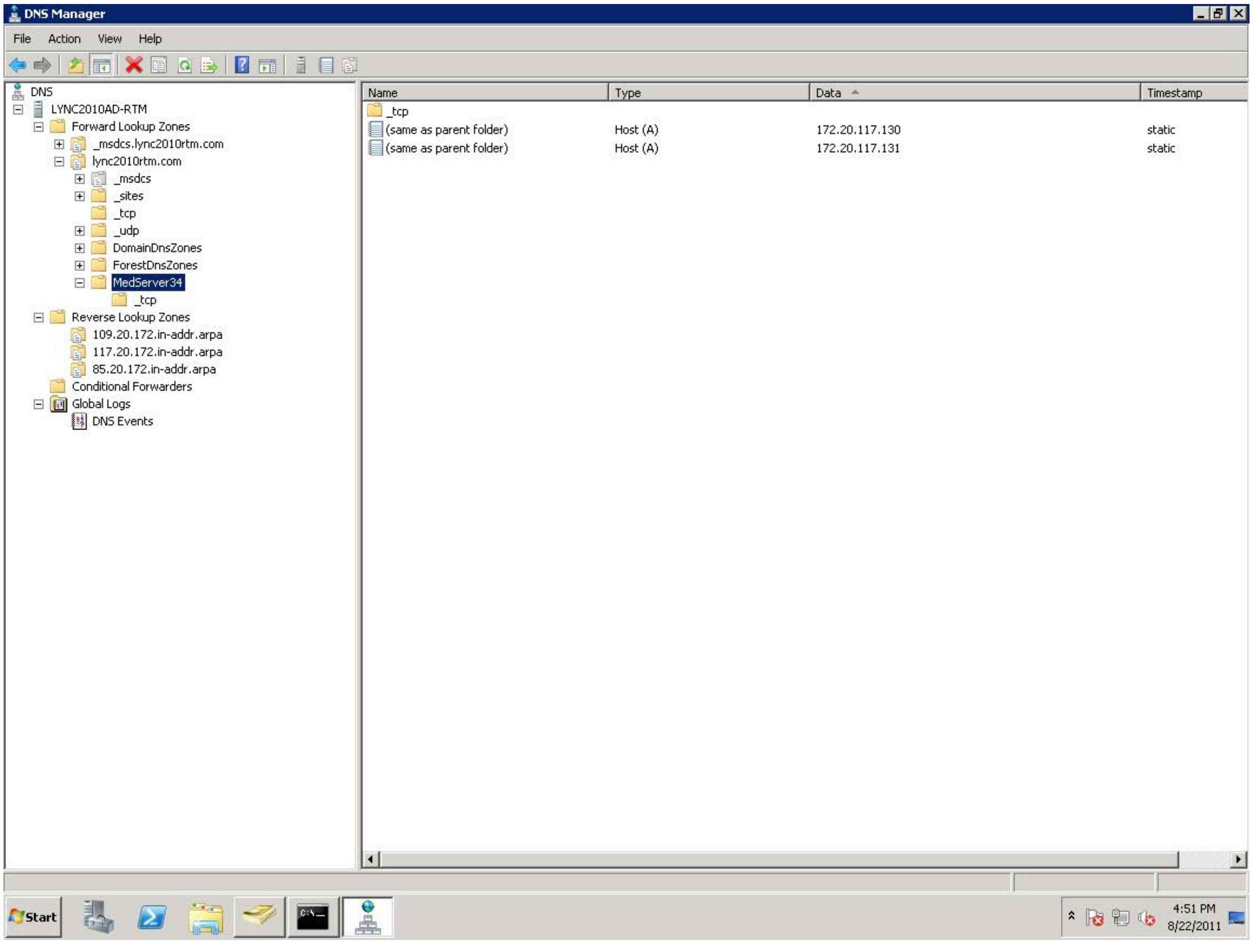
PTR Records added for the Lync server and Active Directory.

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
_kerberos Service Location (SRV) [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
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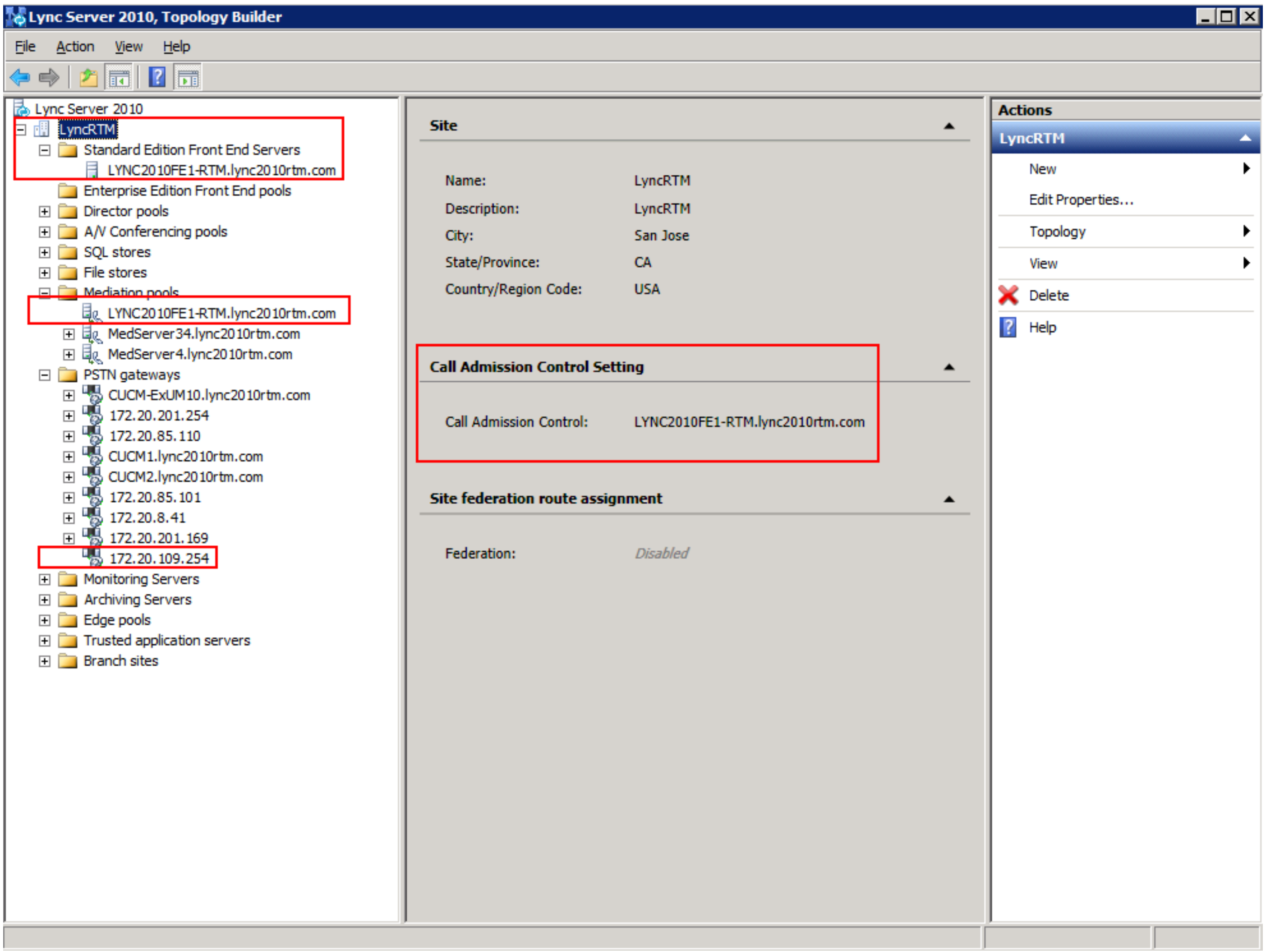
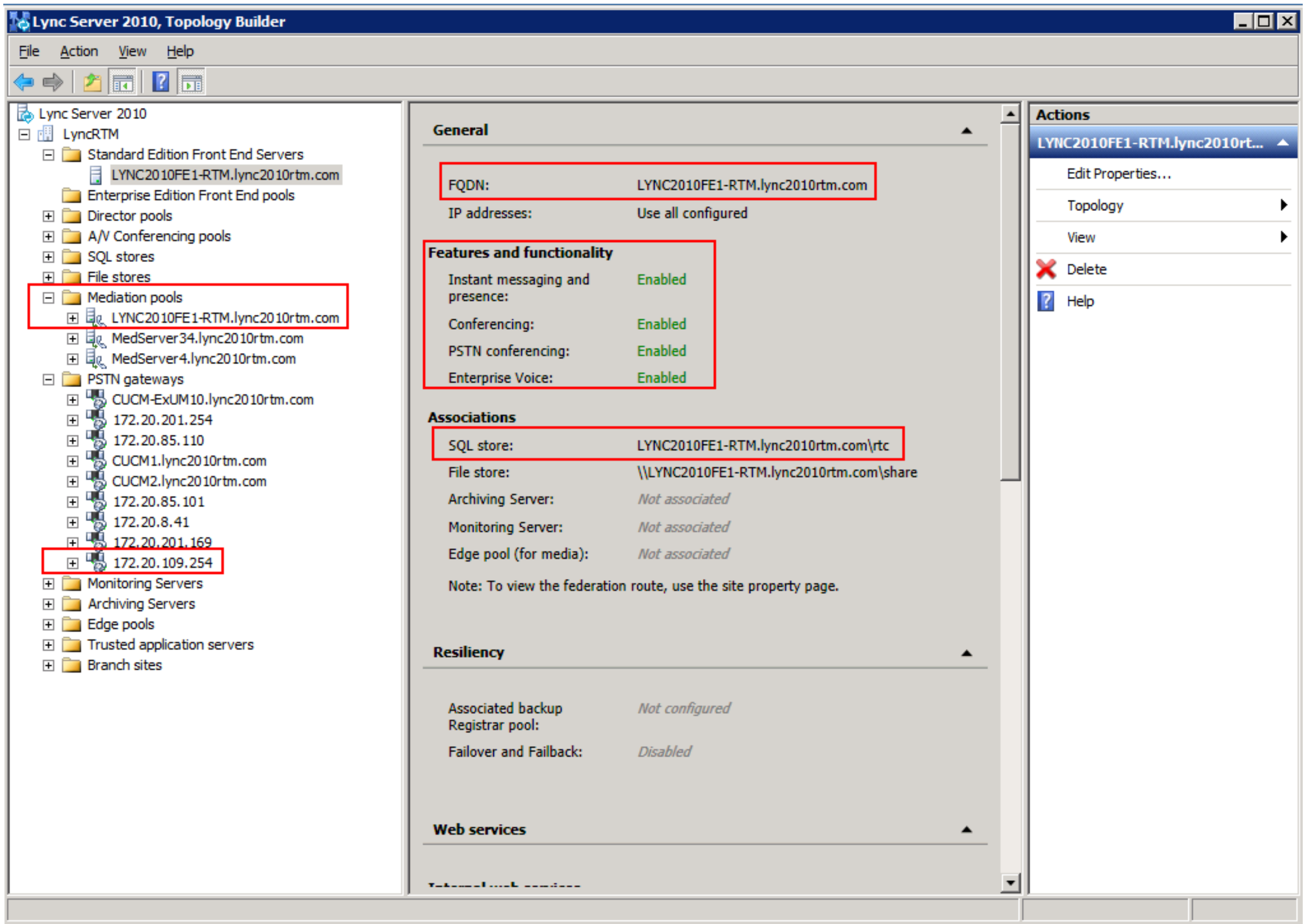
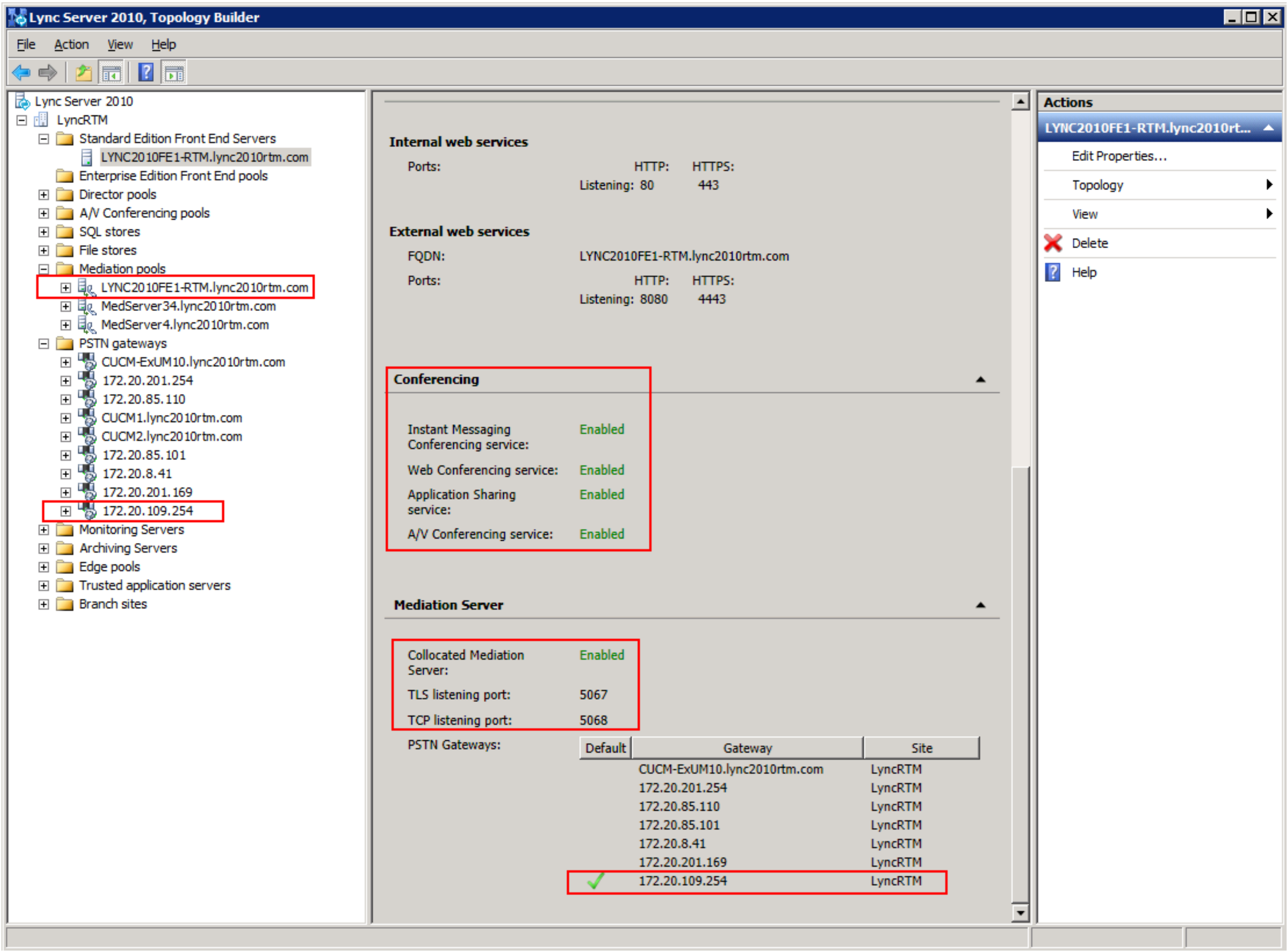
DNS
LYNC2010AD-RTM
Forward Lookup Zones
_msdcs.lync2010rtm.com
lync2010rtm.com
_msdcs
_sites
_tcp
_udp
DomainDnsZones
ForestDnsZones
MedServer34
_tcp
Reverse Lookup Zones
109.20.172.in-addr.arpa
117.20.172.in-addr.arpa
85.20.172.in-addr.arpa
Conditional Forwarders
Global Logs
DNS Events
Name
Type
Data
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

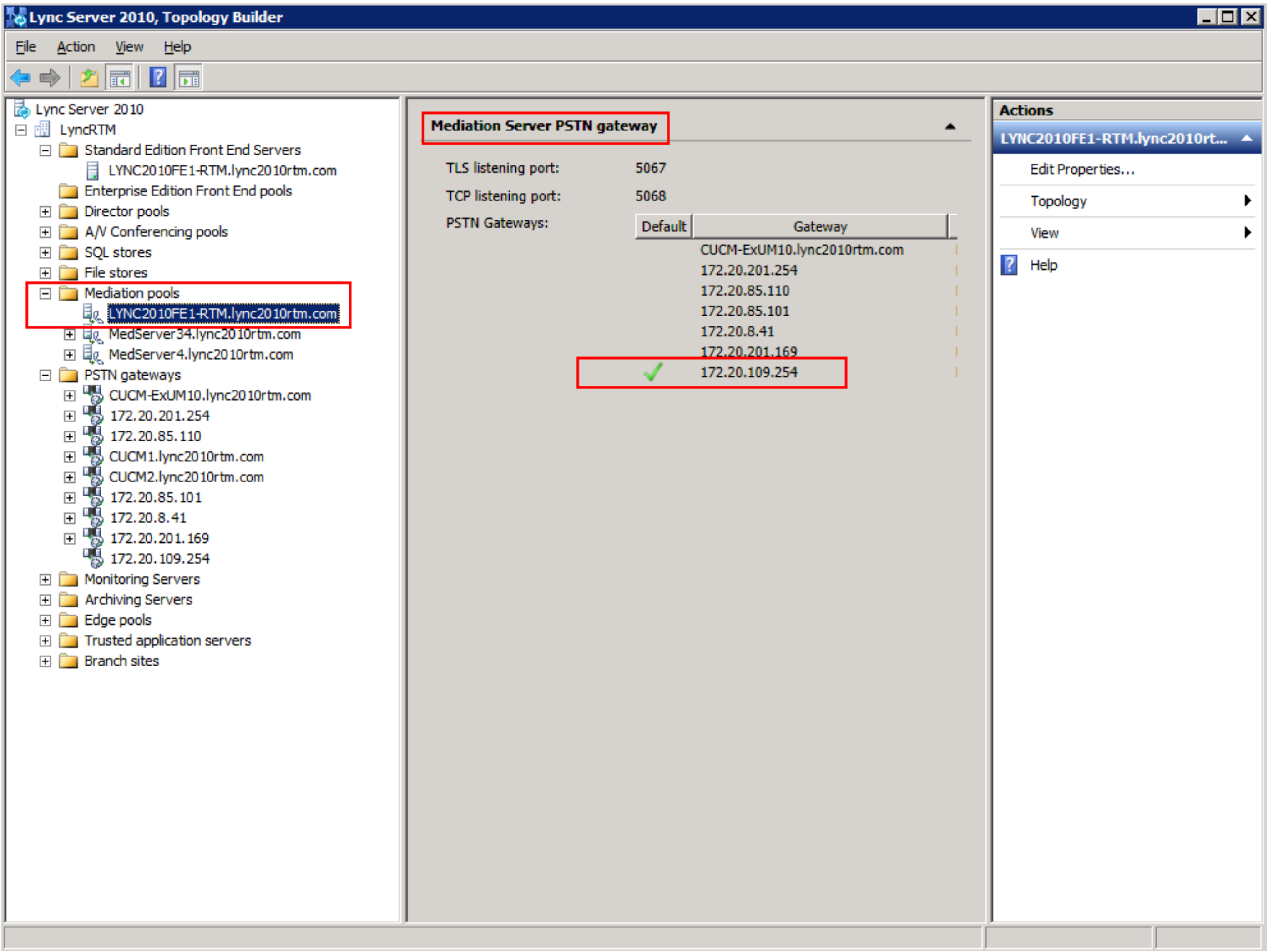


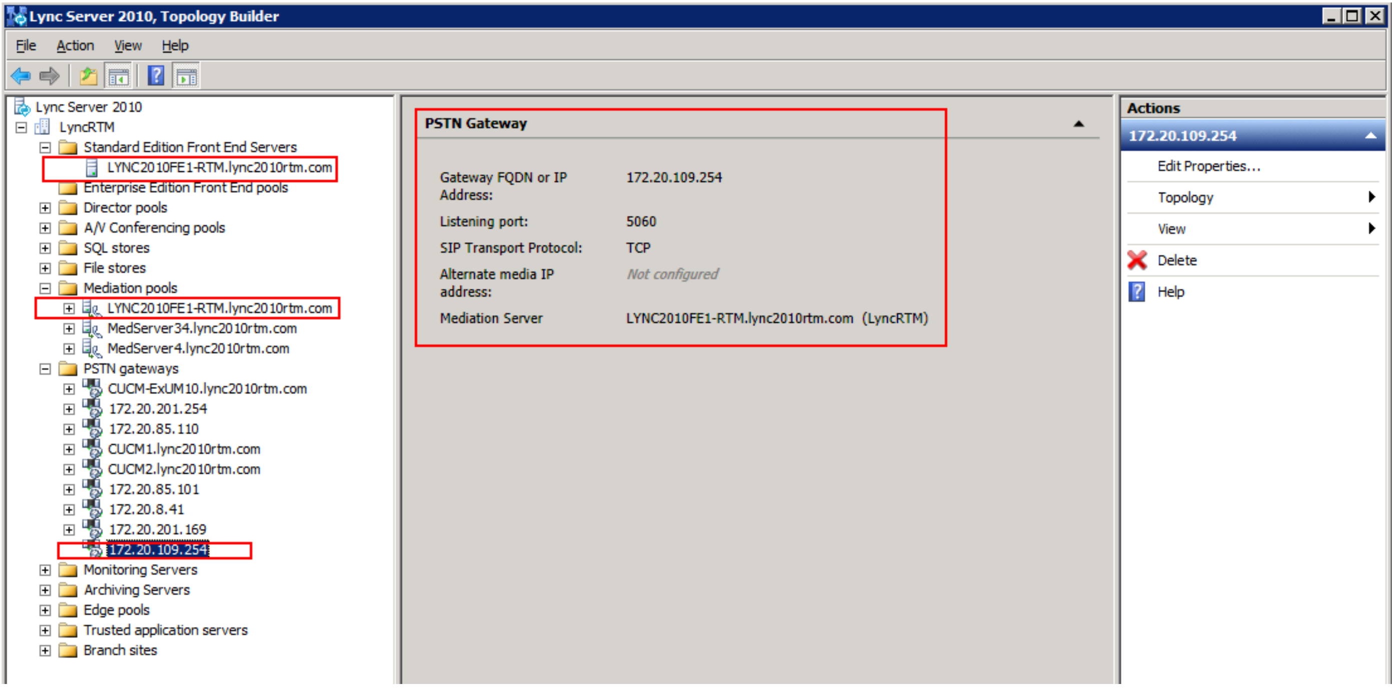

Lync Server 2010, Topology Builder
File Action View Help
Lync Server 2010
LyncRTM
Standard Edition Front End Servers
LYNC2010FE1-RTM.lync2010rtm.com
Enterprise Edition Front End pools
Director pools
A/V Conferencing pools
SQL stores
File stores
Mediation pools
LYNC2010FE1-RTM.lync2010rtm.com
MedServer34.lync2010rtm.com
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
172.20.109.254
Monitoring Servers
Archiving Servers
Edge pools
Trusted application servers
Branch sites
General
FQDN: LYNC2010FE1-RTM.lync2010rtm.com
IP addresses: Use all configured
Features and functionality
Instant messaging and presence: Enabled
Conferencing: Enabled
PSTN conferencing: Enabled
Enterprise Voice: Enabled
Associations
SQL store: LYNC2010FE1-RTM.lync2010rtm.com\rtc
File store: \\LYNC2010FE1-RTM.lync2010rtm.com\share
Archiving Server: Not associated
Monitoring Server: Not associated
Edge pool (for media): Not associated
Note: To view the federation route, use the site property page.
Resiliency
Associated backup Registrar pool: Not configured
Failover and Failback: Disabled
Web services
Actions
LYNC2010FE1-RTM.lync2010rt...
Edit Properties...
Topology
View
Delete
Help



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)

Lync Front End server configuration with FQDN and PSTN Gateway associated with the Mediation Pool

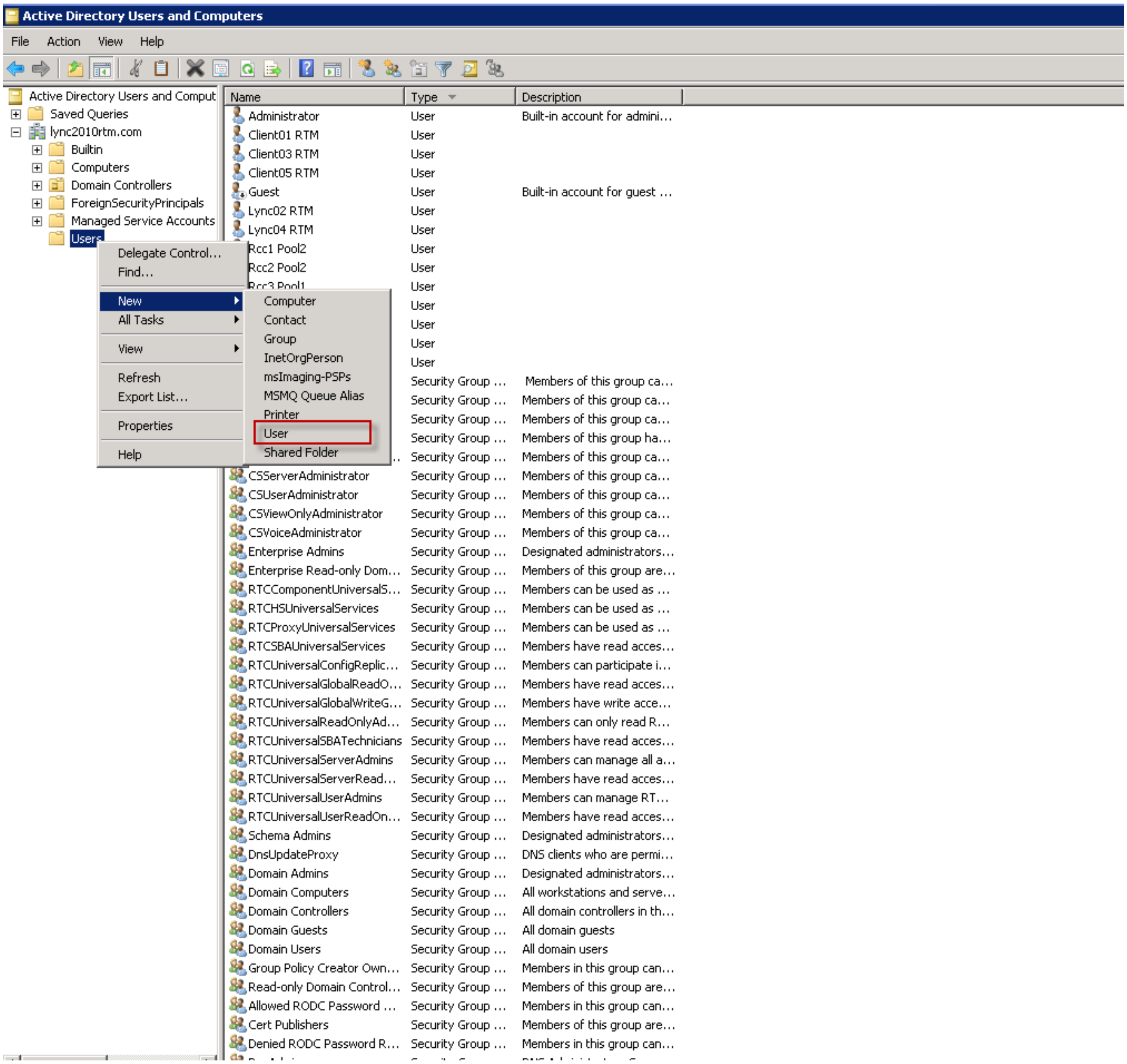


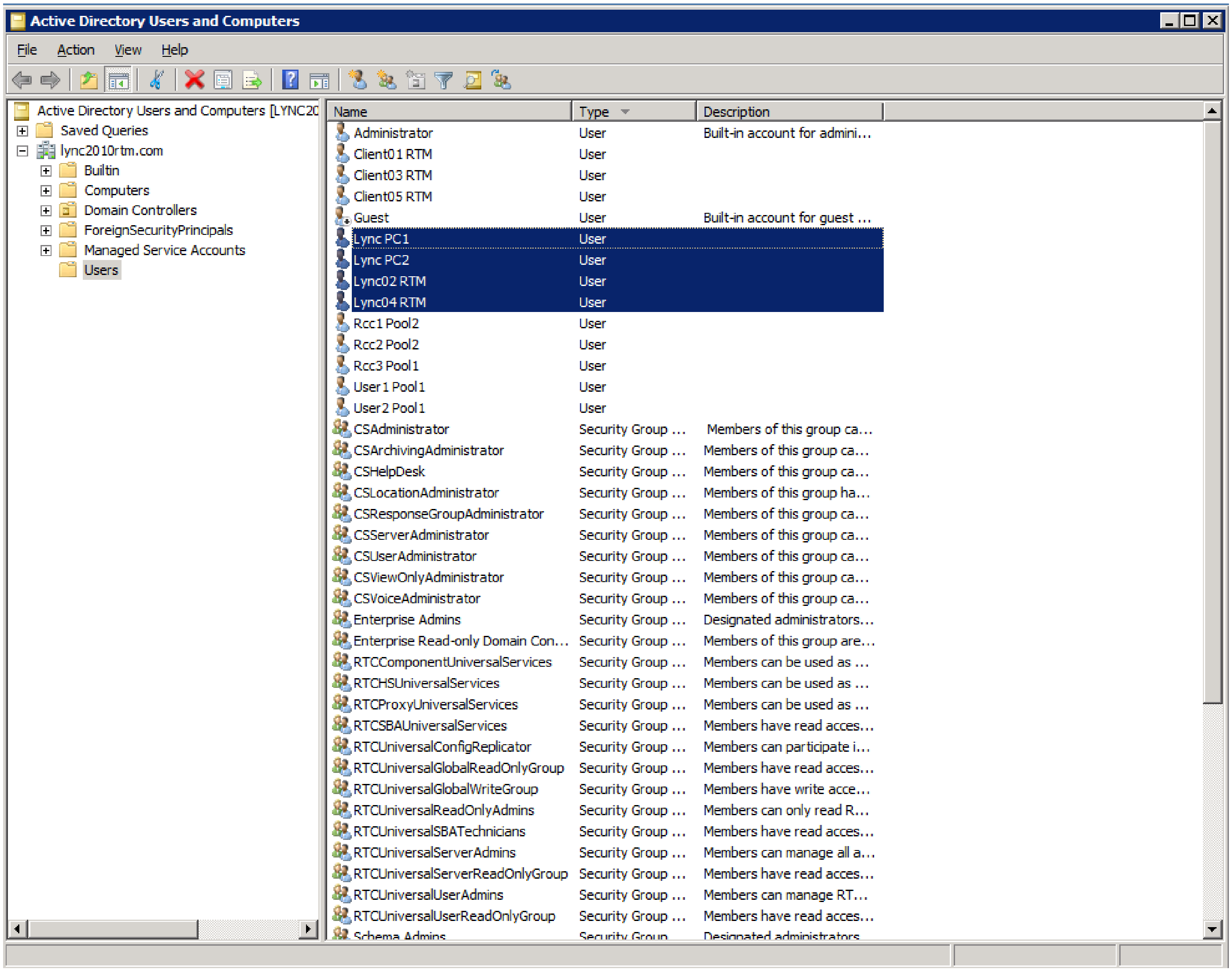


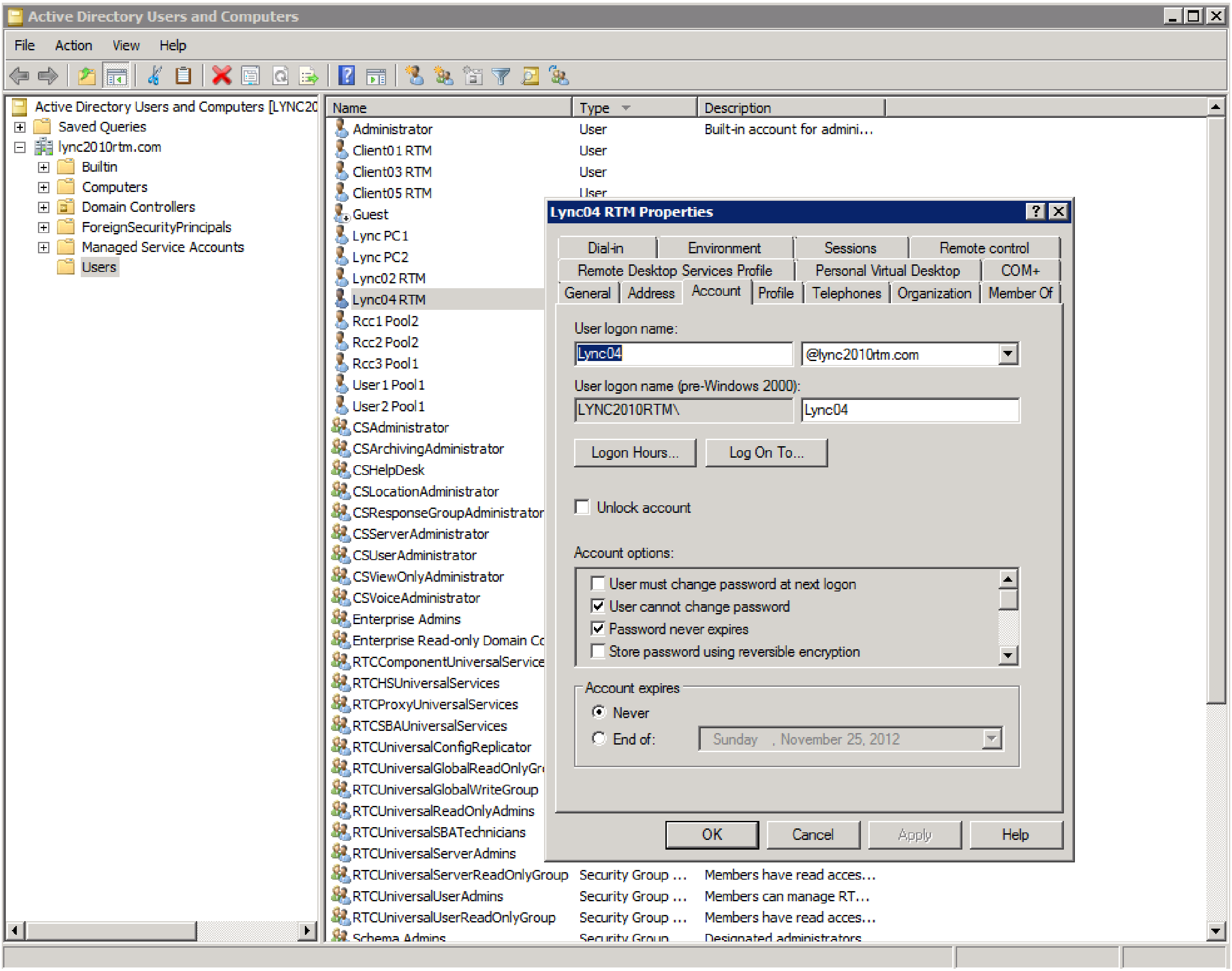
Listening port of the SME is left to default 5060.

Active Directory User Configuration

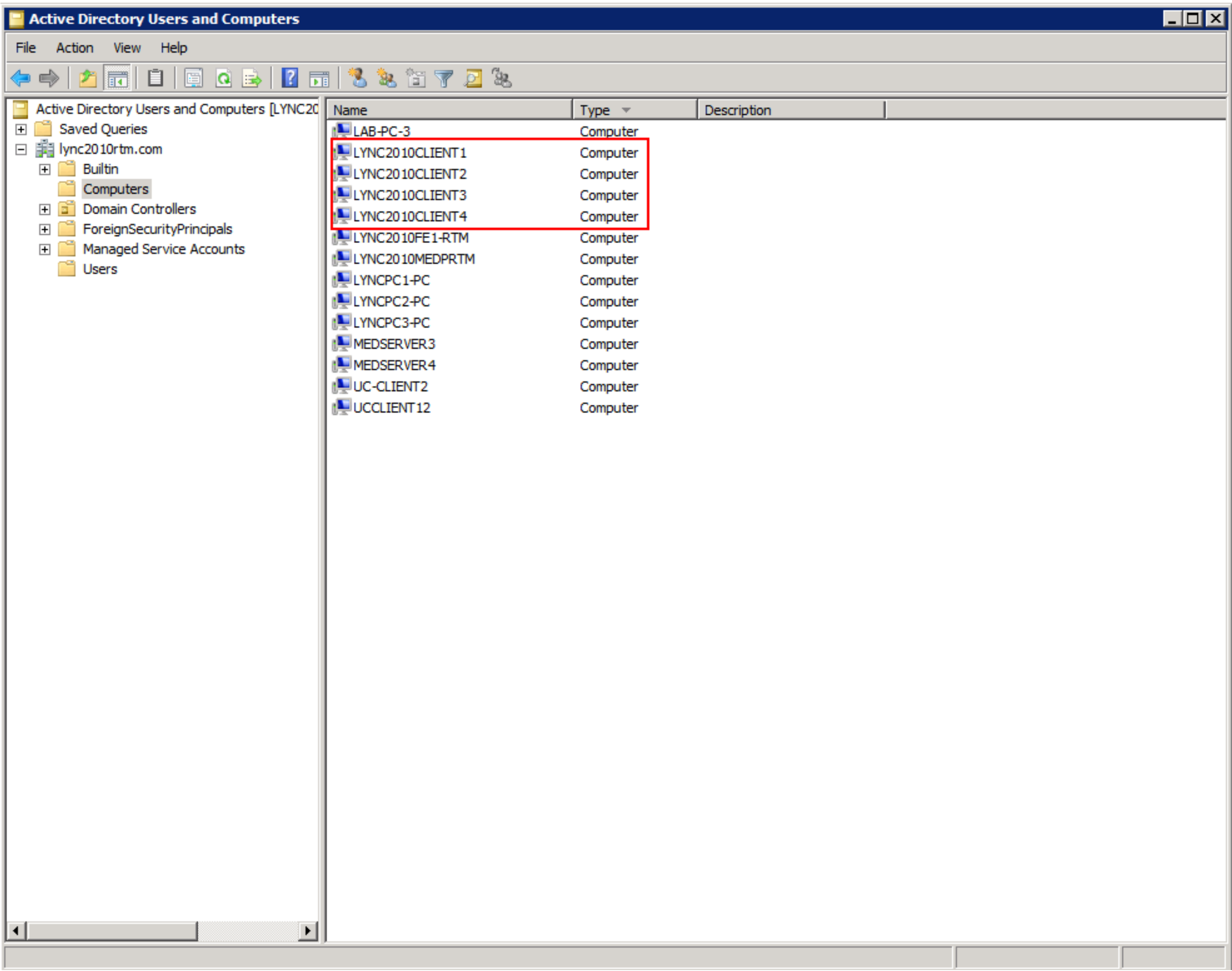
Create users from Front End Server by accessing the Active Directory Users and Computers window




Active Directory Users and Computers
File
Action
View
Help
Active Directory Users and Computers [LYNC20
Saved Queries
lync2010rtm.com
Builtin
Computers
Domain Controllers
ForeignSecurityPrincipals
Managed Service Accounts
Users
Name
Type
Description
Administrator User Built-in account for admini...
Client01 RTM User
Client03 RTM User
Client05 RTM User
Guest User Built-in account for guest ...
Lync PC1 User
Lync PC2 User
Lync02 RTM User
Lync04 RTM User
Rcc1 Pool2 User
Rcc2 Pool2 User
Rcc3 Pool1 User
User1 Pool1 User
User2 Pool1 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...
CSResponseGroupAdministrator 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 Domain Con... Security Group ... Members of this group are...
RTCComponentUniversalServices Security Group ... Members can be used as ...
RTCHSUniversalServices Security Group ... Members can be used as ...
RTCProxyUniversalServices Security Group ... Members can be used as ...
RTCSBAUniversalServices Security Group ... Members have read acces...
RTCUniversalConfigReplicator Security Group ... Members can participate i...
RTCUniversalGlobalReadOnlyGroup Security Group ... Members have read acces...
RTCUniversalGlobalWriteGroup Security Group ... Members have write acce...
RTCUniversalReadOnlyAdmins Security Group ... Members can only read R...
RTCUniversalSBATechnicians Security Group ... Members have read acces...
RTCUniversalServerAdmins Security Group ... Members can manage all a...
RTCUniversalServerReadOnlyGroup Security Group ... Members have read acces...
RTCUniversalUserAdmins Security Group ... Members can manage RT...
RTCUniversalUserReadOnlyGroup Security Group ... Members have read acces...


Active Directory Users and Computers
File
Action
View
Help
Active Directory Users and Computers [LYNC20
Saved Queries
lync2010rtm.com
Builtin
Computers
Domain Controllers
ForeignSecurityPrincipals
Managed Service Accounts
Users
Name
Type
Description
Administrator
User
Built-in account for admini...
Client01 RTM
Client03 RTM
Client05 RTM
Guest
Lync PC1
Lync PC2
Lync02 RTM
Lync04 RTM
Rcc1 Pool2
Rcc2 Pool2
Rcc3 Pool1
User1 Pool1
User2 Pool1
CSAdministrator
CSArchivingAdministrator
CSHelpDesk
CSLocationAdministrator
CSResponseGroupAdministrator
CSServerAdministrator
CSUserAdministrator
CSViewOnlyAdministrator
CSVoiceAdministrator
Enterprise Admins
RTCHSUniversalServices
RTCProxyUniversalServices
RTCSBAUniversalServices
RTCUniversalConfigReplicator
RTCUniversalGlobalWriteGroup
RTCUniversalReadOnlyAdmins
RTCUniversalSBATechnicians
RTCUniversalServerAdmins
RTCUniversalServerReadOnlyGroup
Security Group ...
Members have read acces...
RTCUniversalUserAdmins
Members can manage RT...
RTCUniversalUserReadOnlyGroup
Lync04 RTM Properties
Dial-in
Environment
Sessions
Remote control
Remote Desktop Services Profile
Personal Virtual Desktop
COM+
General
Address
Account
Profile
Telephones
Organization
Member Of
User logon name:
Lync04
@lync2010rtm.com
User logon name (pre-Windows 2000):
LYNC2010RTM\
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:
Sunday , November 25, 2012
OK
Cancel
Apply
Help

Computers on the Active Directory running LYNC clients



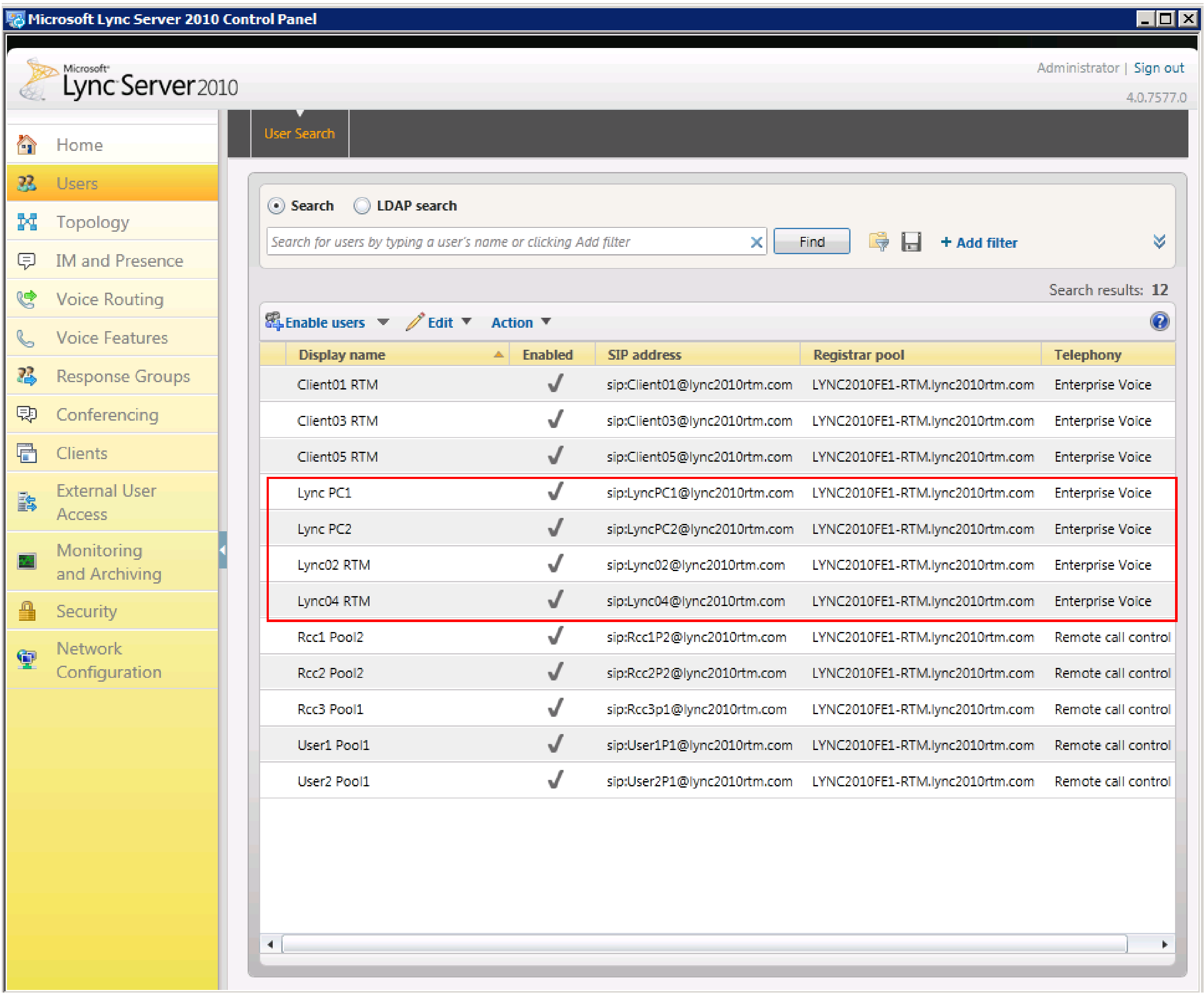
Lync Server 2010 Configuration

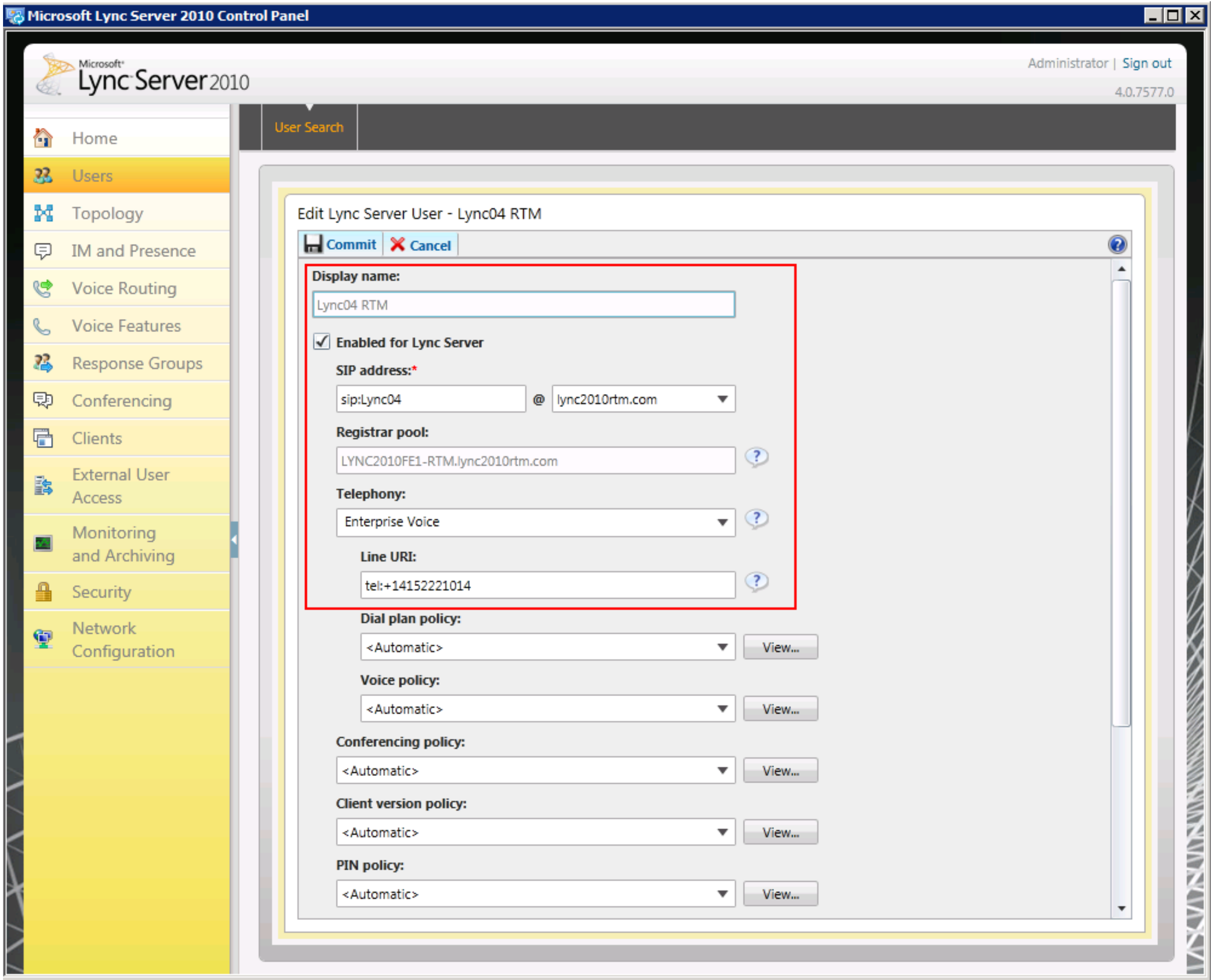
Lync Server configuration can be done in a couple of ways—through the Microsoft Lync Server Control Panel or through Lync Server Management Shell. For the purposes of configuring a direct SIP connection with CUCM, we will illustrate the configuration by using the Lync Server Control Panel and Topology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

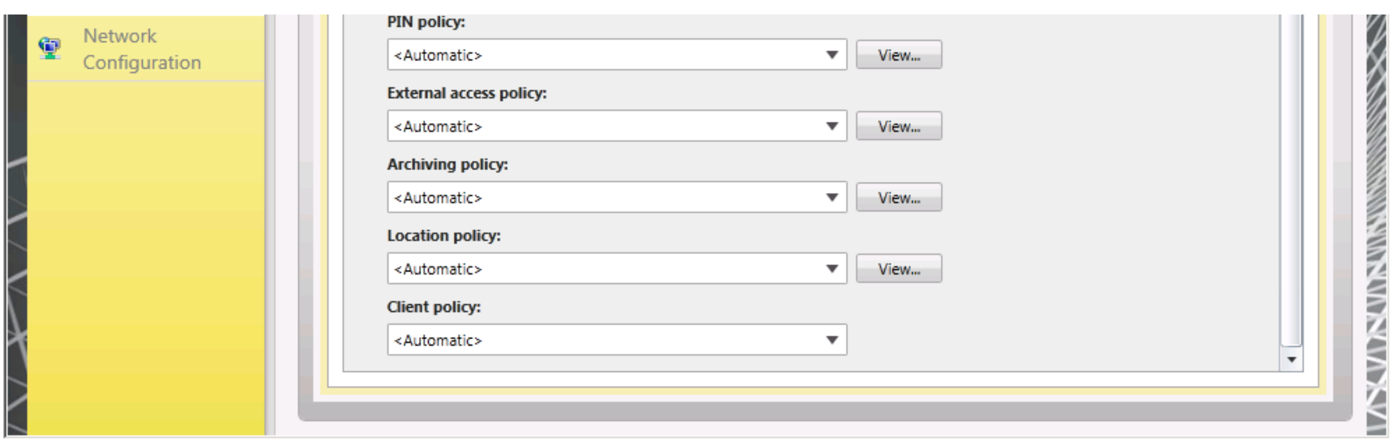
User Configuration from Control Panel

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.

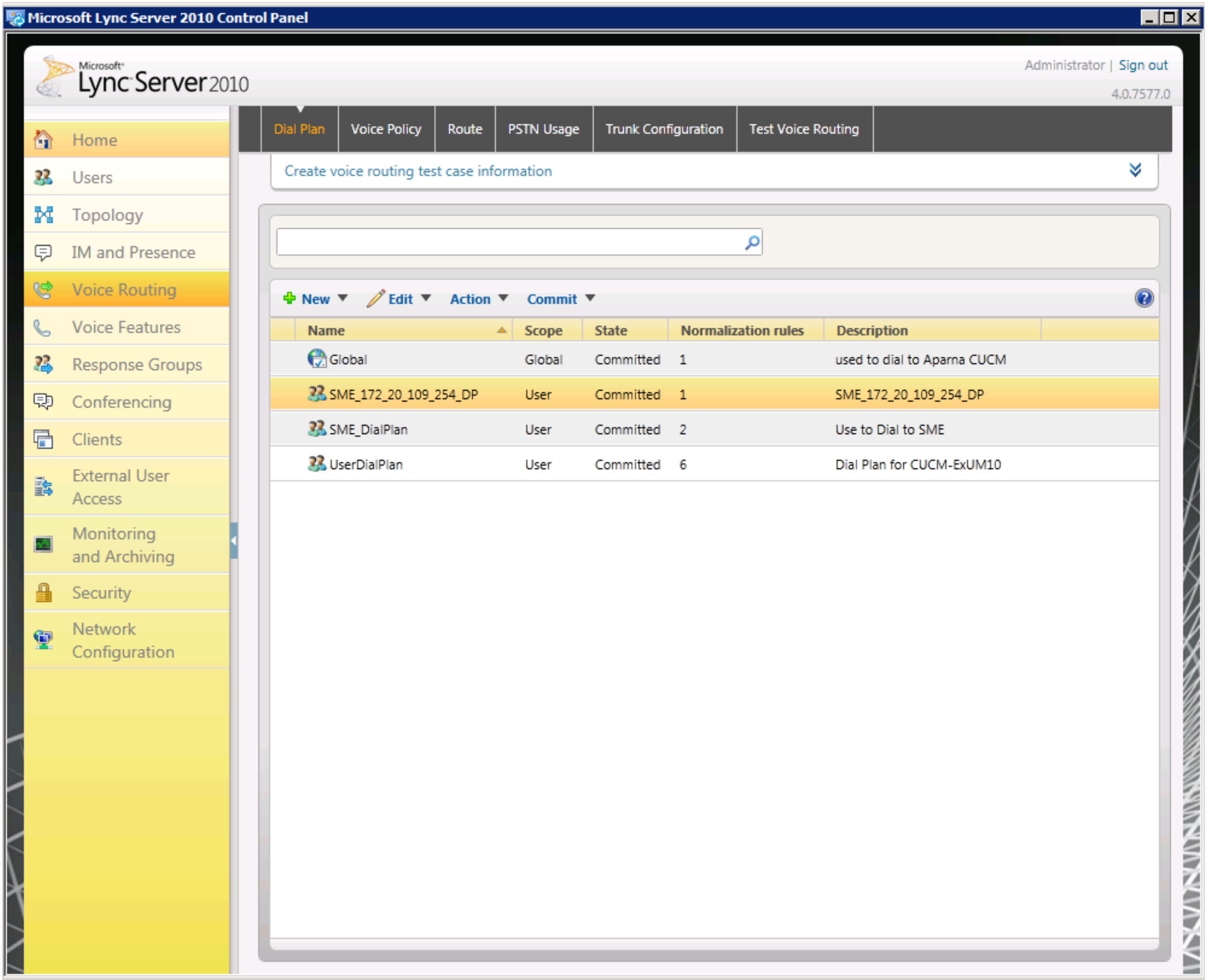



Microsoft Lync Server 2010 Control Panel
Lync Server 2010
Administrator | Sign out
4.0.7577.0
Home
Users
Topology
IM and Presence
Voice Routing
Voice Features
Response Groups
Conferencing
Clients
External User Access
Monitoring and Archiving
Security
Network Configuration
User Search
Edit Lync Server User - Lync04 RTM
Commit
Cancel
Display name:
Lync04 RTM
Enabled for Lync Server
SIP address:*
sip:Lync04
@
lync2010rtm.com
Registrar pool:
LYNC2010FE1-RTM.lync2010rtm.com
Telephony:
Enterprise Voice
Line URI:
tel:+14152221014
Dial plan policy:
<Automatic>
View...
Voice policy:
Conferencing policy:
Client version policy:
PIN policy:

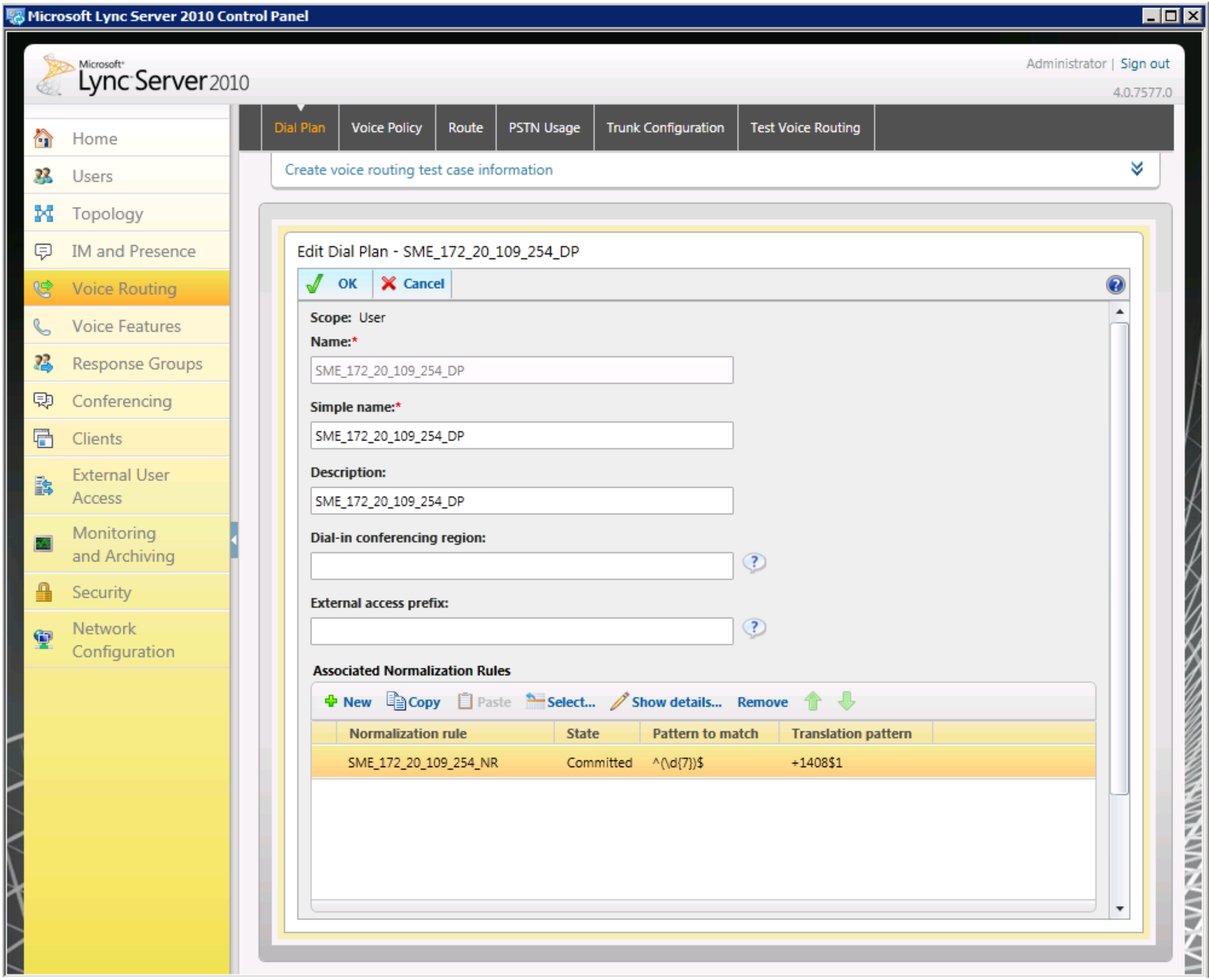

Network Configuration
PIN policy:
<Automatic>
View...
External access policy:
Archiving policy:
Location policy:
Client policy:

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

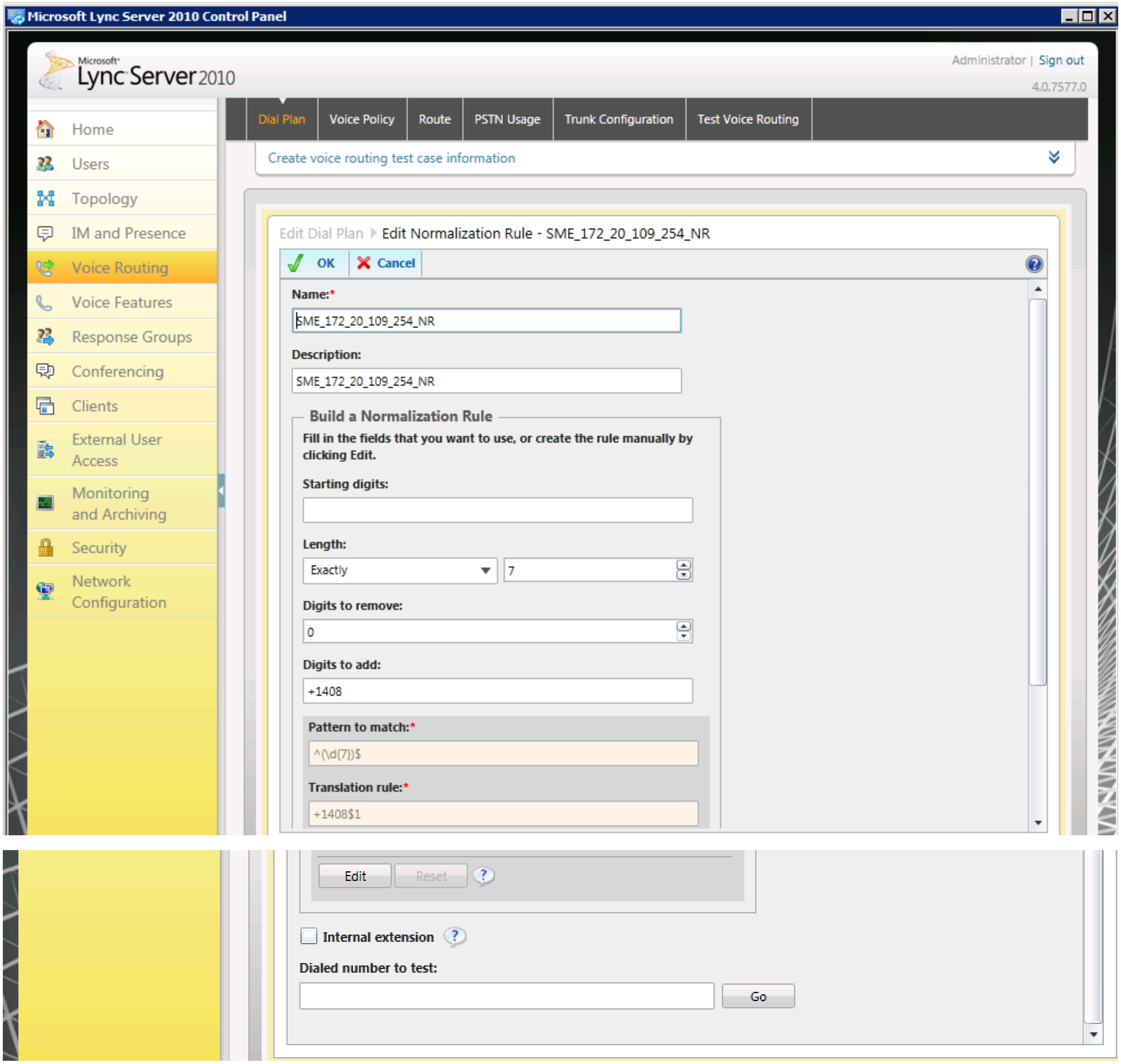


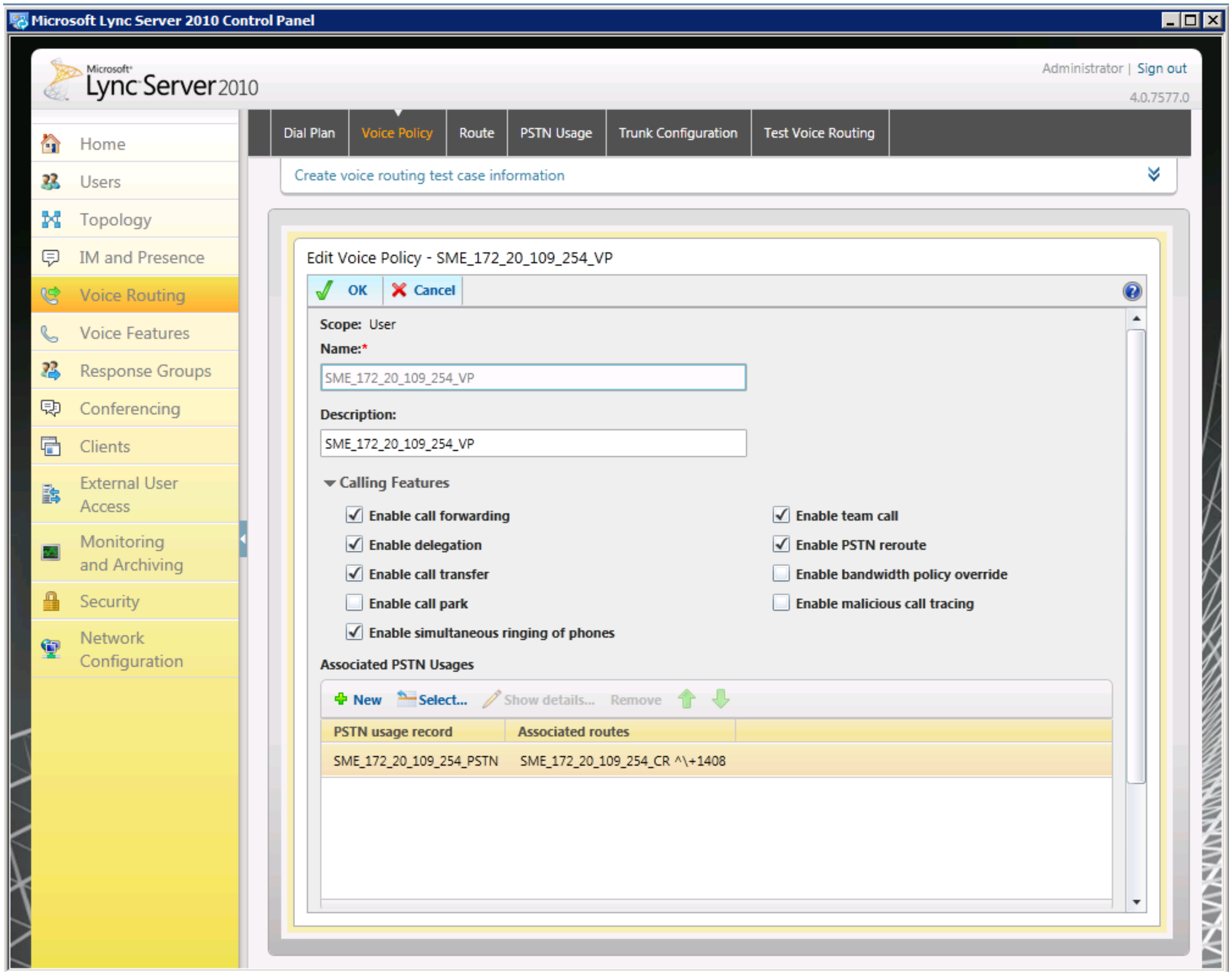
Click on the dial plan “SME_172_20_109_254_DP” in the above figure.



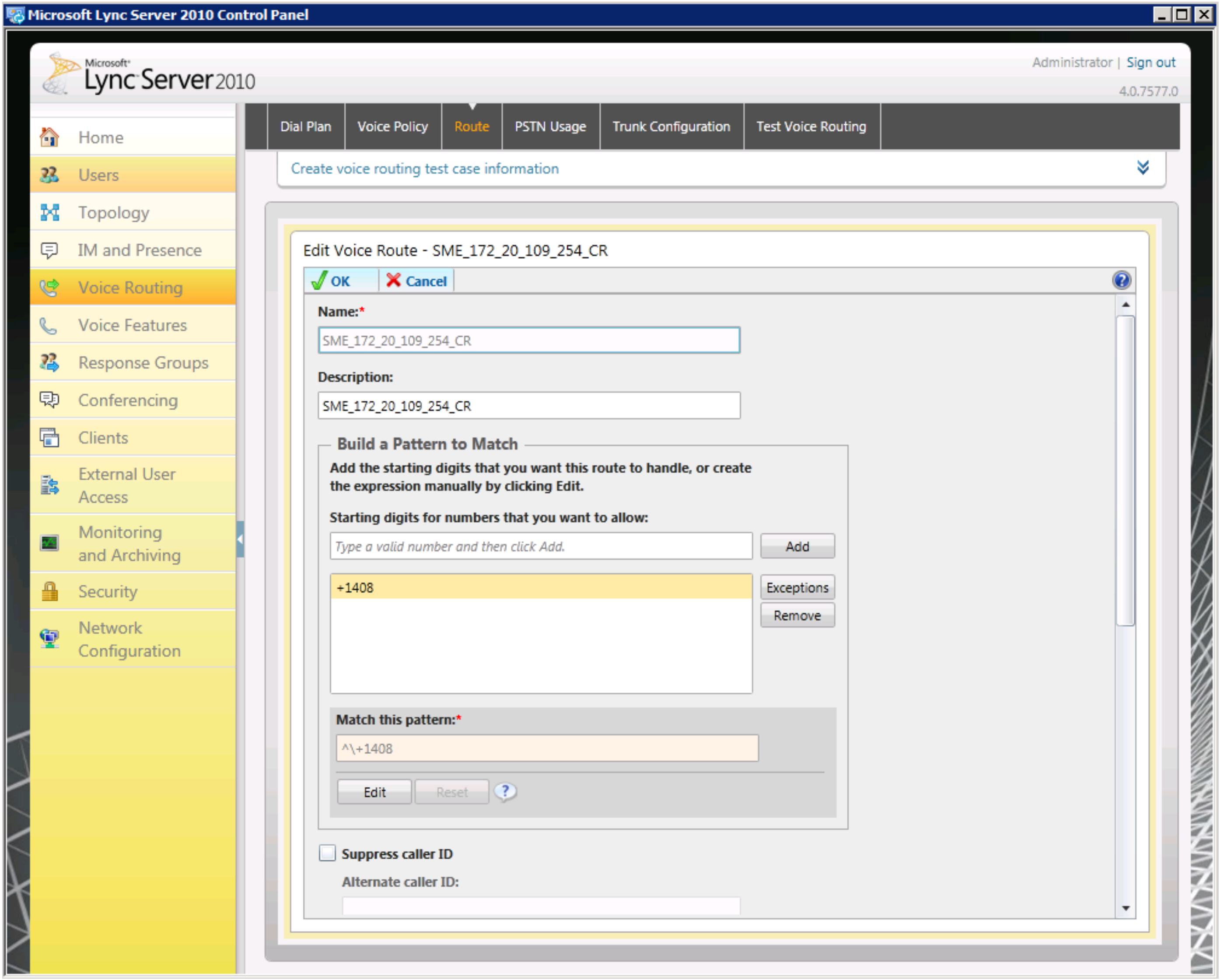
Lync Server normalizes all outbound calls as E.164. This allows uniform routing that scales globally across the Lync Server deployment. Phone numbers are normalized to E.164 by normalization rules in this case the rule "SME_172_20_109_254_NR". Normalization rules are added in the Dial Plan tab.

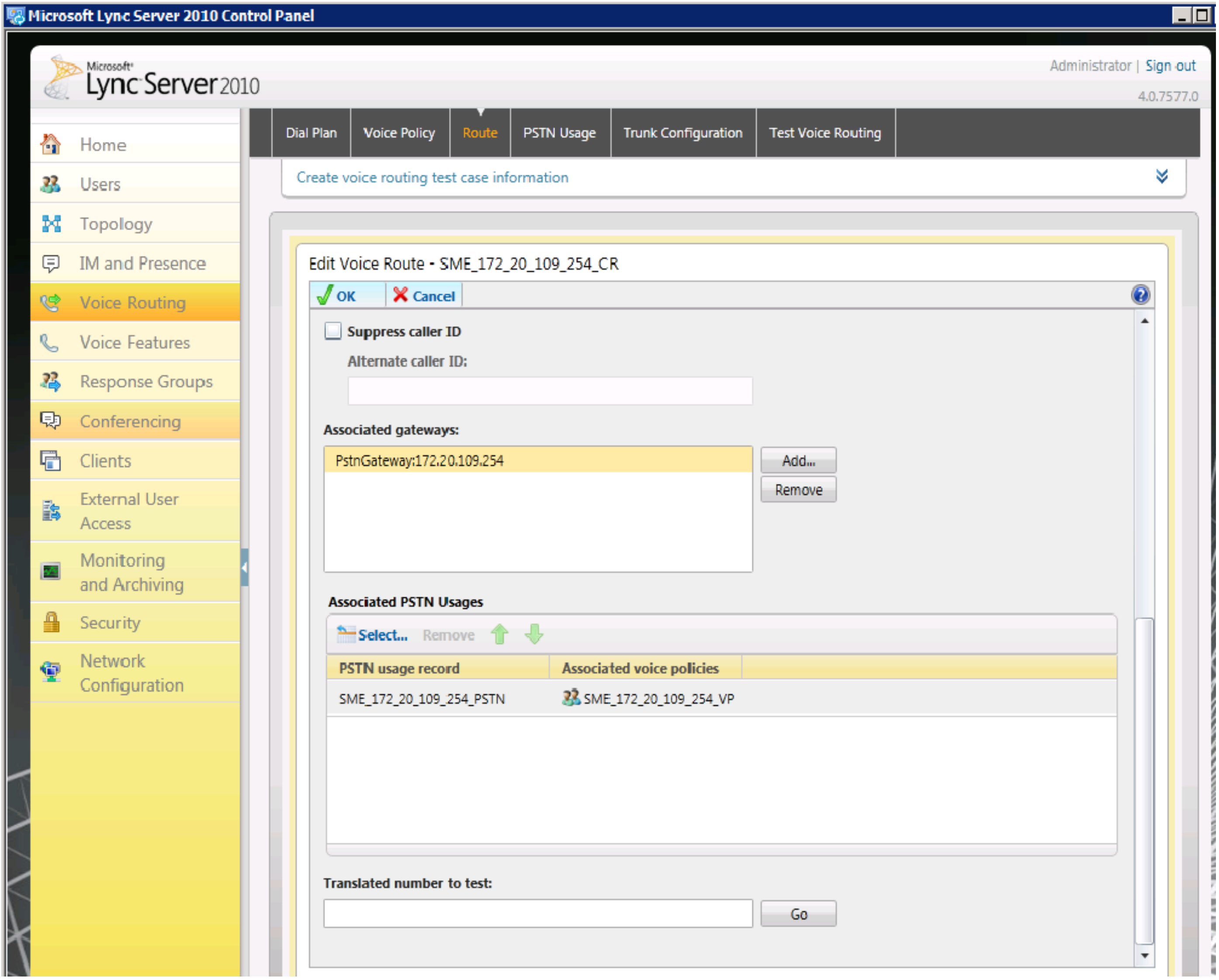
Click on the Normalization Rule "SME_172_20_109_254_NR" in the above figure.


Microsoft Lync Server 2010 Control Panel
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 Dial Plan
Edit Normalization Rule - SME_172_20_109_254_NR
OK
Cancel
Name:*
SME_172_20_109_254_NR
Description:
SME_172_20_109_254_NR
Build a Normalization Rule
Fill in the fields that you want to use, or create the rule manually by clicking Edit.
Starting digits:
Length:
Exactly
7
Digits to remove:
0
Digits to add:
+1408
Pattern to match:*
^(\d{7})$
Translation rule:*
+1408$1
Edit
Reset
Internal extension
Dialed number to test:
Go



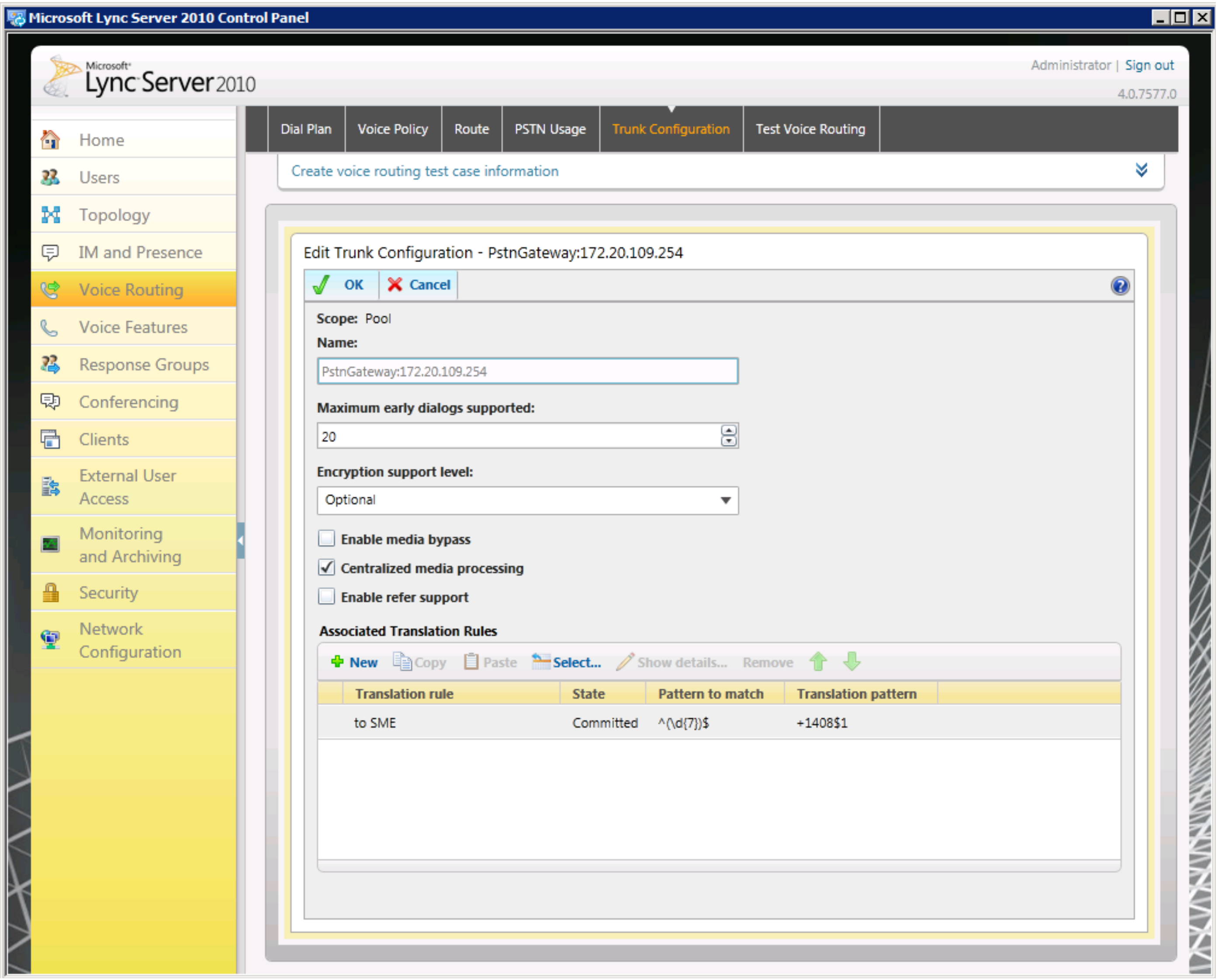
PSTN usage is associated on the Voice Policy. Calling features are selected in Voice policy tab. New user policy is created to dial to the SME.


Microsoft Lync Server 2010 Control Panel
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 Voice Route - SME_172_20_109_254_CR
OK
Cancel
Name:*
SME_172_20_109_254_CR
Description:
SME_172_20_109_254_CR
Build a Pattern to Match
Add the starting digits that you want this route to handle, or create the expression manually by clicking Edit.
Starting digits for numbers that you want to allow:
Type a valid number and then click Add.
Add
+1408
Exceptions
Remove
Match this pattern:*
^\+1408
Edit
Reset
Suppress caller ID
Alternate caller ID:


Microsoft Lync Server 2010 Control Panel
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 Voice Route - SME_172_20_109_254_CR
OK
Cancel
Suppress caller ID
Alternate caller ID:
Associated gateways:
PstnGateway:172.20.109.254
Add...
Remove
Associated PSTN Usages
Select...
Remove
PSTN usage record
Associated voice policies
SME_172_20_109_254_PSTN
SME_172_20_109_254_VP
Translated number to test:
Go

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.



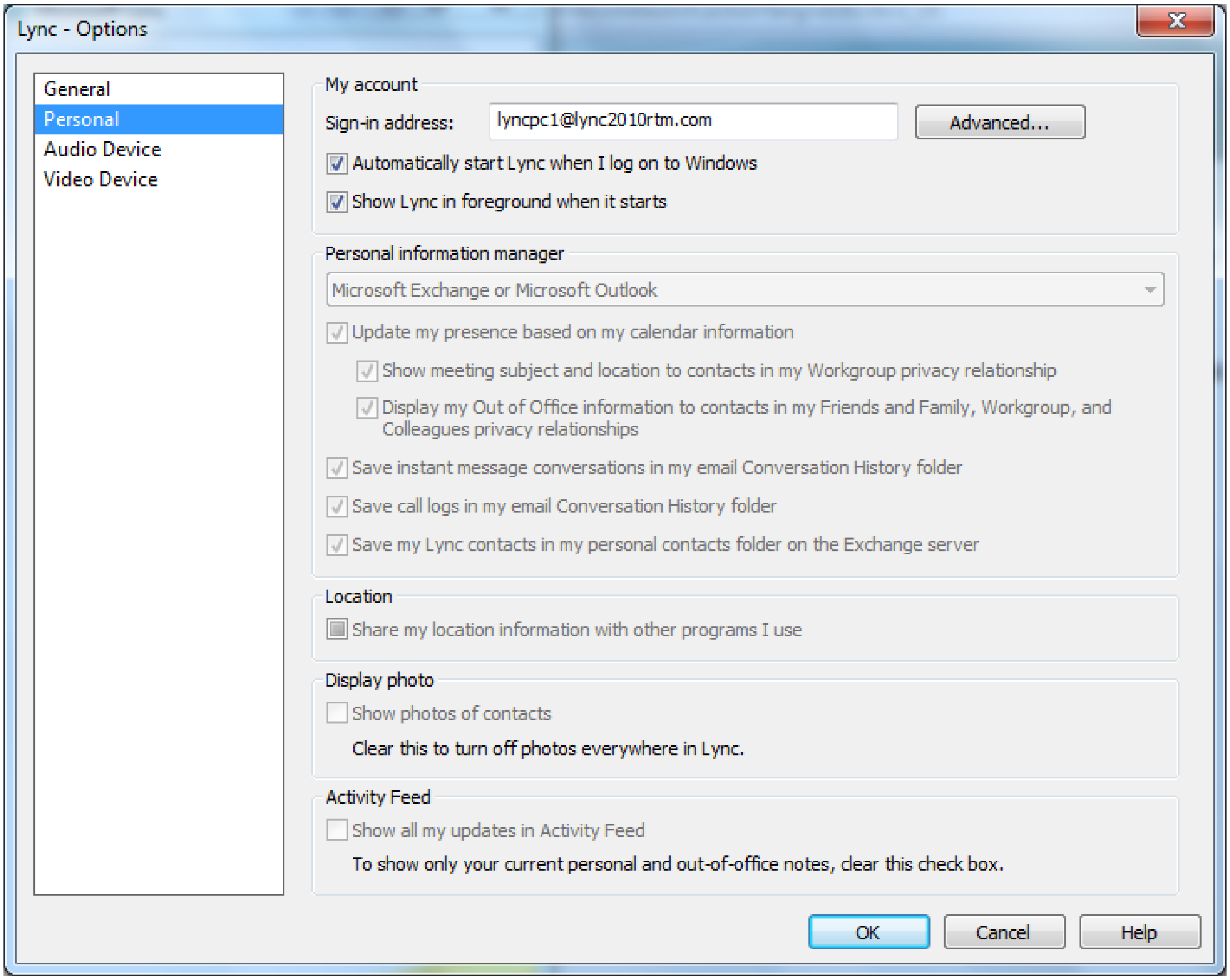
Lync 2010 Server Management Shell Commands

Lync 2010 Server draining mode
Stop-CsWindowsService -Graceful rtcmedsrv

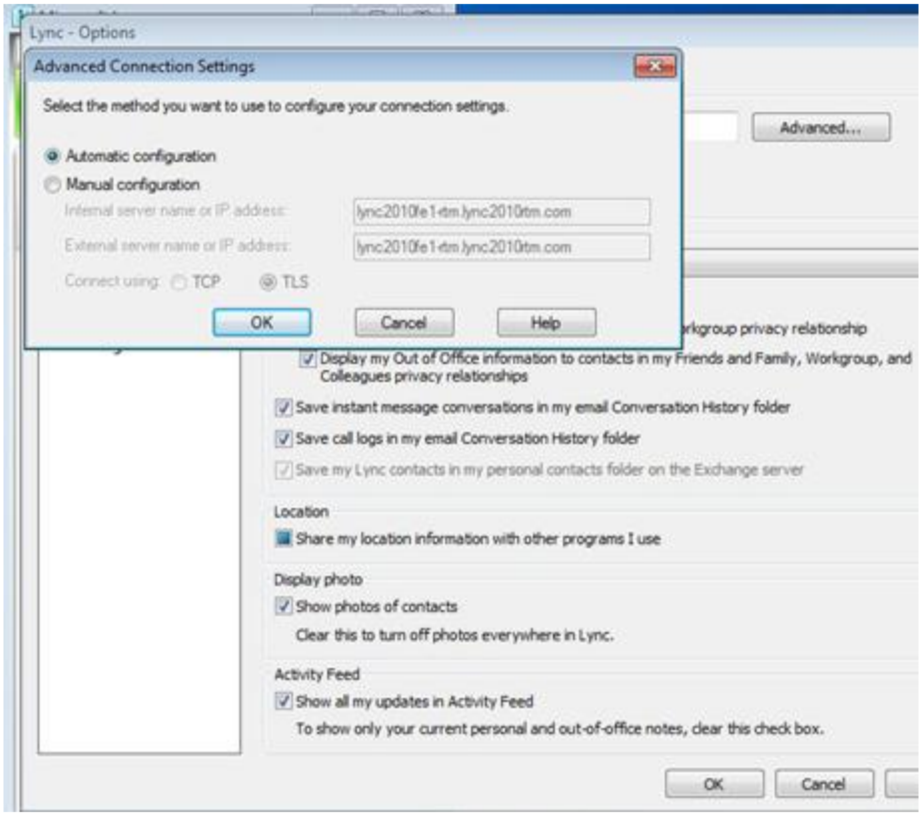
Lync2010 - enable music on hold:
set-csclientpolicy -EnableClientMusicOnHold $TRUE

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

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

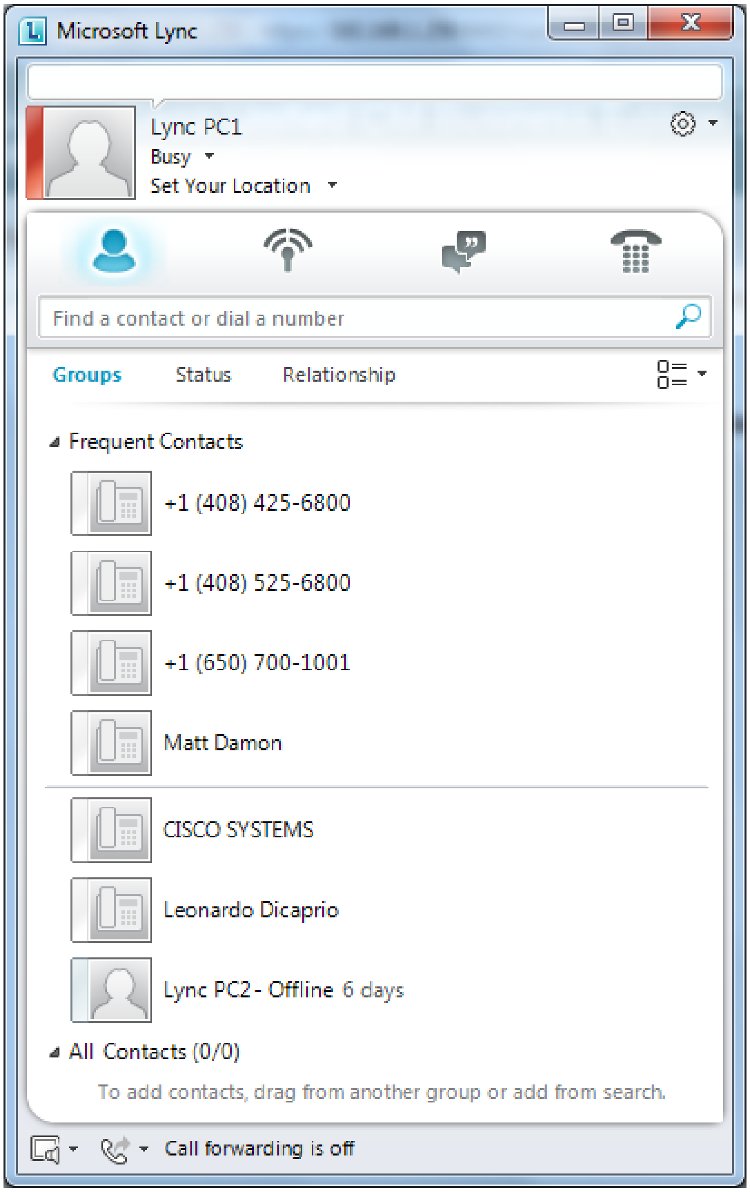


Click Advanced button to select the Advanced Connection Settings.

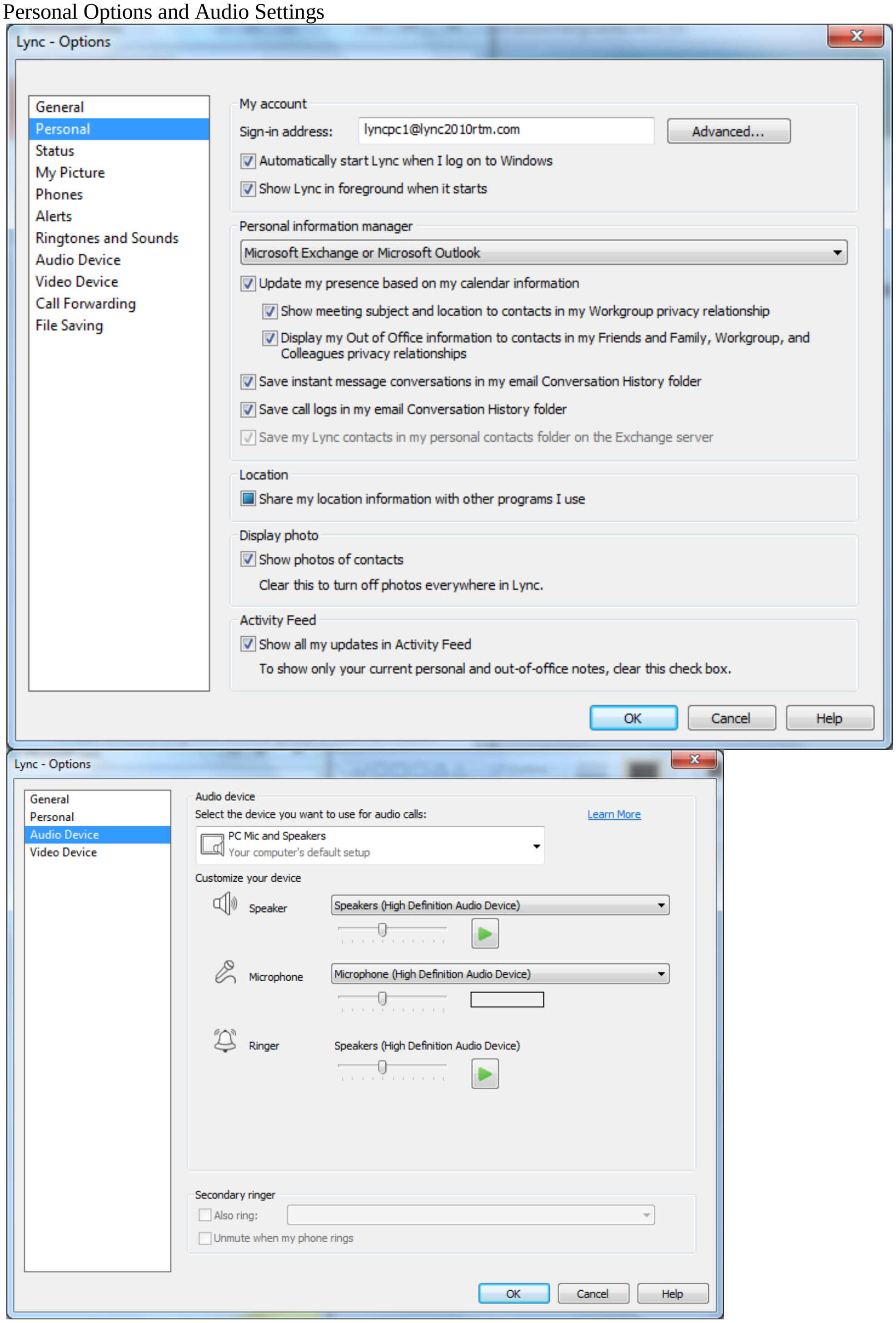


If there are DNS entries for this Microsoft Lync, automatic configuration can be used if not select manual configuration.

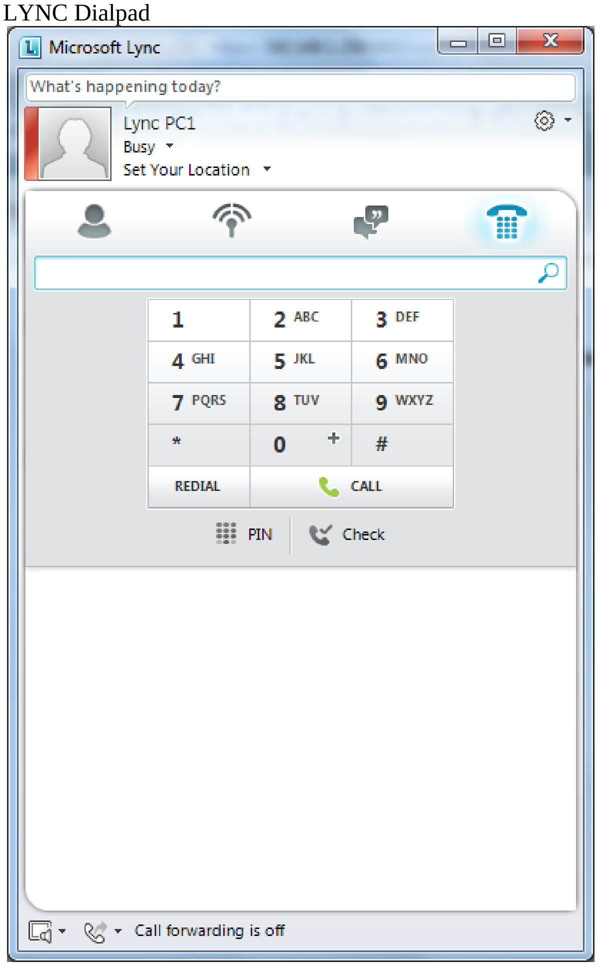
After signing add contacts



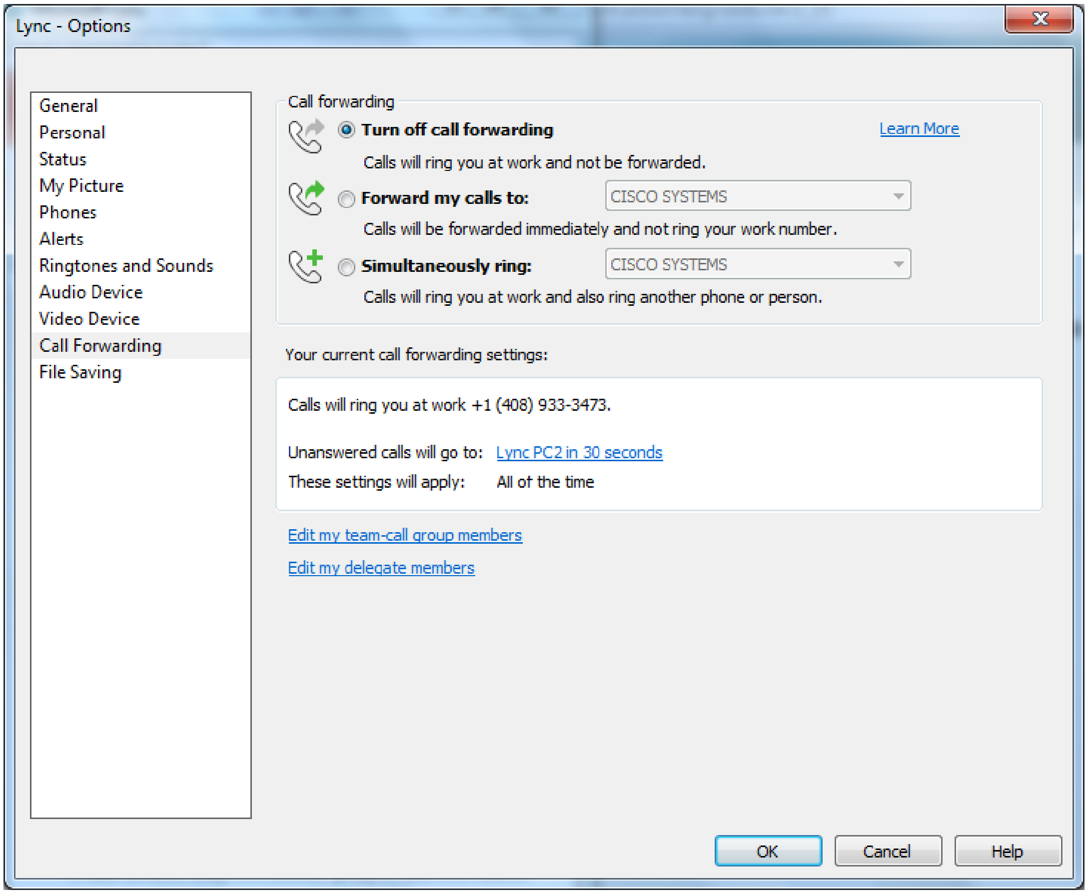
Personal Options and Audio Settings



LYNC Dialpad



Call Forwarding Configurations



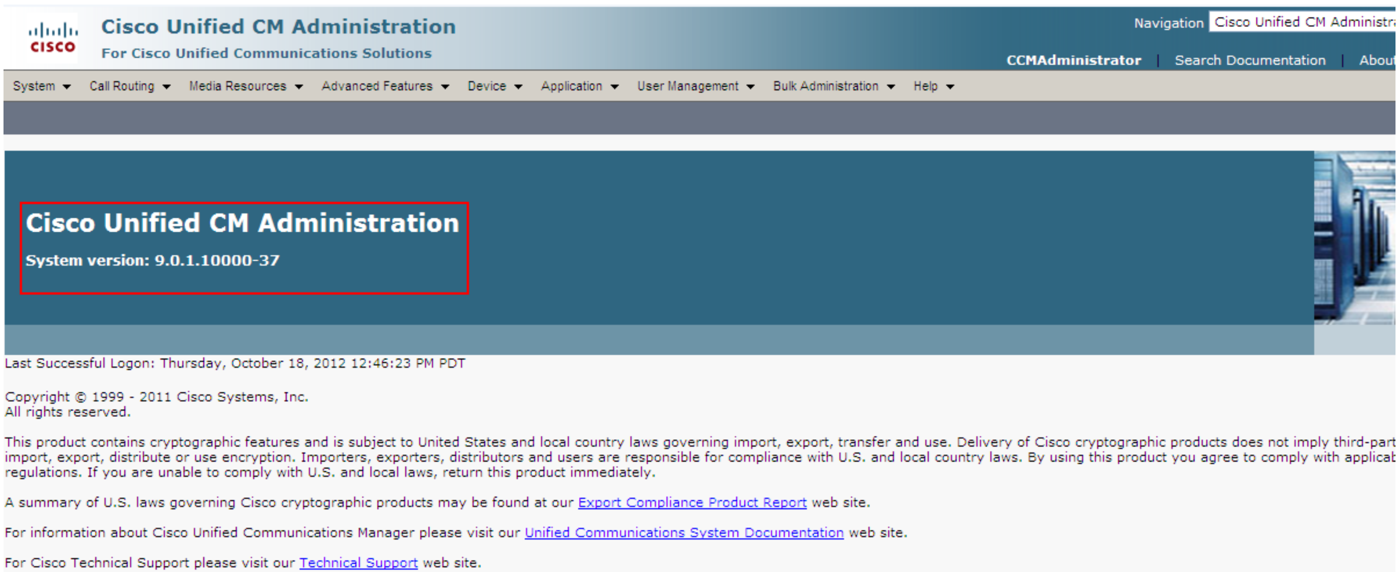
All call forward settings are selected in this tab. You can turn on/off the call forwarding settings and also select for phones to simultaneously ring.

Cisco Unified Communications Manager- Session Manager Edition Configurations

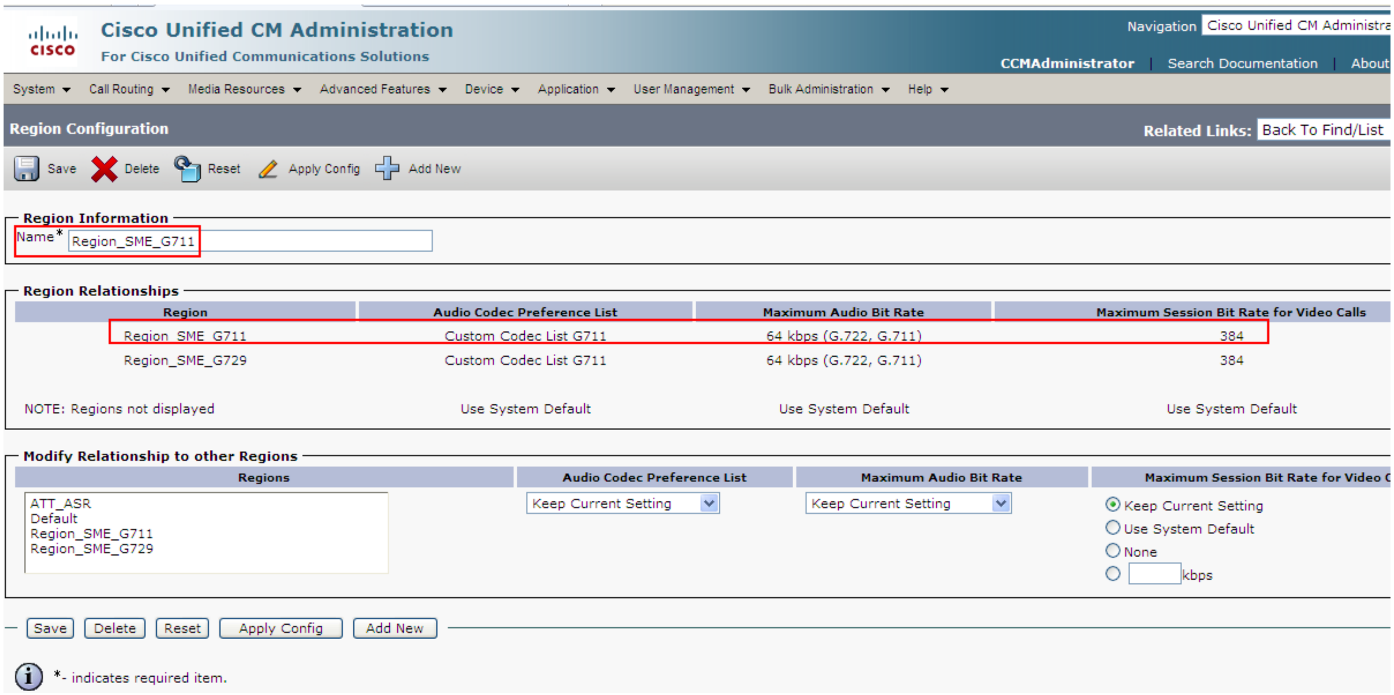
UC deployments using Unified CM Session Manager Edition are a variation on the Multi-site Distributed Call Processing deployment model and are typically used where large numbers of UC end systems need to be interconnected via a single UC system - i.e. The Unified CM Session Manager. A deployment using 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-SME Software release

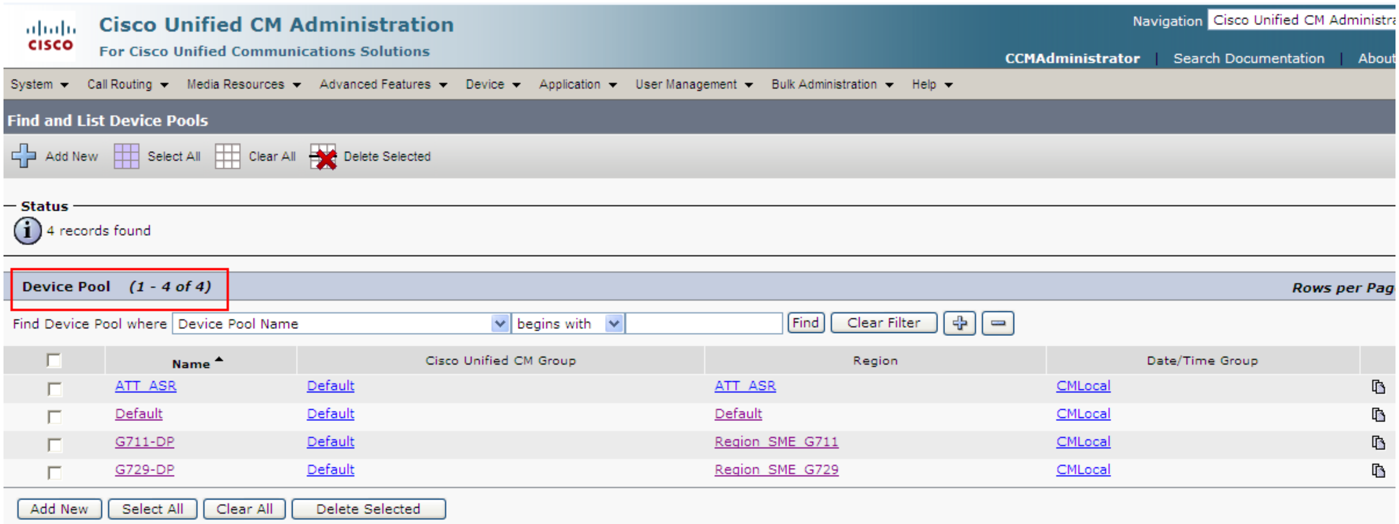


Cisco Unified Communications Manager-SME Region Configuration

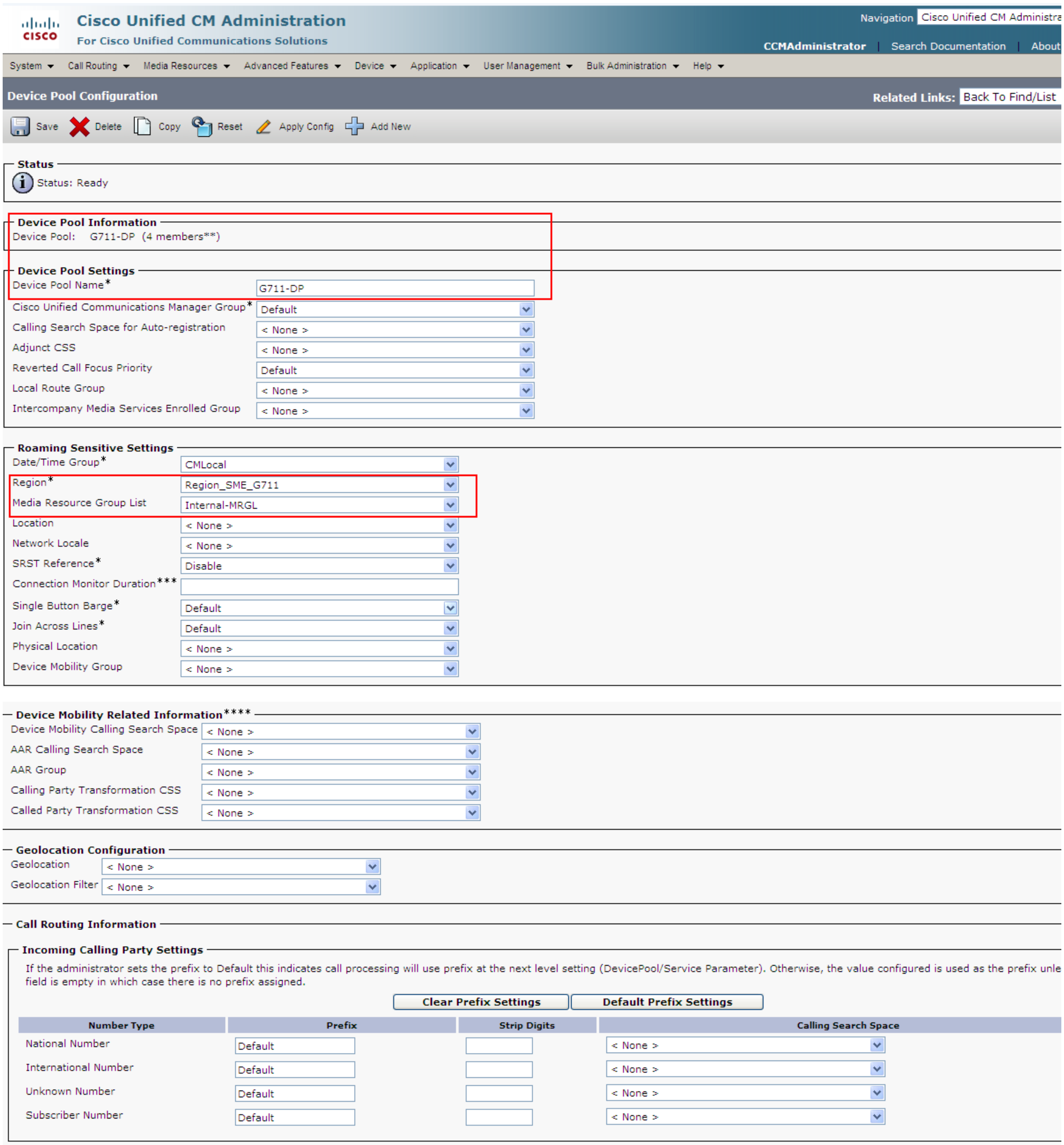


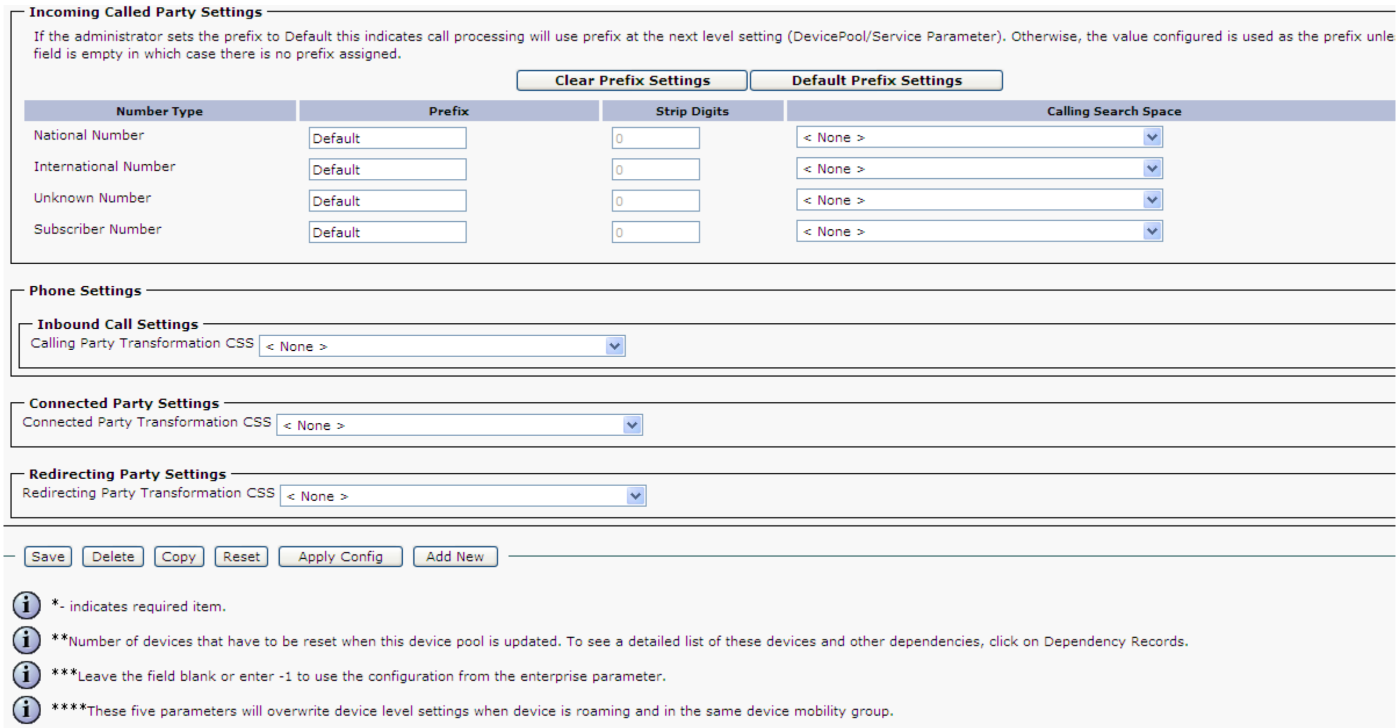
Region configurations are used to determine the codecs used by the endpoint within the region and across the region.

Cisco Unified Communications Manager-SME Device Pool Configuration



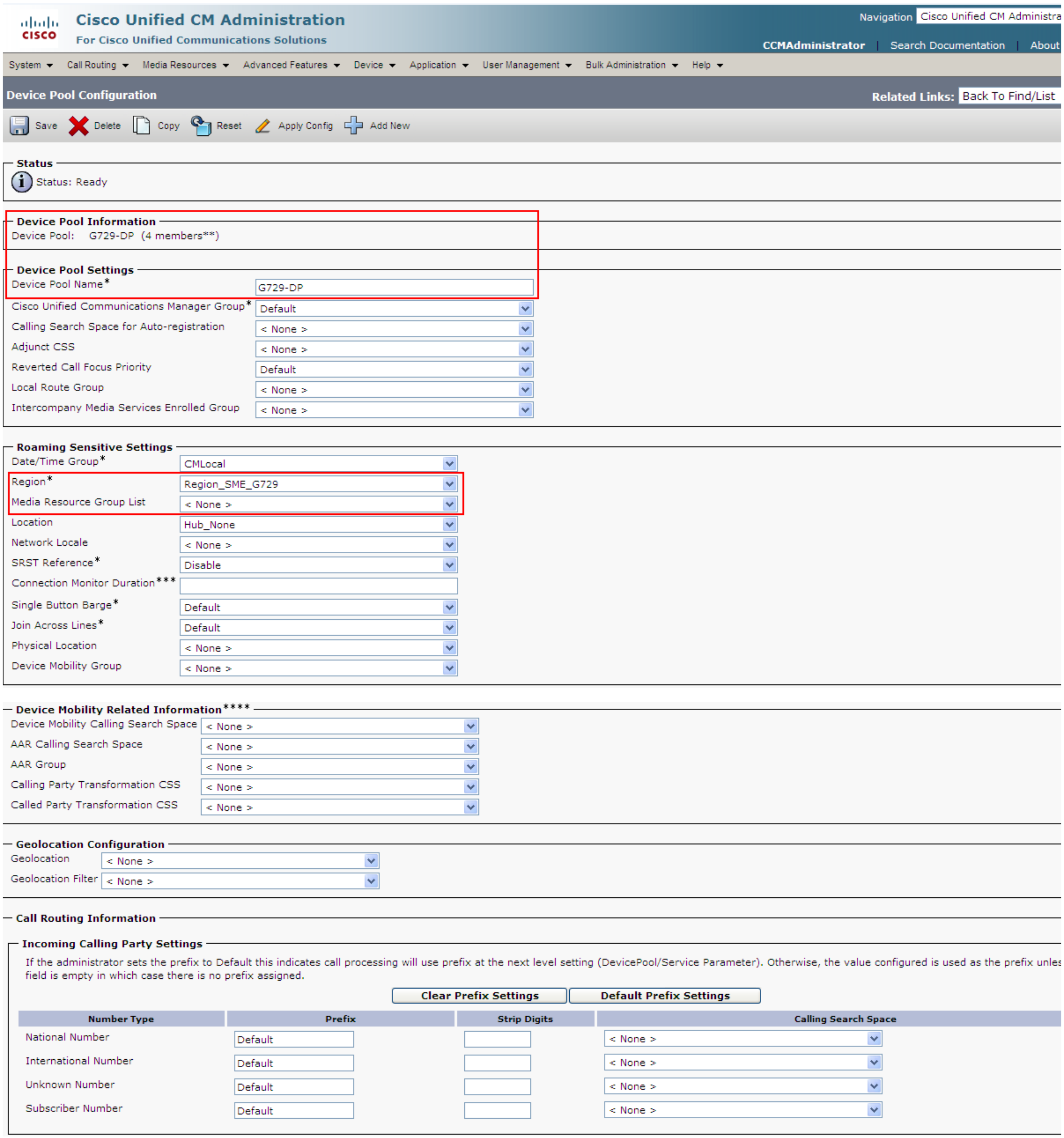
Cisco Unified Communications Manager-SME Device Pool Configuration-G711-DP





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

Cisco Unified Communications Manager – SME – Device Pool Configuration-G729



Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unles 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	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >

Phone Settings

Inbound Call Settings

Calling Party Transformation CSS: < None >

Connected Party Settings

Connected Party Transformation CSS: < None >

Redirecting Party Settings

Redirecting Party Transformation CSS: < None >

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

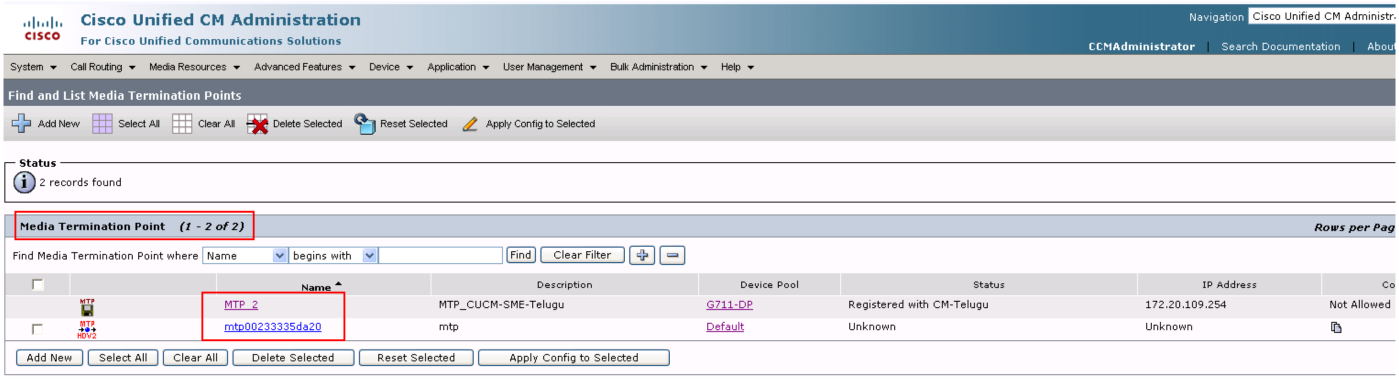
*- indicates required item.

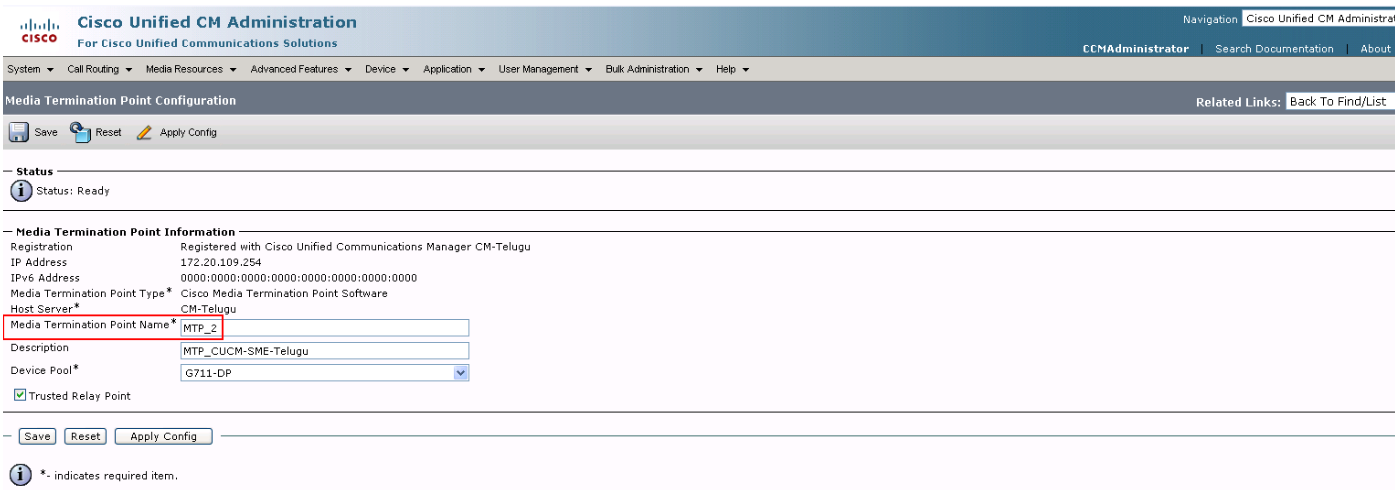
**Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.

***Leave the field blank or enter -1 to use the configuration from the enterprise parameter.

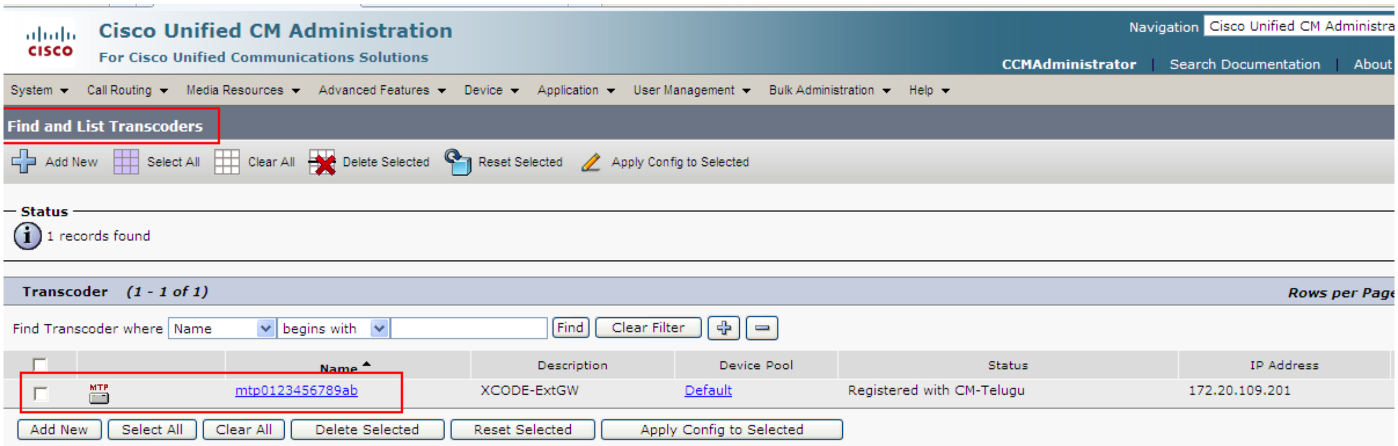
****These five parameters will overwrite device level settings when device is roaming and in the same device mobility group.

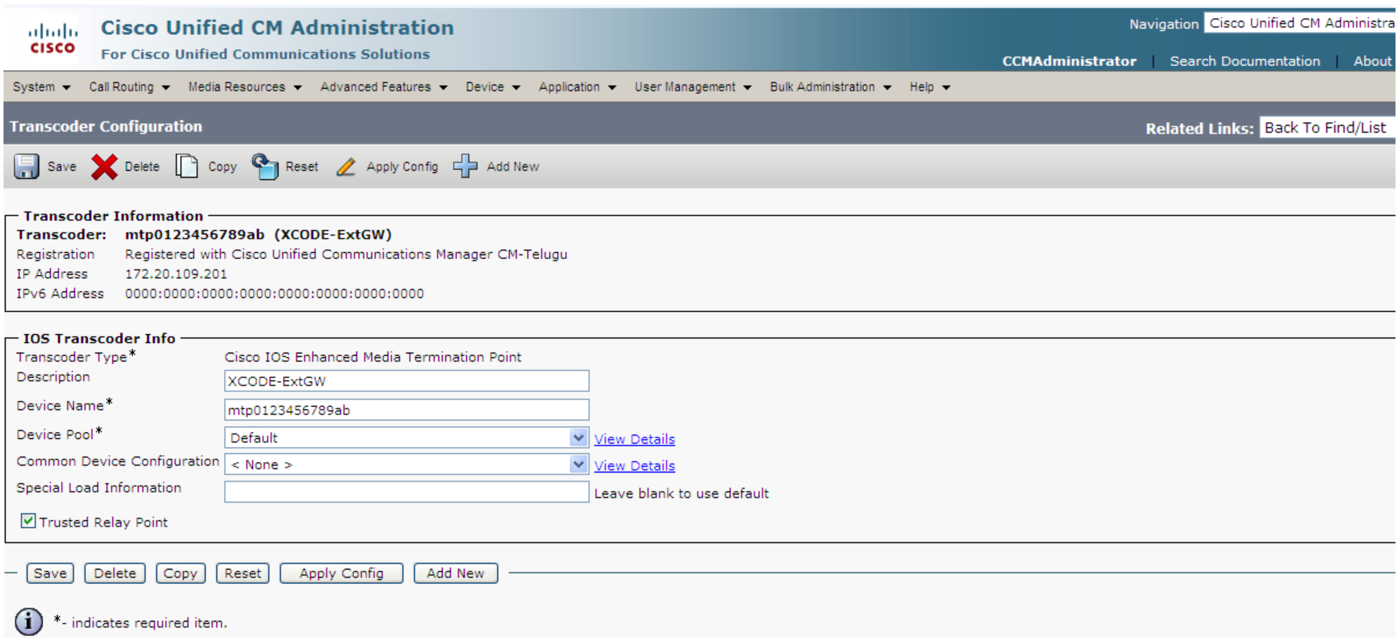
Cisco Unified Communications Manager-SME Media Termination Point Configuration



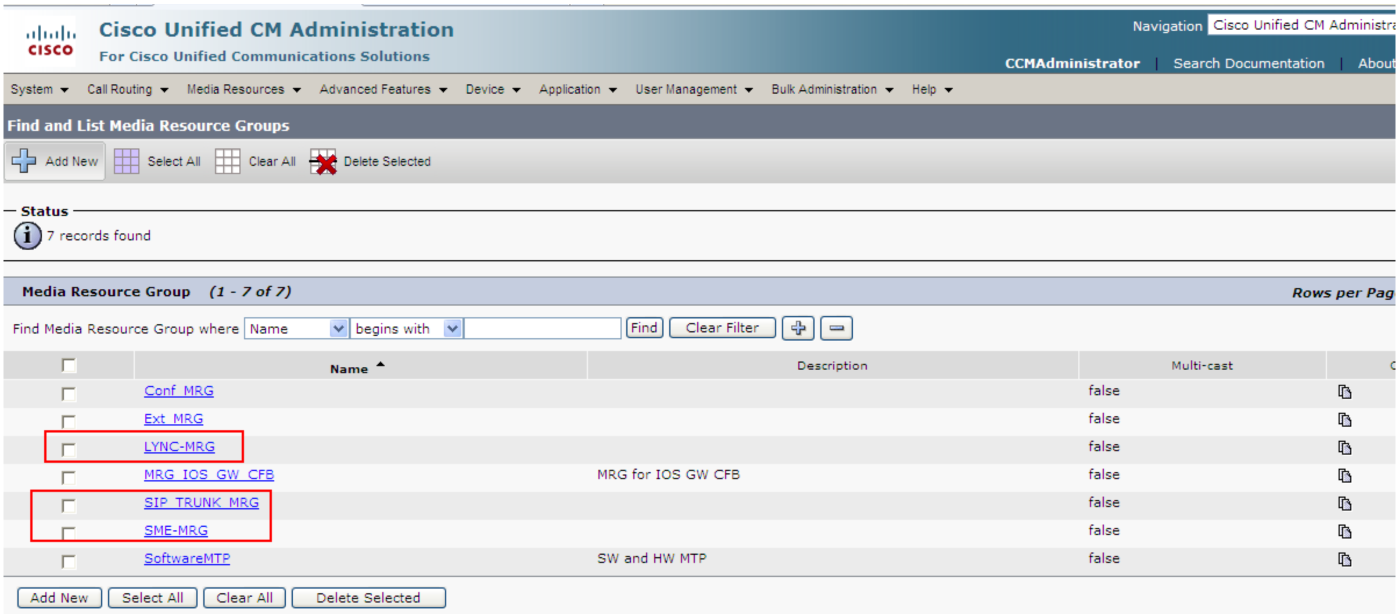


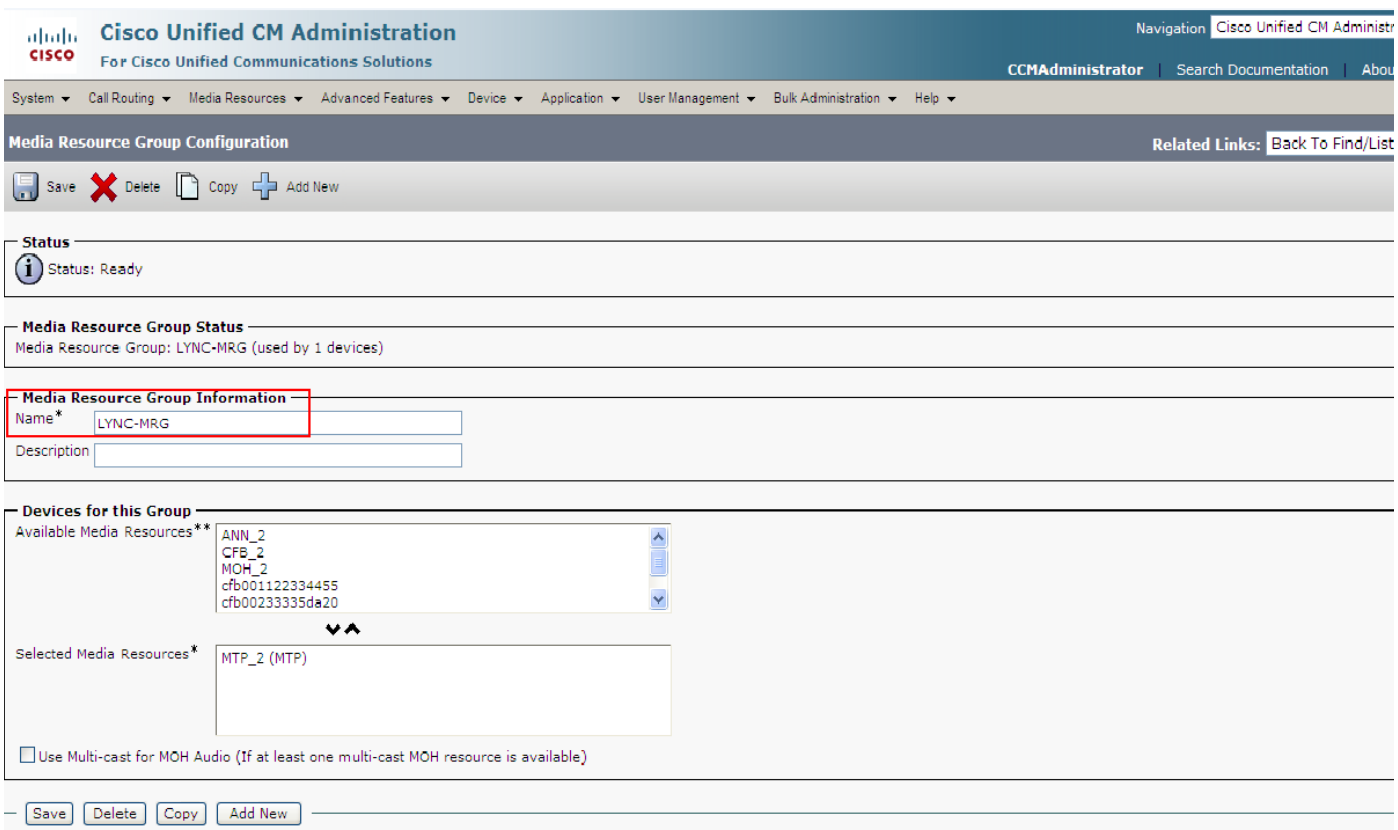
Cisco Unified Communications Manager-SME Transcoder Configuration

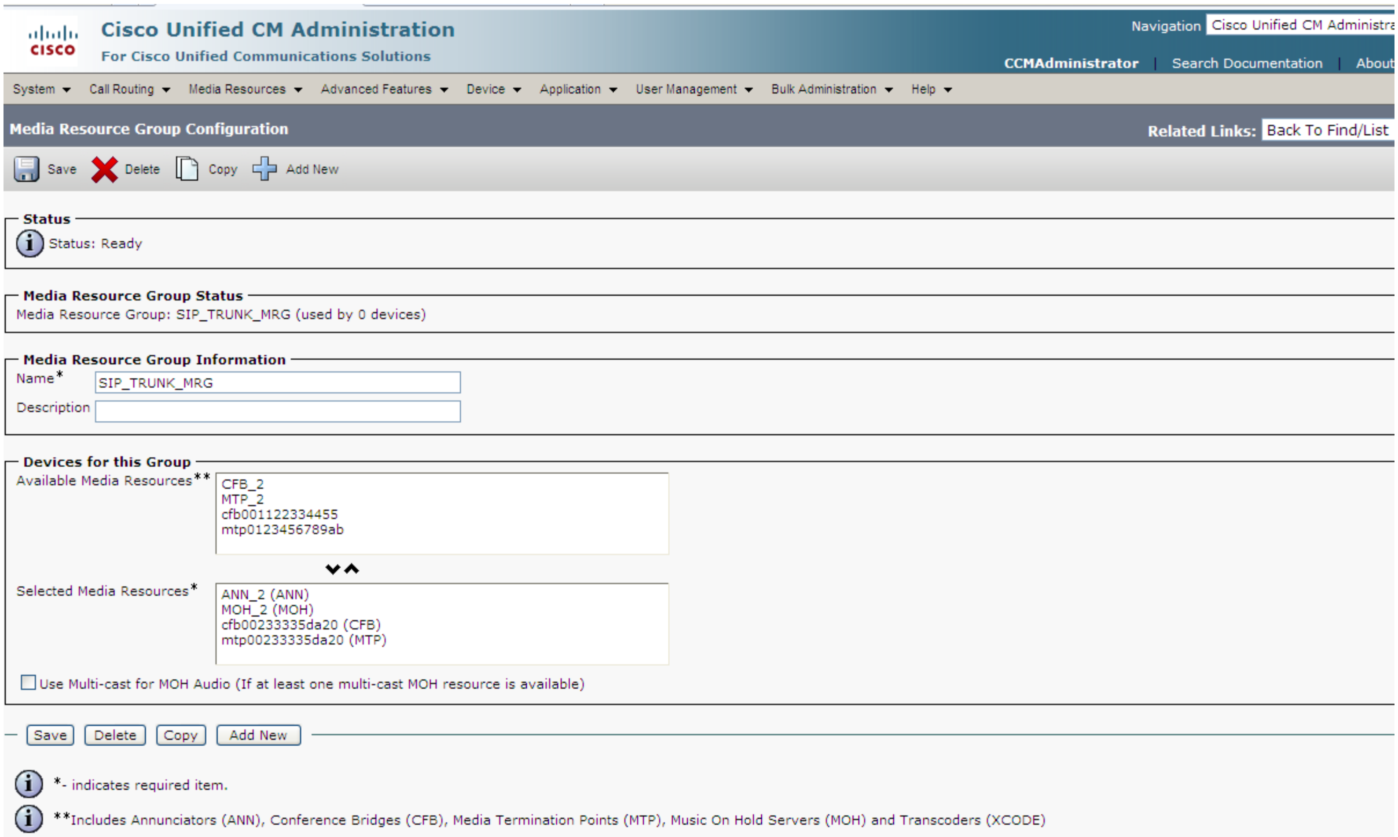




Cisco Unified Communications Manager-SME Media Resource Groups configuration



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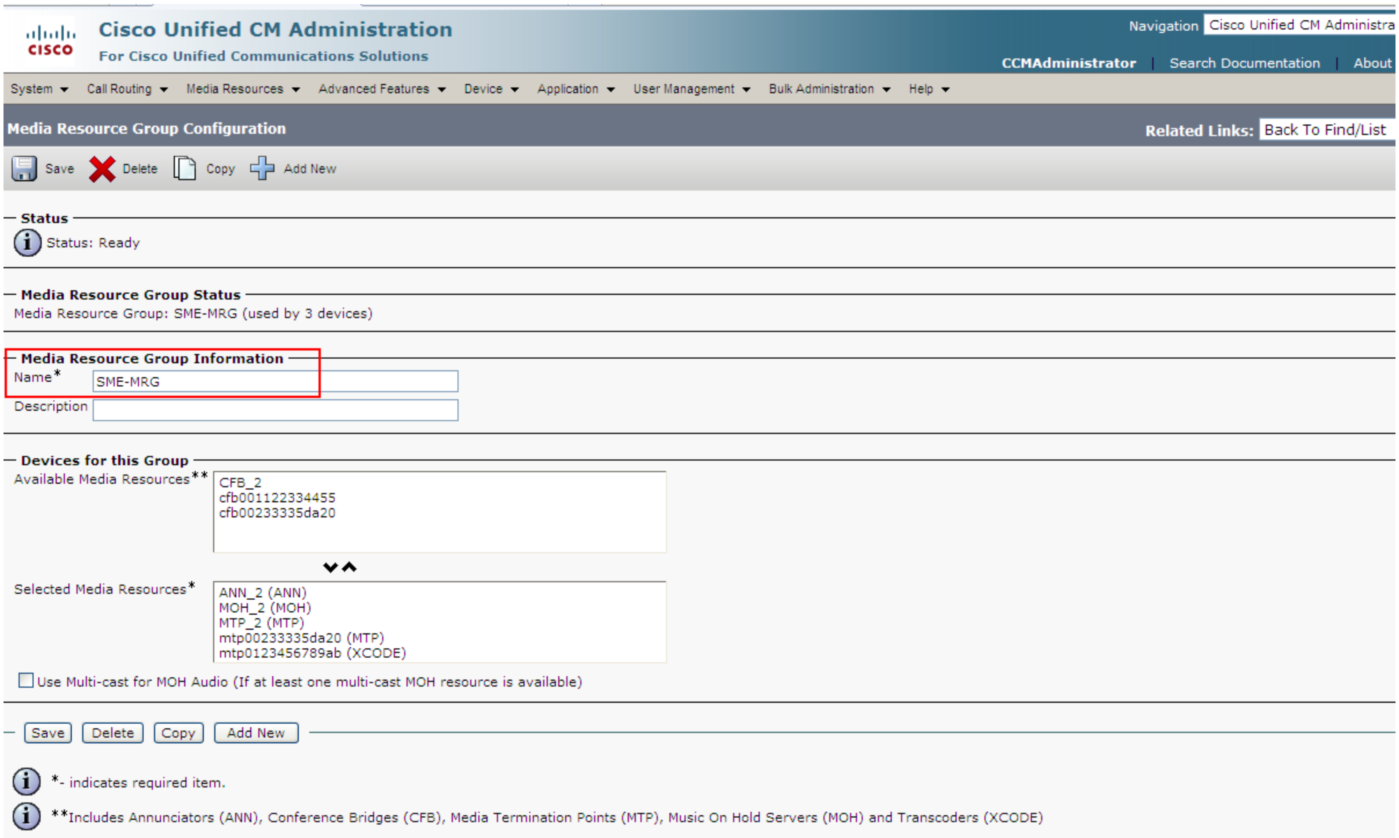
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Media Resource Group Configuration
Related Links: Back To Find/List
Save Delete Copy Add New
Status
Status: Ready
Media Resource Group Status
Media Resource Group: SIP_TRUNK_MRG (used by 0 devices)
Media Resource Group Information
Name*
SIP_TRUNK_MRG
Description
Devices for this Group
Available Media Resources**
CFB_2
MTP_2
cfb001122334455
mtp0123456789ab
Selected Media Resources*
ANN_2 (ANN)
MOH_2 (MOH)
cfb00233335da20 (CFB)
mtp00233335da20 (MTP)
Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)
Save Delete Copy Add New
*- indicates required item.
**Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)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Application
User Management
Bulk Administration
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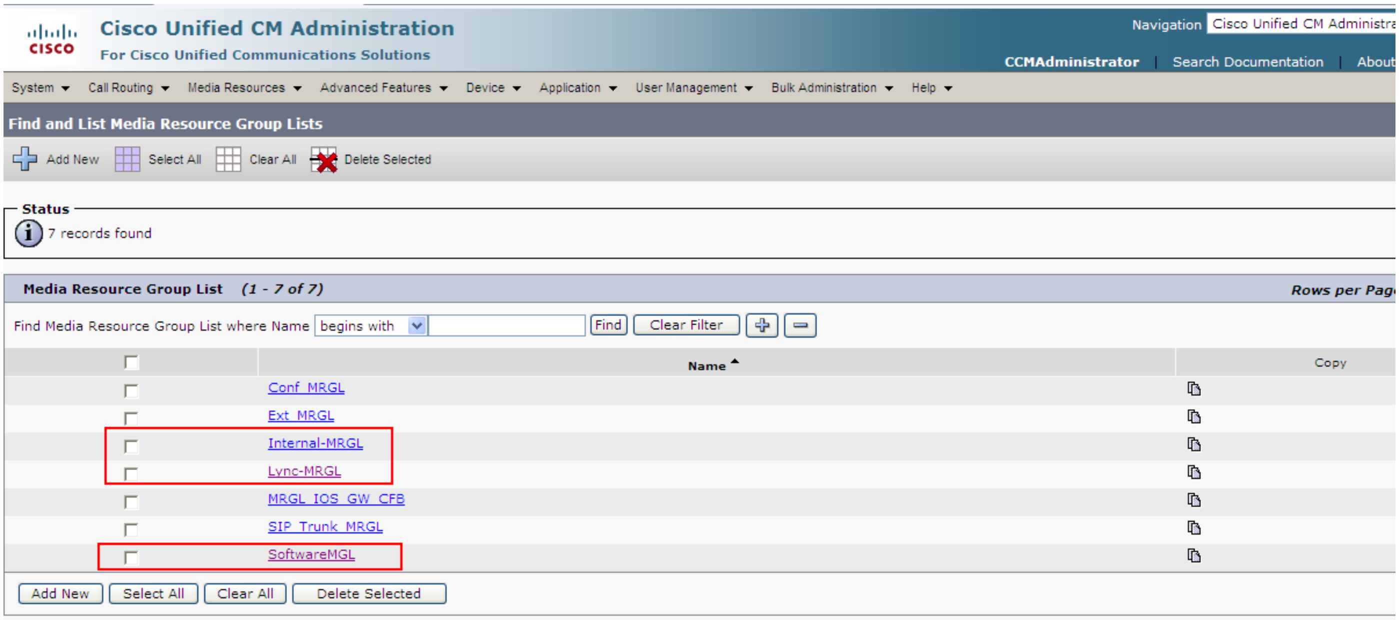
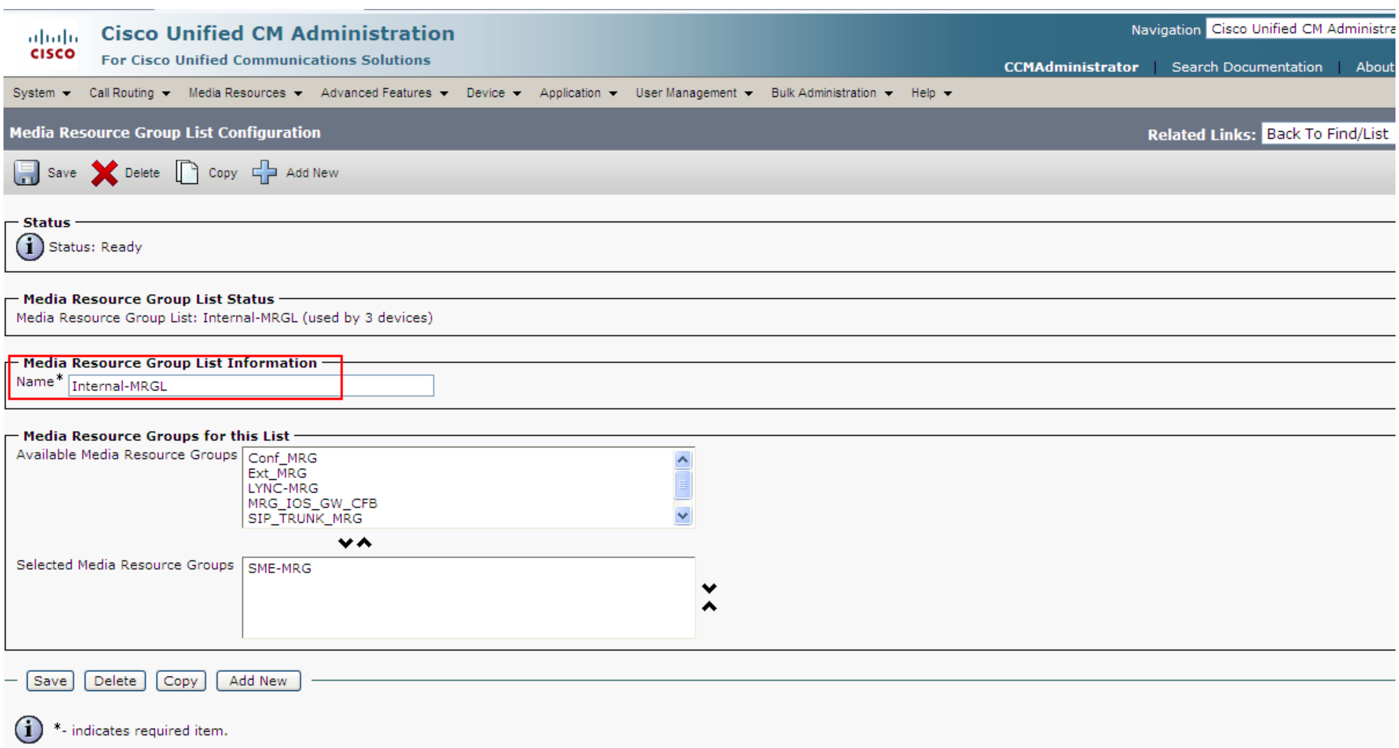
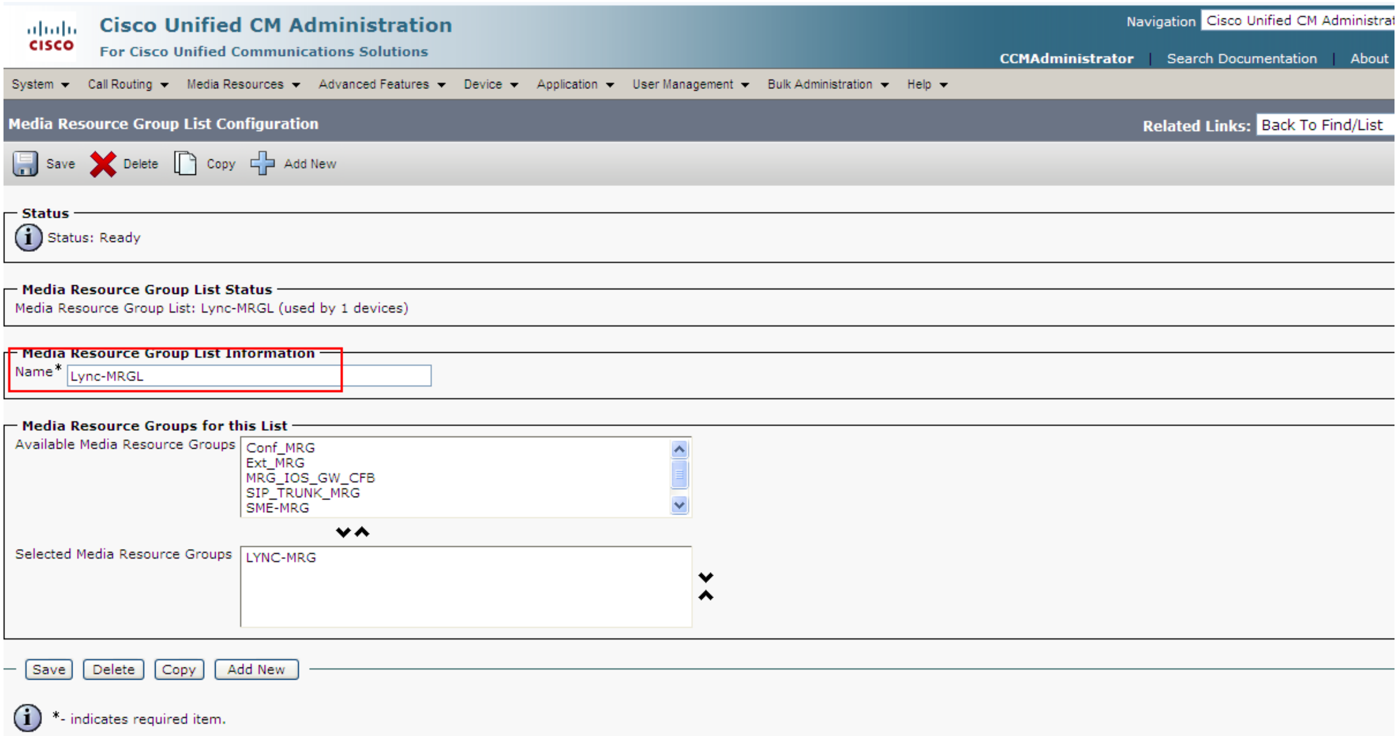
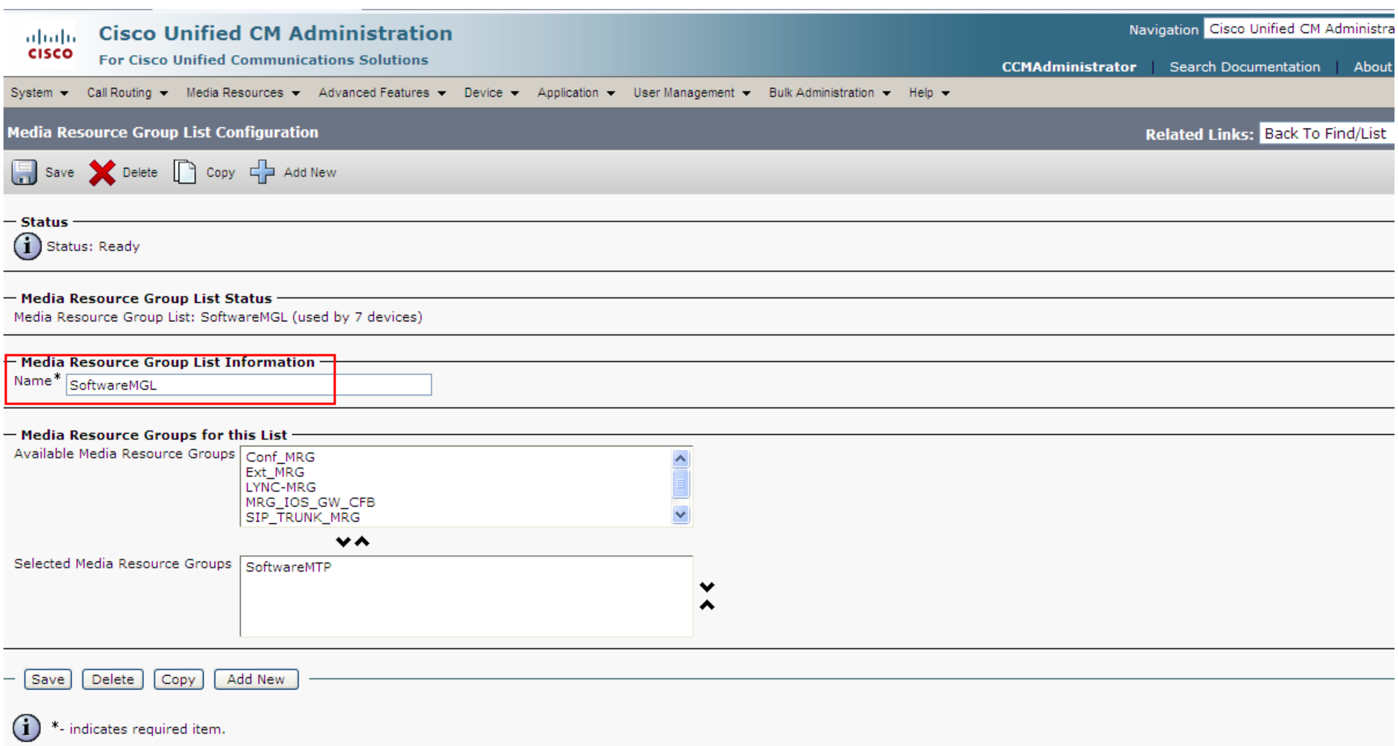
Related Links:
Back To Find/List
Save
Delete
Copy
Add New
Status
Status: Ready
Media Resource Group Status
Media Resource Group: SME-MRG (used by 3 devices)
Media Resource Group Information
Name*
SME-MRG
Description
Devices for this Group
Available Media Resources**
CFB_2
cfb001122334455
cfb00233335da20
Selected Media Resources*
ANN_2 (ANN)
MOH_2 (MOH)
MTP_2 (MTP)
mtp00233335da20 (MTP)
mtp0123456789ab (XCODE)
Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)
Save
Delete
Copy
Add New
*- indicates required item.
**Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)

Cisco Unified Communications Manager-SME Media Resource Group Lists configuration



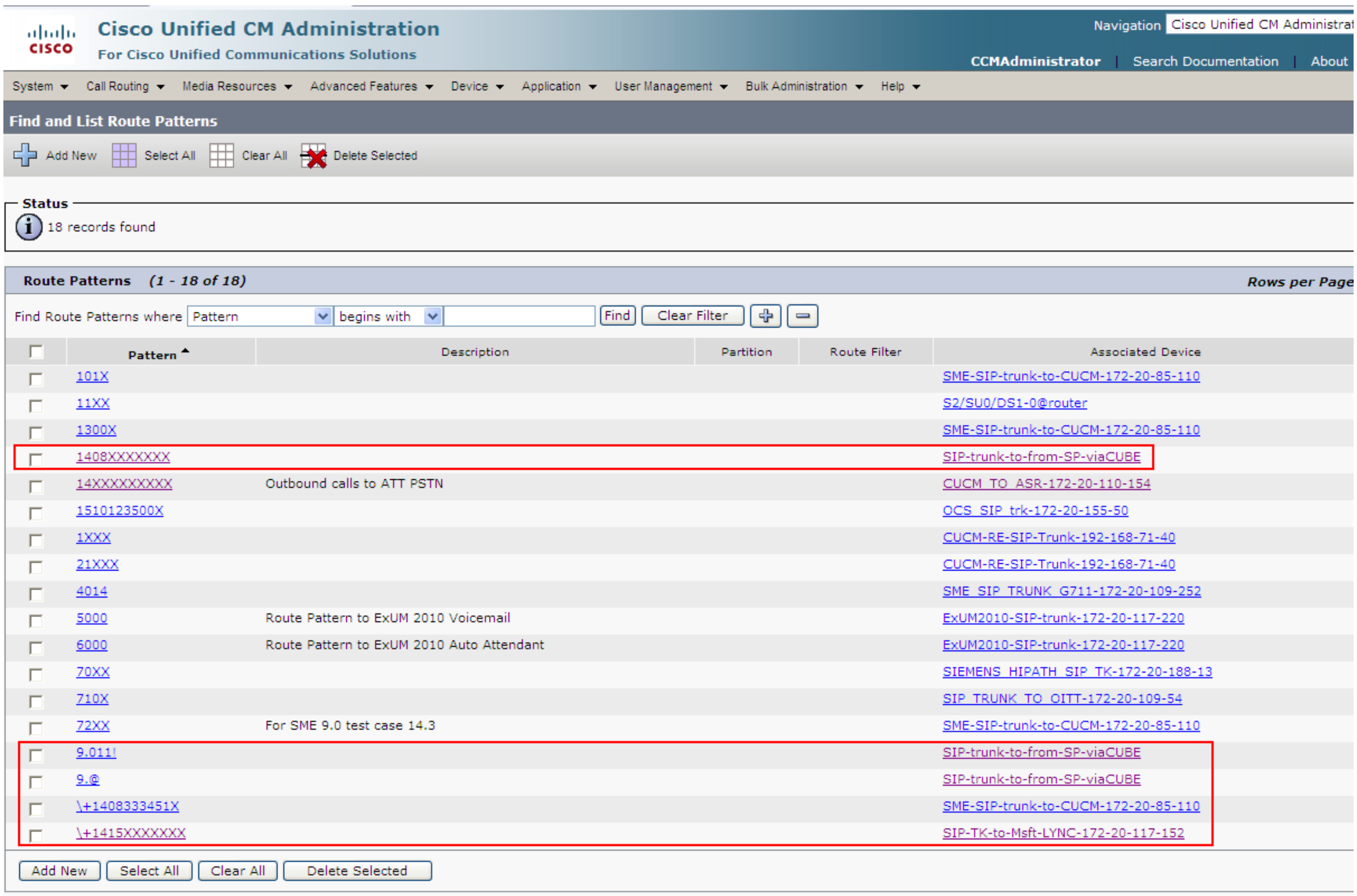






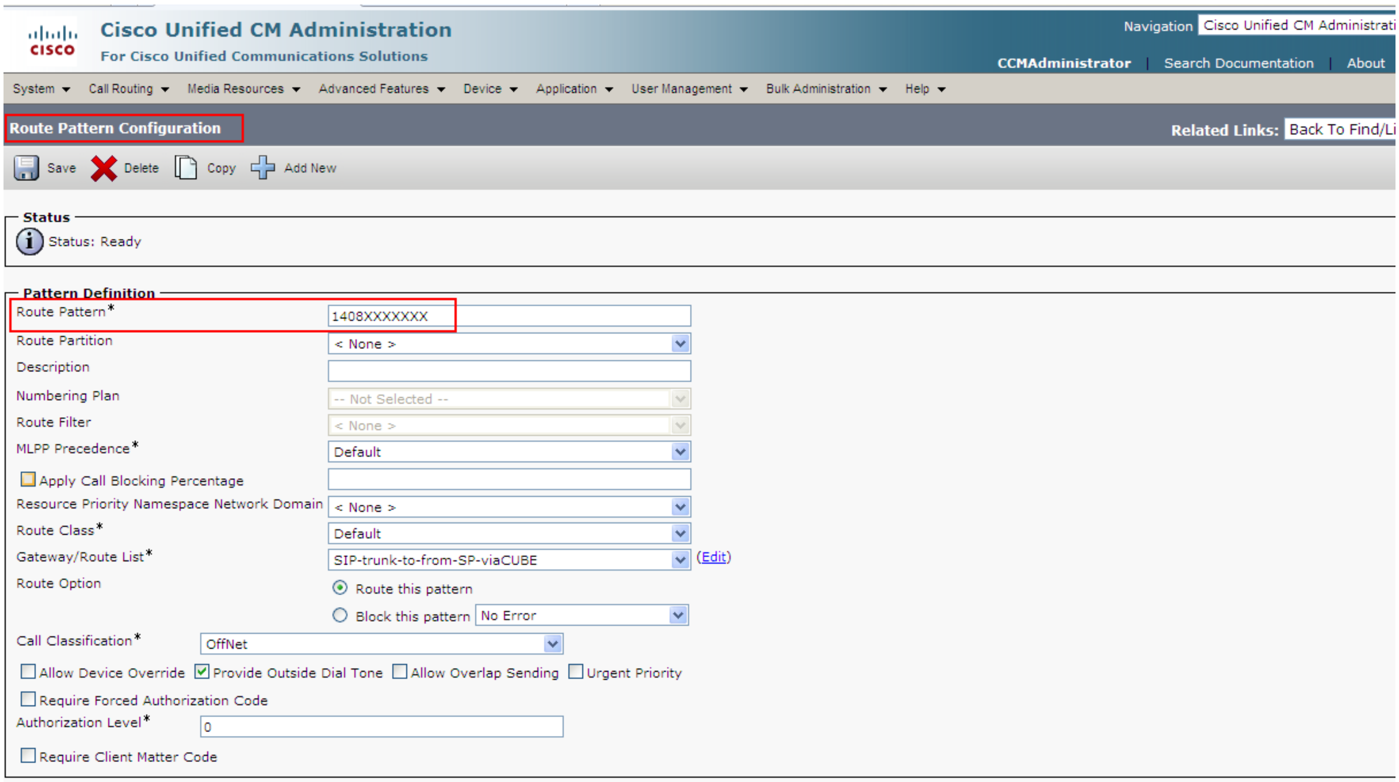
LYNC will need a separate MRGL to force MTP based early offer from the SME.
MRGL to be associated with SIP trunks except LYNC trunk
MRGL to be associated with SIP trunk connecting Microsoft LYNC 2010 server

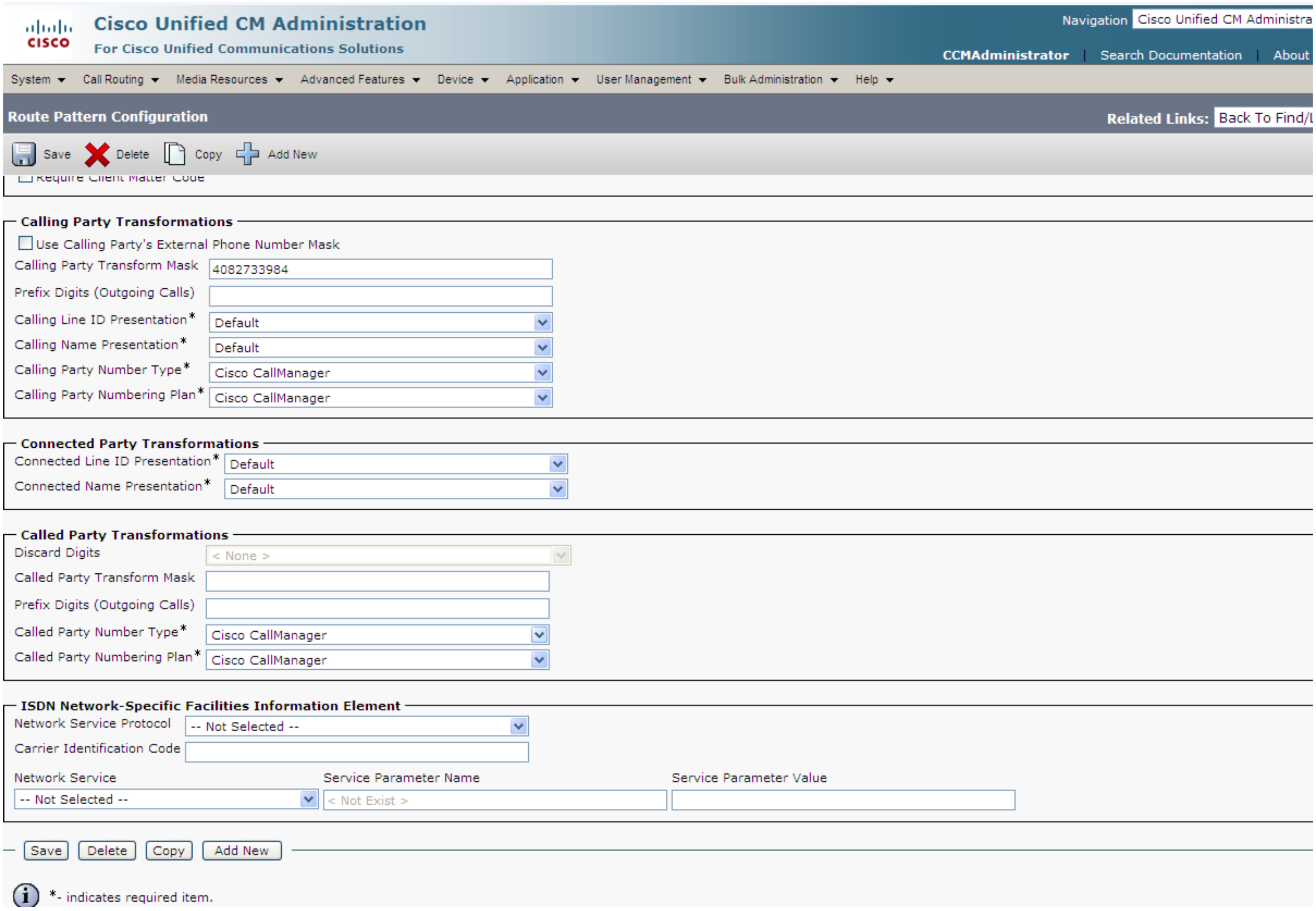
Cisco Unified Communications Manager-SME Route Patterns



Pattern	Description	Partition	Route Filter	Associated Device
101X				SME-SIP-trunk-to-CUCM-172-20-85-110
11XX				S2/SU0/DS1-0@router
1300X				SME-SIP-trunk-to-CUCM-172-20-85-110
1408XXXXXXX				SIP-trunk-to-from-SP-viaCUBE
14XXXXXXXXX	Outbound calls to ATT PSTN			CUCM_TO_ASR-172-20-110-154
1510123500X				OCS_SIP_trk-172-20-155-50
1XXX				CUCM-RE-SIP-Trunk-192-168-71-40
21XXX				CUCM-RE-SIP-Trunk-192-168-71-40
4014				SME_SIP_TRUNK_G711-172-20-109-252
5000	Route Pattern to ExUM 2010 Voicemail			ExUM2010-SIP-trunk-172-20-117-220
6000	Route Pattern to ExUM 2010 Auto Attendant			ExUM2010-SIP-trunk-172-20-117-220
70XX				SIEMENS_HIPATH_SIP_TK-172-20-188-13
710X				SIP_TRUNK_TO_OITT-172-20-109-54
72XX	For SME 9.0 test case 14.3			SME-SIP-trunk-to-CUCM-172-20-85-110
9.011!				SIP-trunk-to-from-SP-viaCUBE
9.@				SIP-trunk-to-from-SP-viaCUBE
\+1408333451X				SME-SIP-trunk-to-CUCM-172-20-85-110
\+1415XXXXXXX				SIP-TK-to-Msft-LYNC-172-20-117-152

Route Pattern to Service Provider

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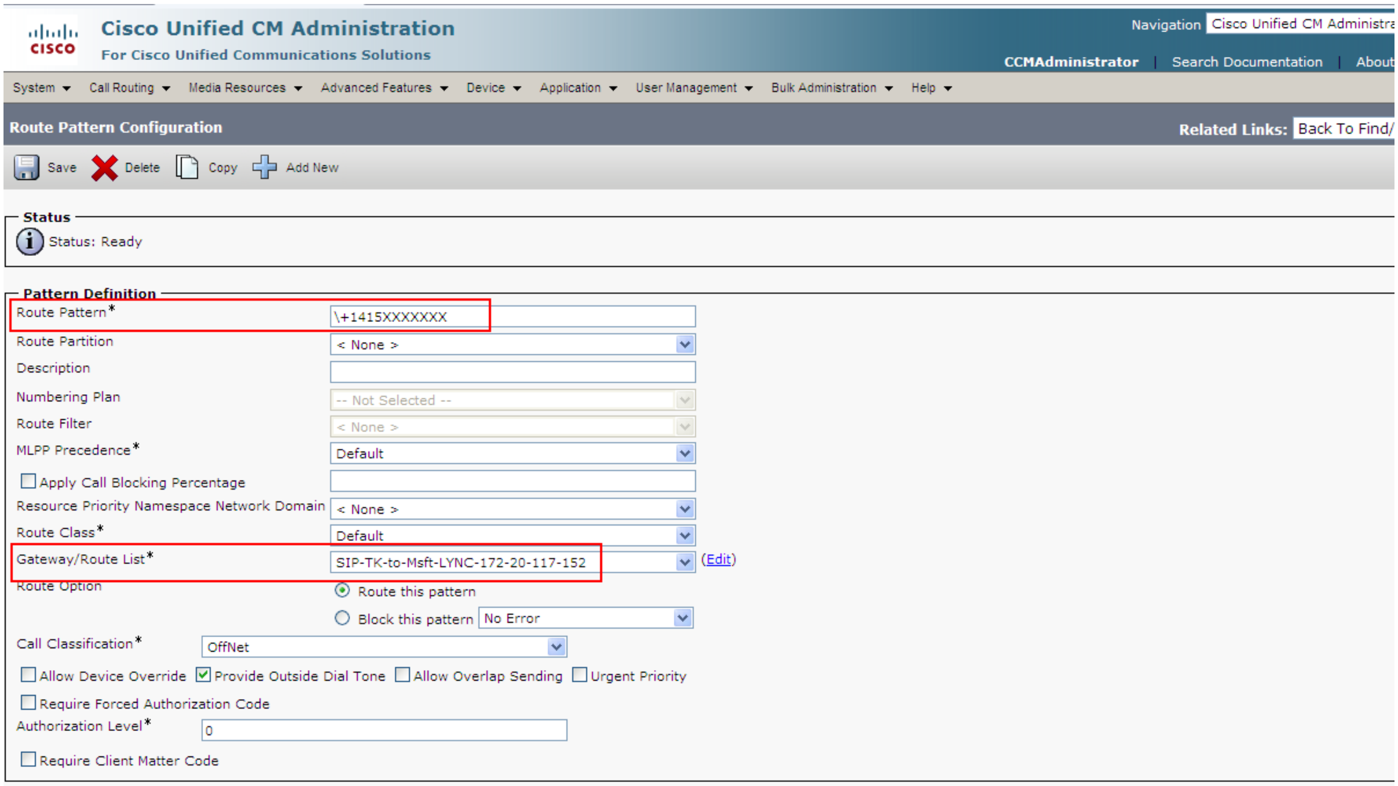
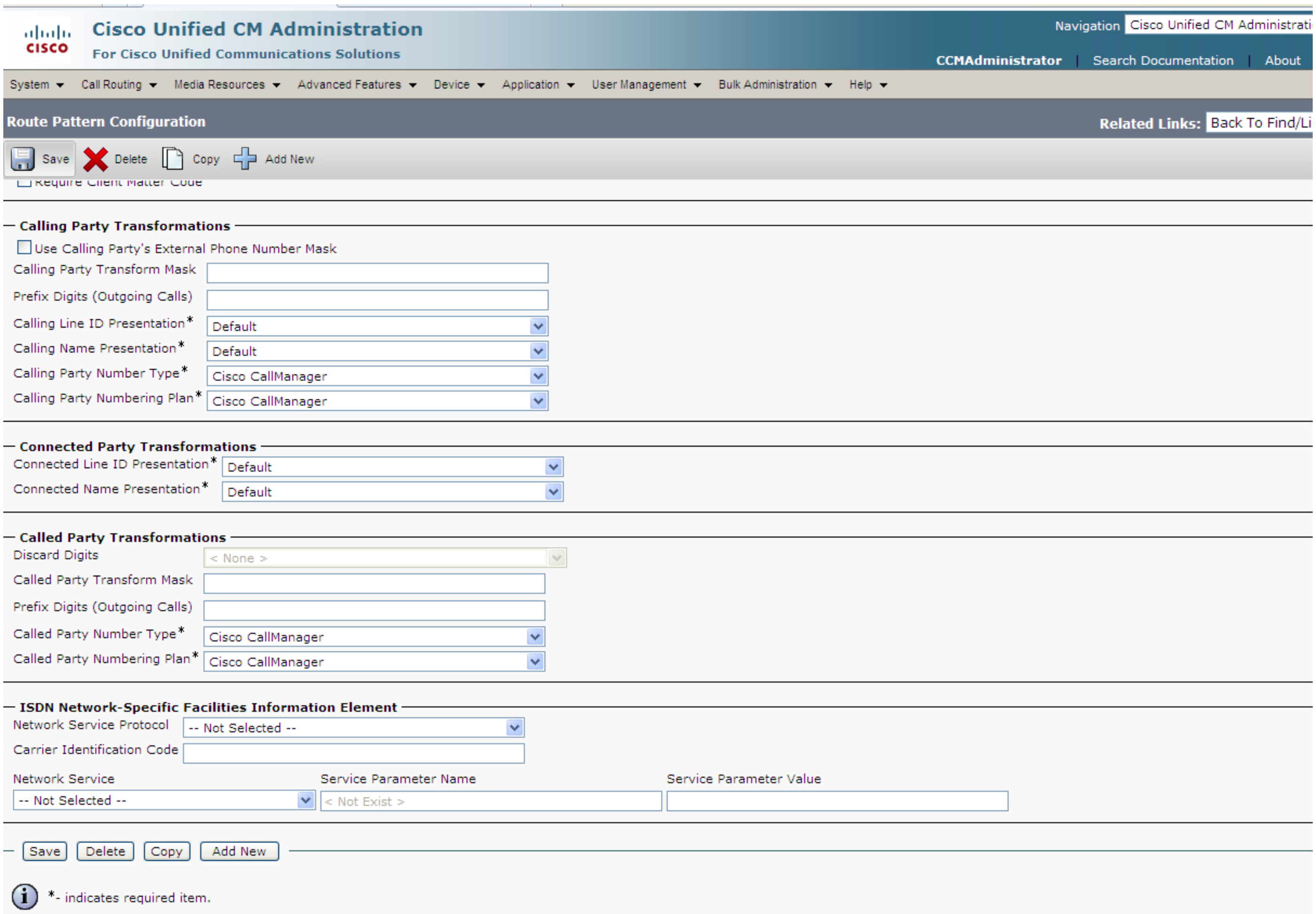
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask 4082733984
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling Party Number Type* Cisco CallManager
Calling Party Numbering Plan* Cisco 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.

Route Pattern to the Microsoft LYNC 2010 server

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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

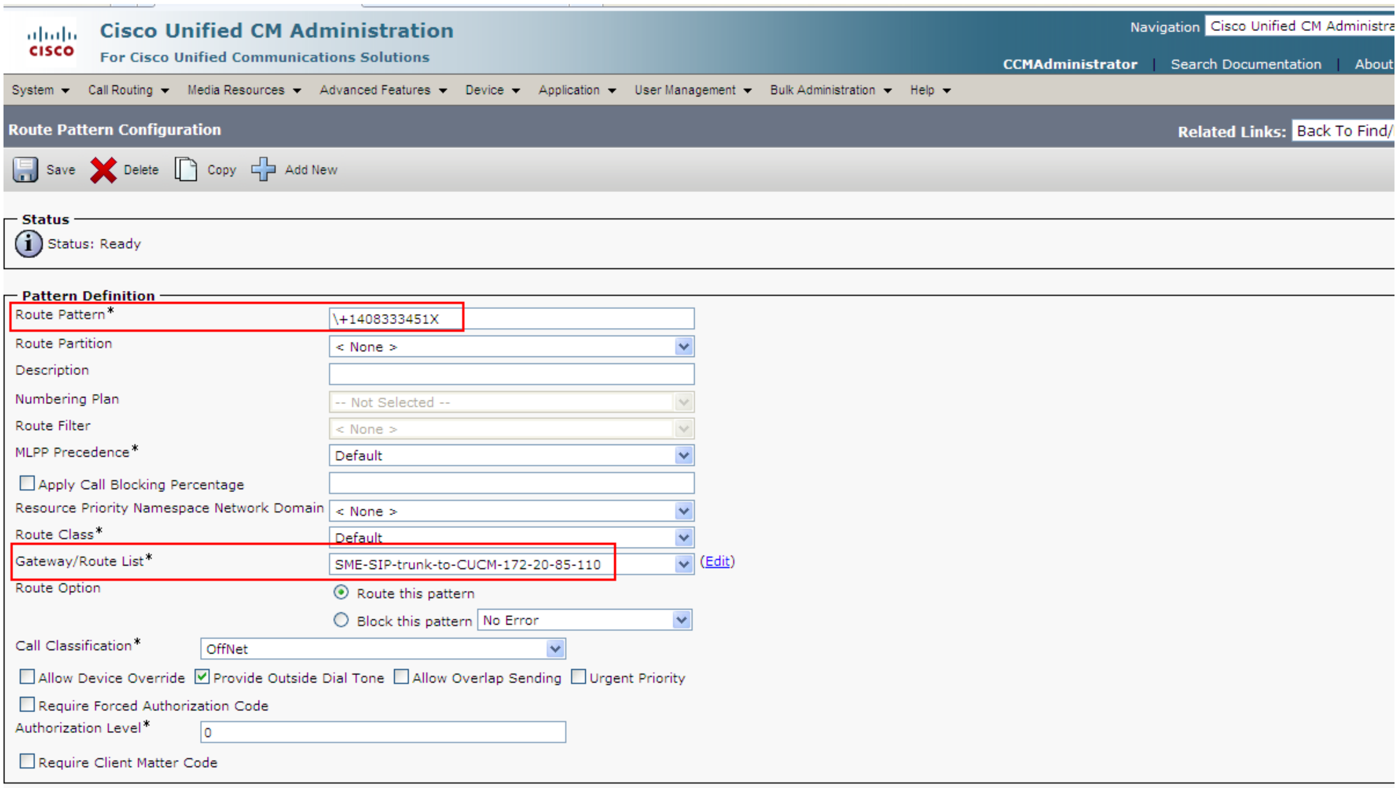
Called Party Transformations
Discard Digits < None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type* Cisco CallManager
Called Party Numbering Plan* Cisco CallManager

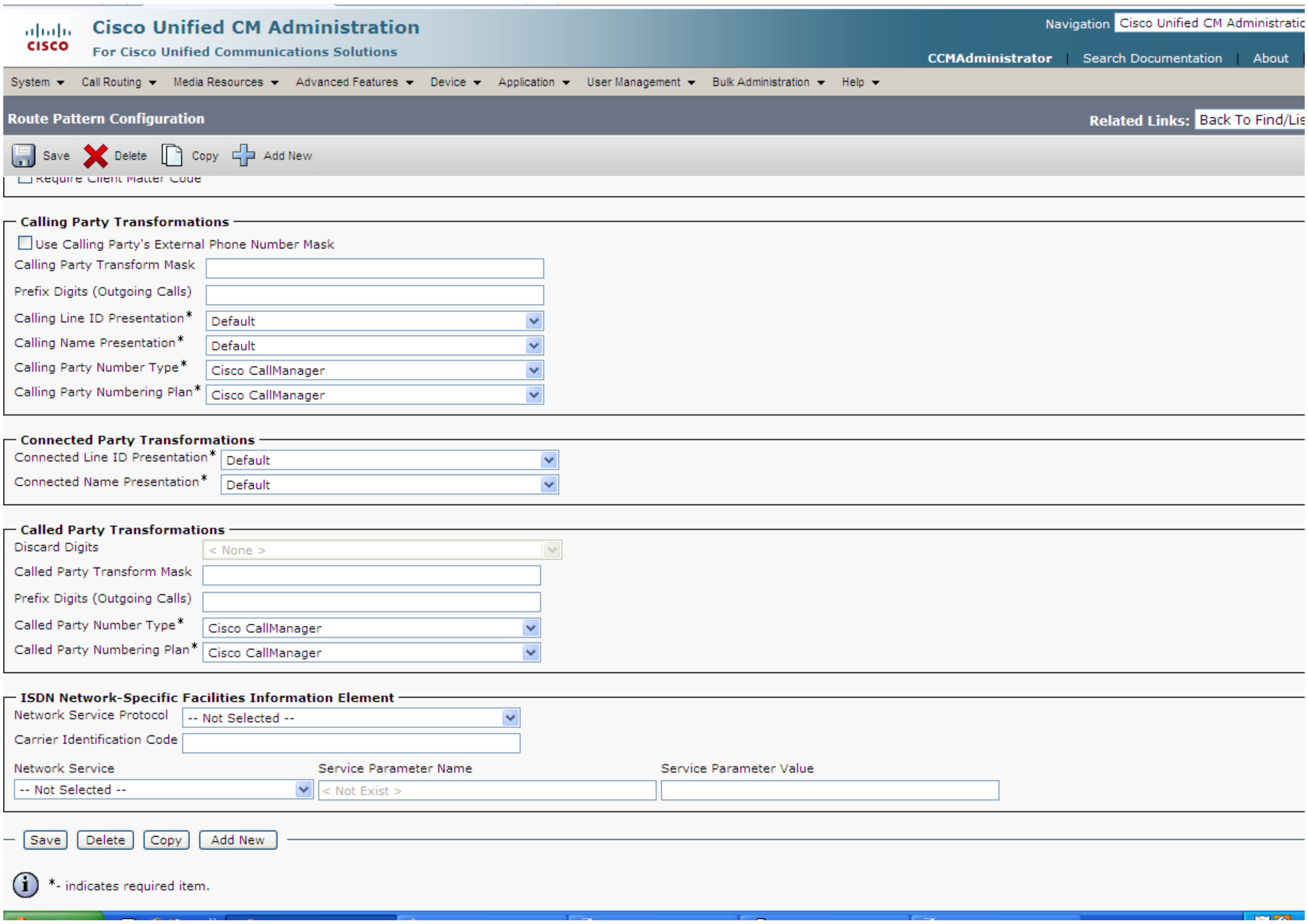
ISDN Network-Specific Facilities Information Element
Network Service Protocol -- Not Selected --
Carrier Identification Code
Network Service -- Not Selected --
Service Parameter Name < Not Exist >
Service Parameter Value

Save Delete Copy Add New

*- indicates required item.

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

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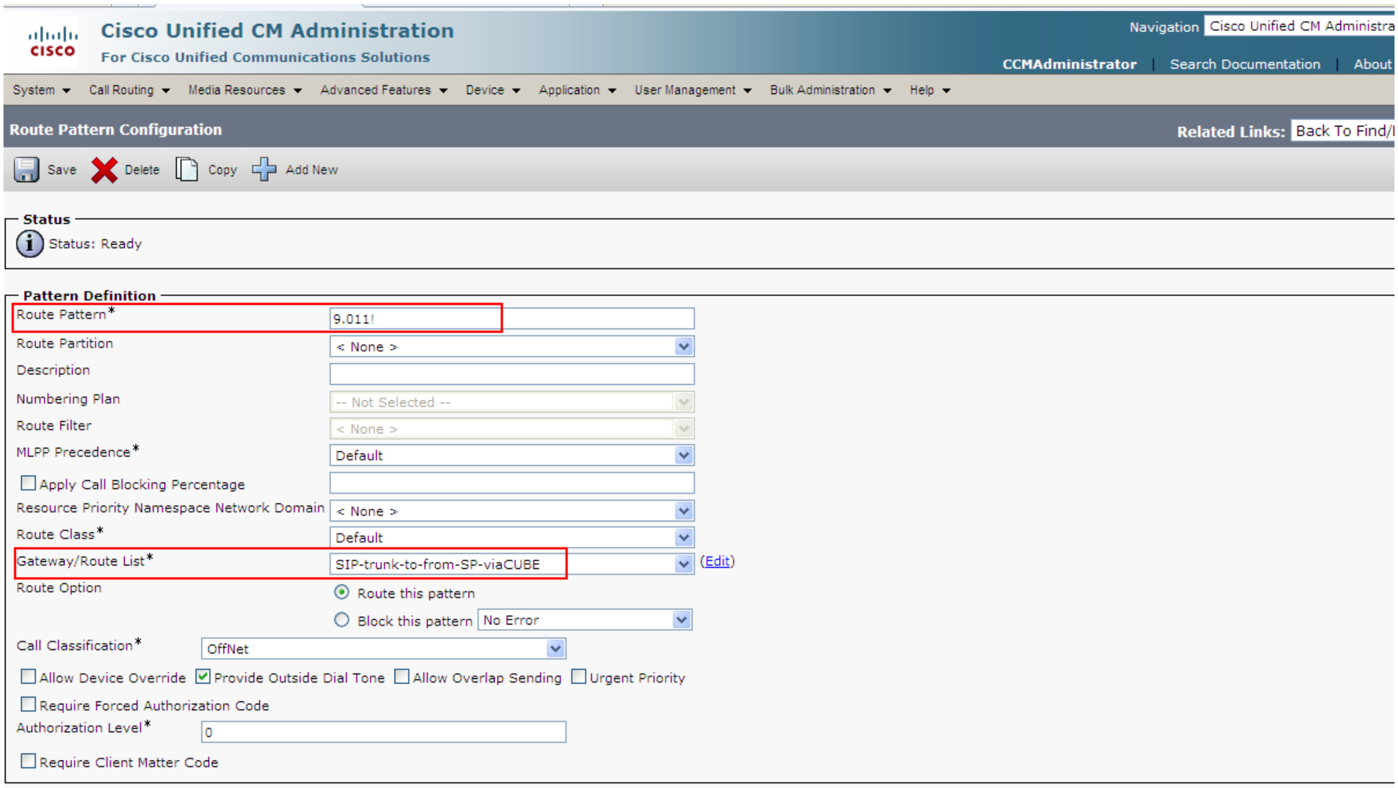
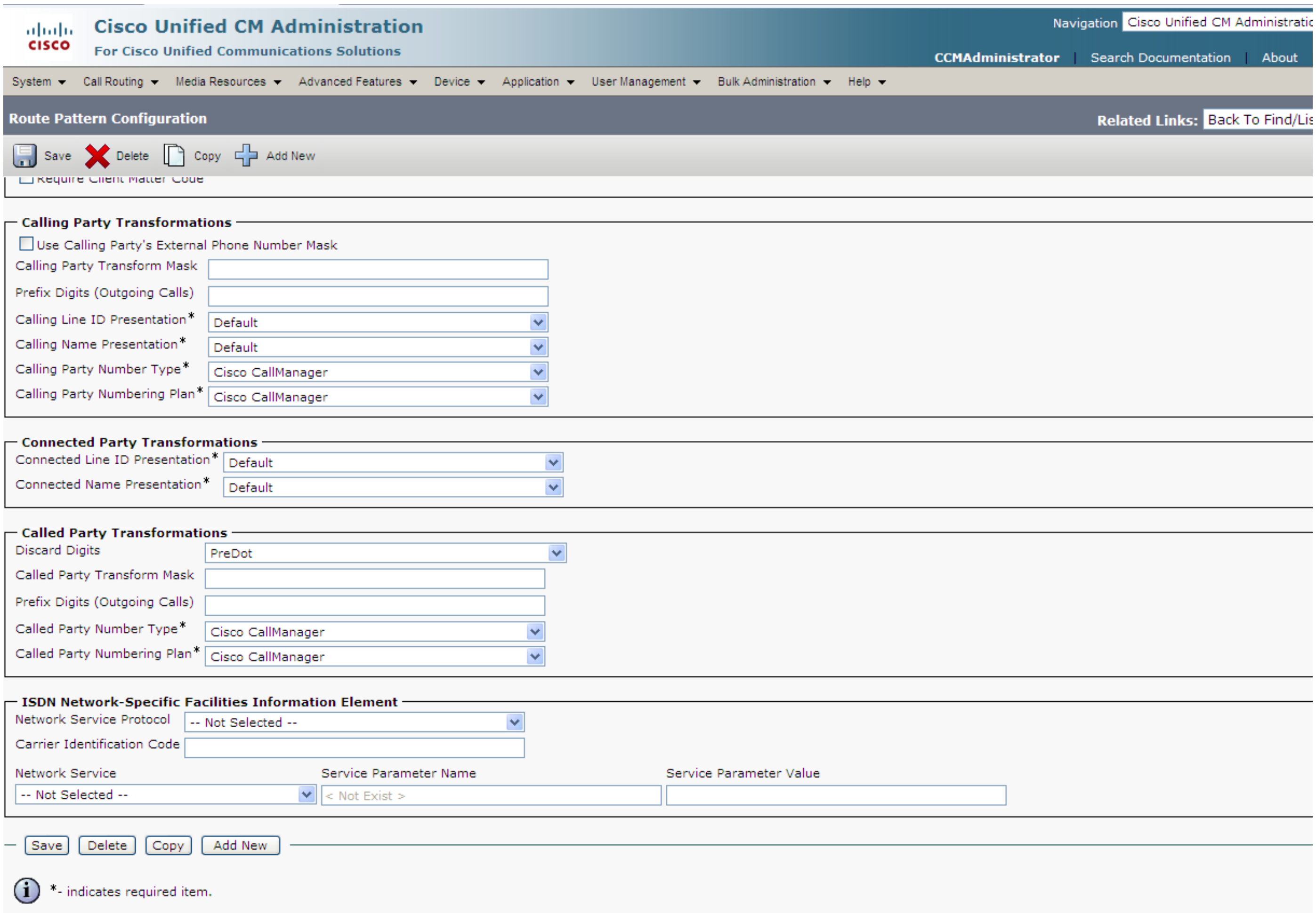
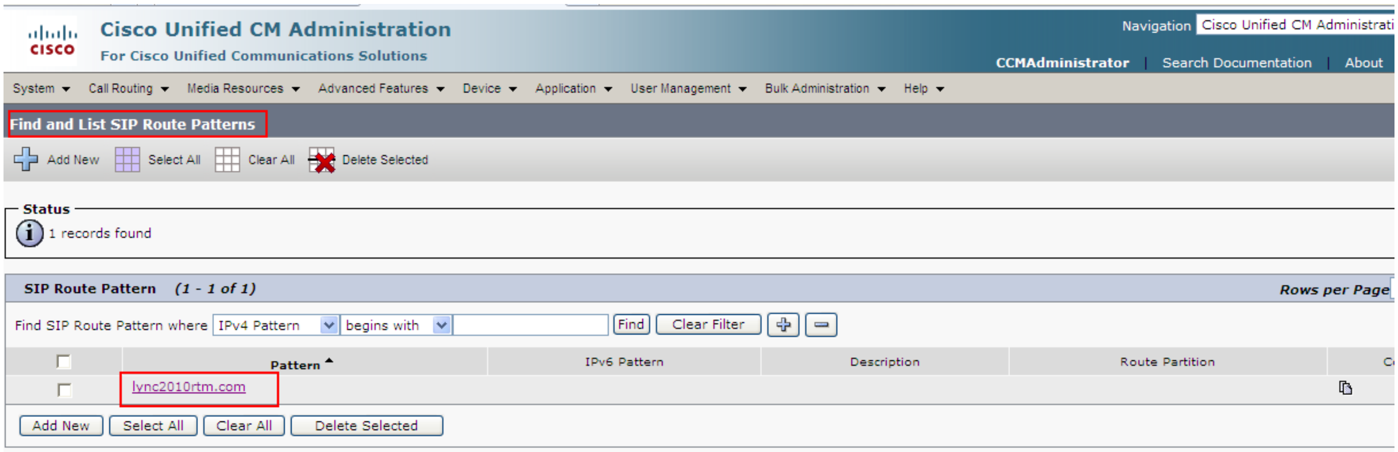
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling Party Number Type* Cisco CallManager
Calling Party Numbering Plan* Cisco CallManager
Connected Party Transformations
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.

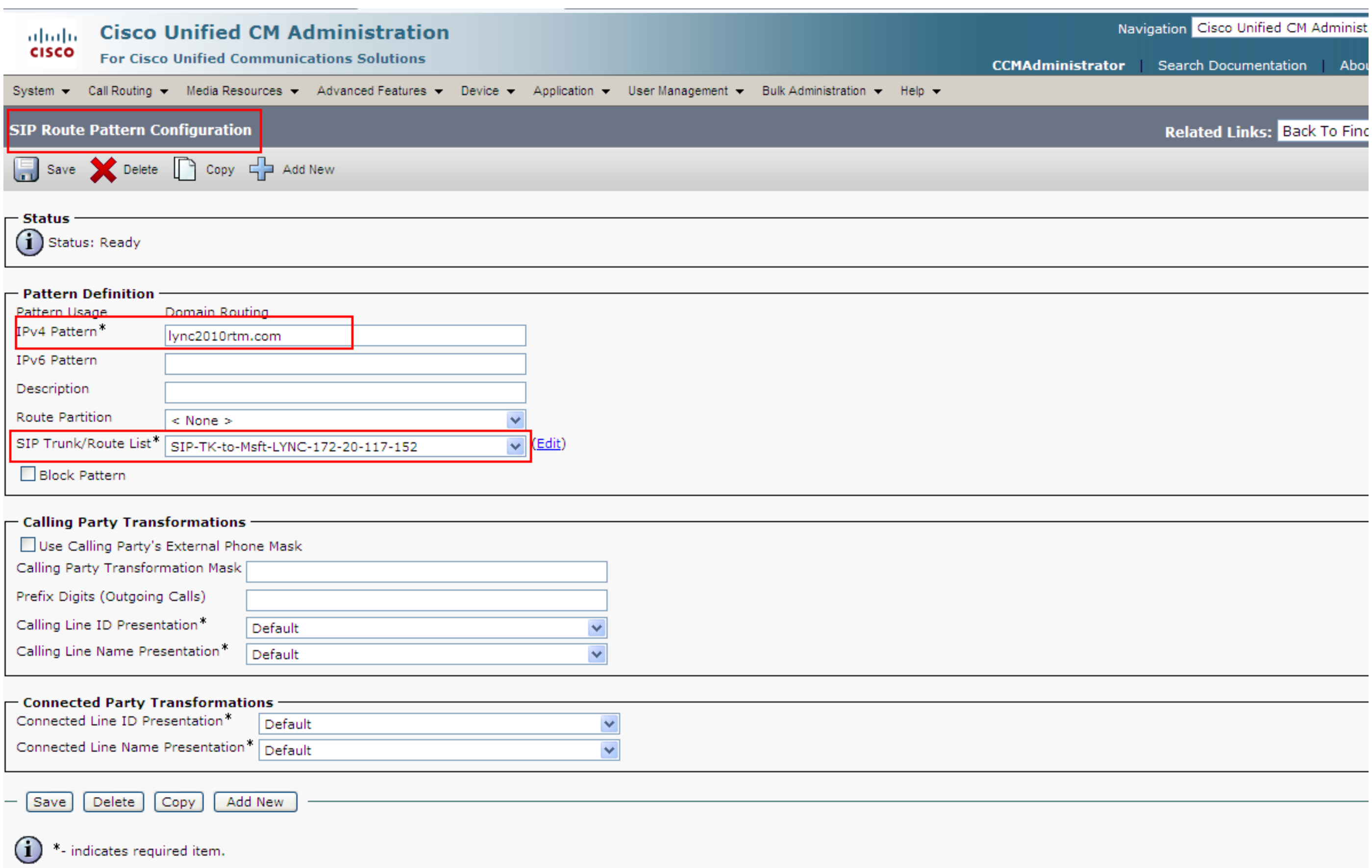
International Route Pattern




Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administratio
CCMAdministrator | Search Documentation | About
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Route Pattern Configuration
Related Links: Back To Find/Lis
Save Delete Copy Add New
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.

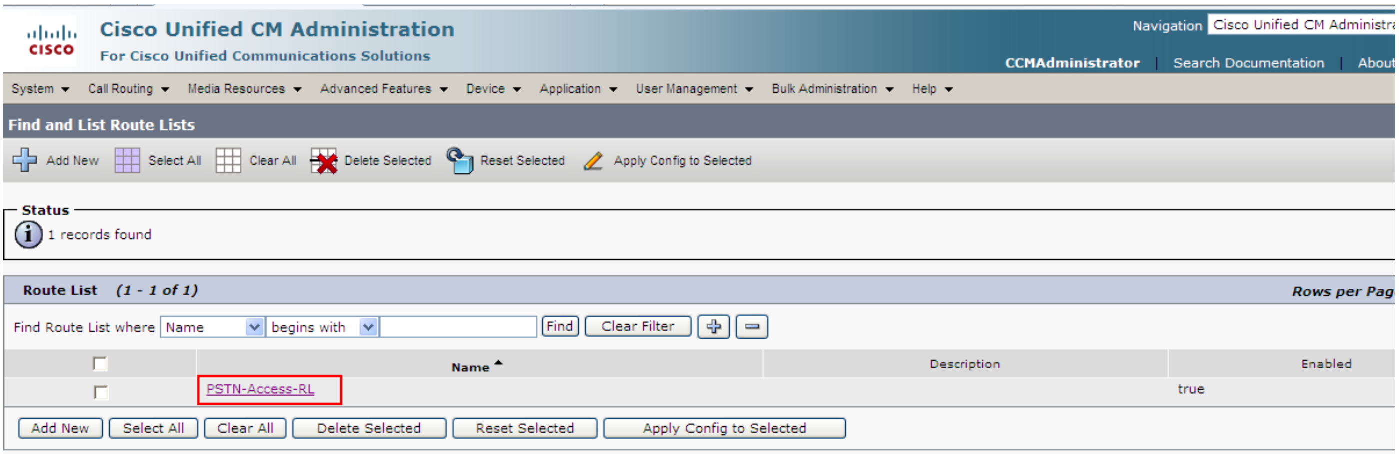
SIP Route Pattern *

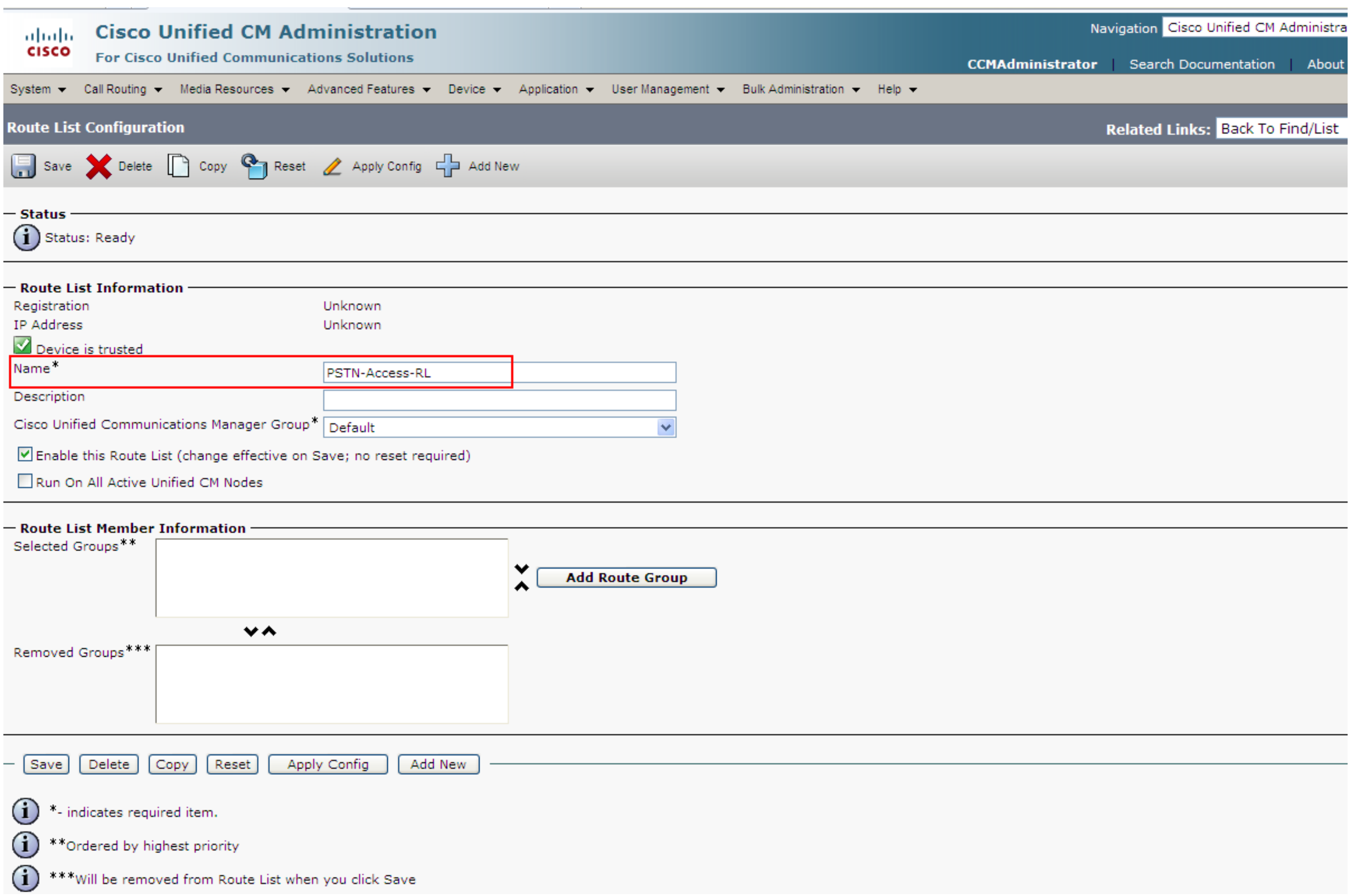




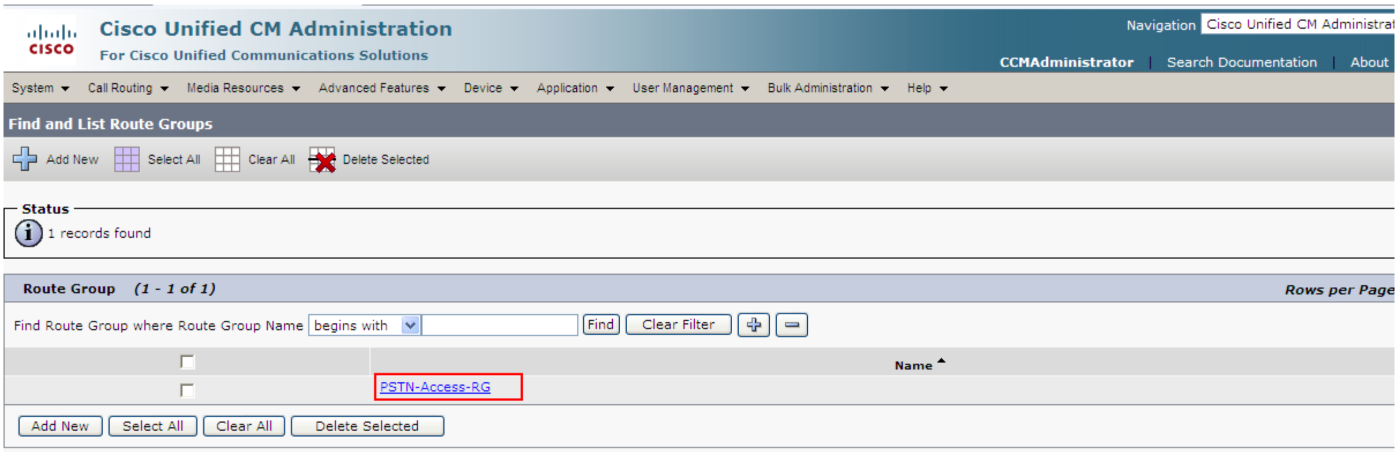
*This SIP route pattern is needed for the calls originating from CUCM and destined to the Lync via CUCM-SME using URI dialing and not the normal E.164 number dialing

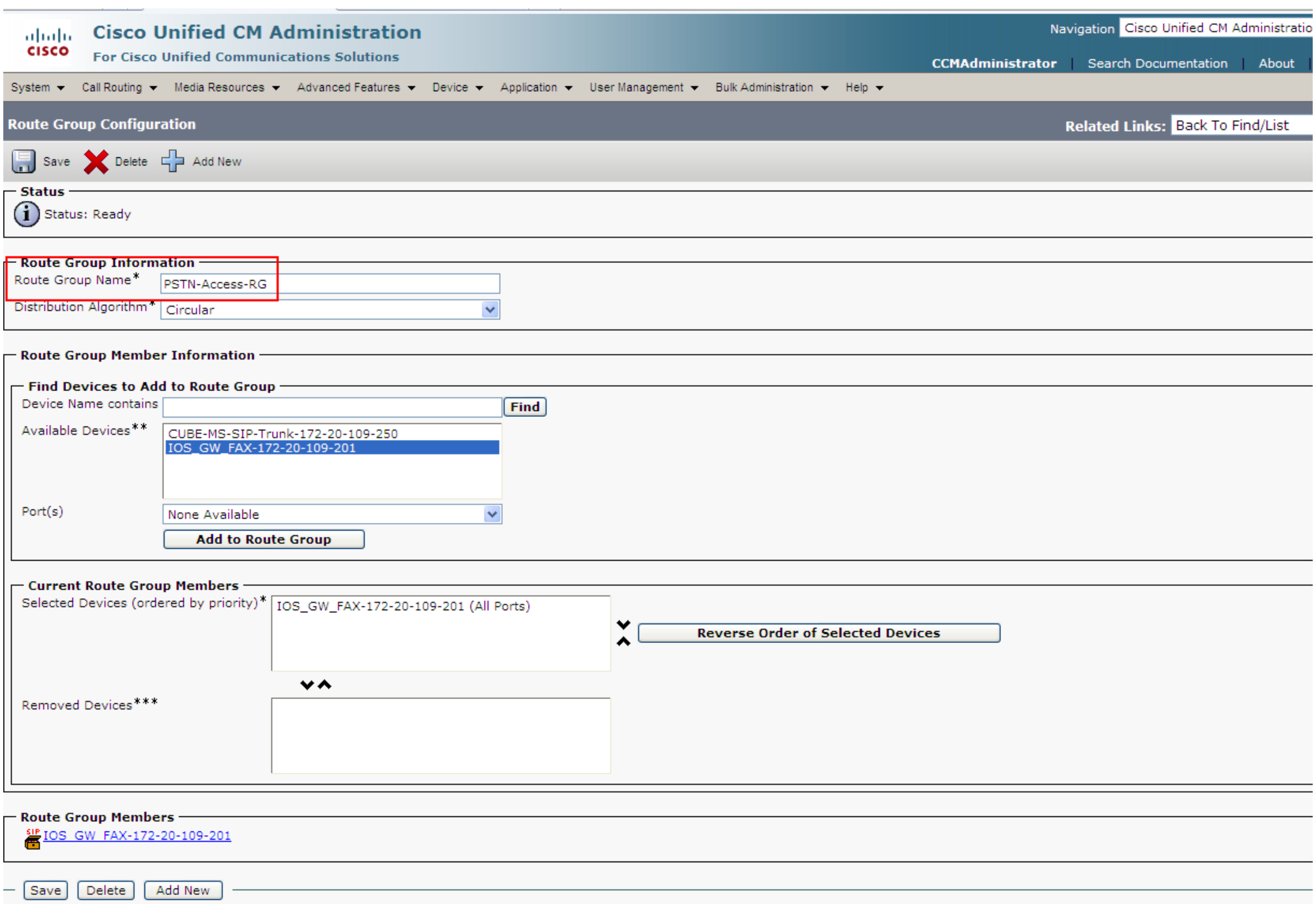
Cisco Unified Communications Manager-SME Route list for PSTN (CUBE) access

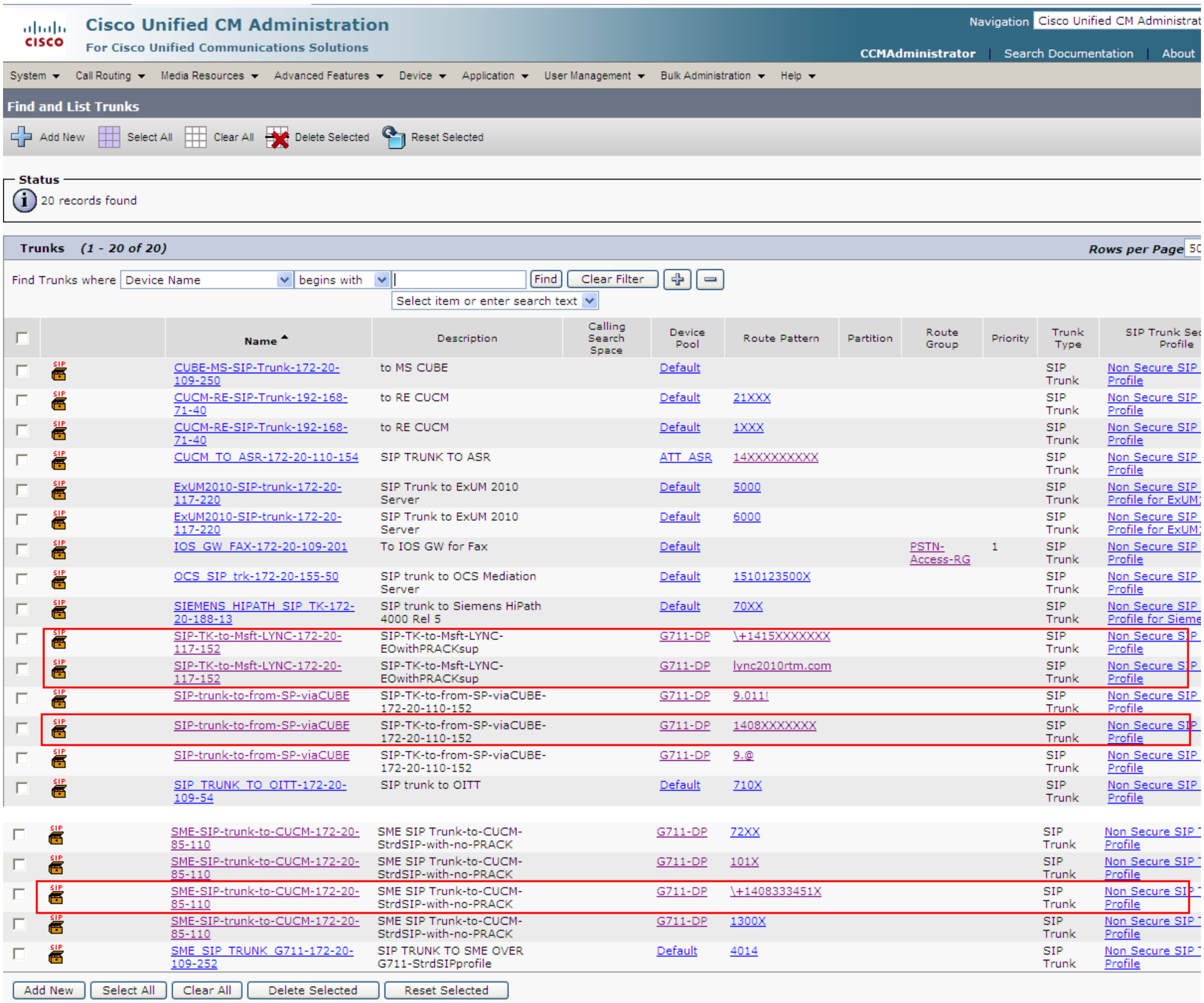




Cisco Unified Communications Manager-SME Route Group for PSTN Trunk Access

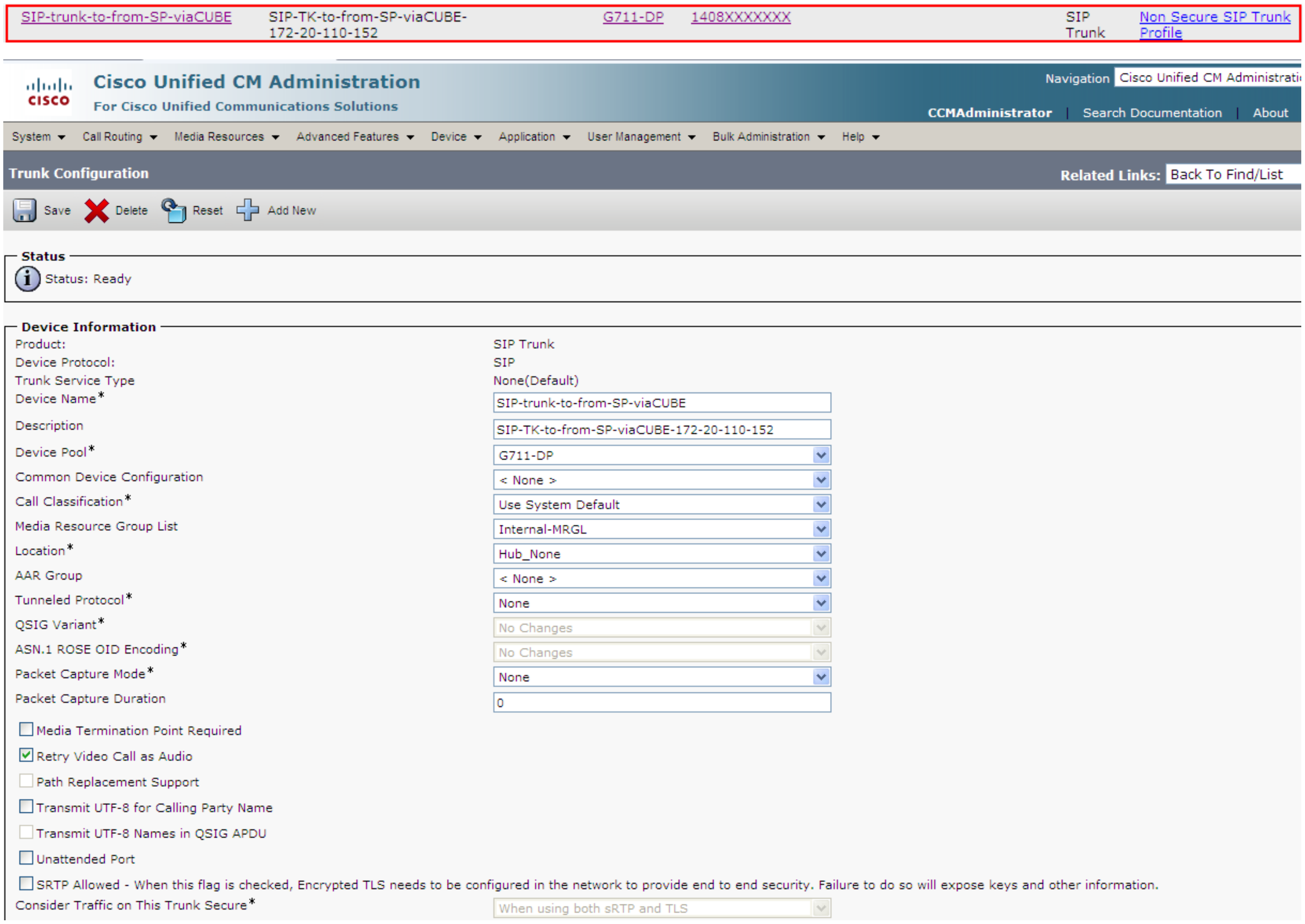






Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Sec Profile
CUBE-MS-SIP-Trunk-172-20-109-250	to MS CUBE		Default					SIP Trunk	Non Secure SIP Profile
CUCM-RE-SIP-Trunk-192-168-71-40	to RE CUCM		Default	21XXX				SIP Trunk	Non Secure SIP Profile
CUCM-RE-SIP-Trunk-192-168-71-40	to RE CUCM		Default	1XXX				SIP Trunk	Non Secure SIP Profile
CUCM_TO_ASR-172-20-110-154	SIP TRUNK TO ASR		ATT_ASR	14XXXXXXXXX				SIP Trunk	Non Secure SIP Profile
ExUM2010-SIP-trunk-172-20-117-220	SIP Trunk to ExUM 2010 Server		Default	5000				SIP Trunk	Non Secure SIP Profile for ExUM
ExUM2010-SIP-trunk-172-20-117-220	SIP Trunk to ExUM 2010 Server		Default	6000				SIP Trunk	Non Secure SIP Profile for ExUM
IOS_GW_FAX-172-20-109-201	To IOS GW for Fax		Default			PSTN-Access-RG	1	SIP Trunk	Non Secure SIP Profile
OCS_SIP_trk-172-20-155-50	SIP trunk to OCS Mediation Server		Default	1510123500X				SIP Trunk	Non Secure SIP Profile
SIEMENS_HIPATH_SIP_TK-172-20-188-13	SIP trunk to Siemens HiPath 4000 Rel 5		Default	70XX				SIP Trunk	Non Secure SIP Profile for Sieme
SIP-TK-to-Msft-LYNC-172-20-117-152	SIP-TK-to-Msft-LYNC-EOwithPRACKsup		G711-DP	\+1415XXXXXXX				SIP Trunk	Non Secure SIP Profile
SIP-TK-to-Msft-LYNC-172-20-117-152	SIP-TK-to-Msft-LYNC-EOwithPRACKsup		G711-DP	lync2010rtm.com				SIP Trunk	Non Secure SIP Profile
SIP-trunk-to-from-SP-viaCUBE	SIP-TK-to-from-SP-viaCUBE-172-20-110-152		G711-DP	9.011!				SIP Trunk	Non Secure SIP Profile
SIP-trunk-to-from-SP-viaCUBE	SIP-TK-to-from-SP-viaCUBE-172-20-110-152		G711-DP	1408XXXXXXX				SIP Trunk	Non Secure SIP Profile
SIP-trunk-to-from-SP-viaCUBE	SIP-TK-to-from-SP-viaCUBE-172-20-110-152		G711-DP	9.@				SIP Trunk	Non Secure SIP Profile
SIP_TRUNK_TO_OITT-172-20-109-54	SIP trunk to OITT		Default	710X				SIP Trunk	Non Secure SIP Profile
SME-SIP-trunk-to-CUCM-172-20-85-110	SME SIP Trunk-to-CUCM-StrdSIP-with-no-PRACK		G711-DP	72XX				SIP Trunk	Non Secure SIP Profile
SME-SIP-trunk-to-CUCM-172-20-85-110	SME SIP Trunk-to-CUCM-StrdSIP-with-no-PRACK		G711-DP	101X				SIP Trunk	Non Secure SIP Profile
SME-SIP-trunk-to-CUCM-172-20-85-110	SME SIP Trunk-to-CUCM-StrdSIP-with-no-PRACK		G711-DP	\+1408333451X				SIP Trunk	Non Secure SIP Profile
SME-SIP-trunk-to-CUCM-172-20-85-110	SME SIP Trunk-to-CUCM-StrdSIP-with-no-PRACK		G711-DP	1300X				SIP Trunk	Non Secure SIP Profile
SME_SIP_TRUNK_G711-172-20-109-252	SIP TRUNK TO SME OVER G711-StrdSIPprofile		Default	4014				SIP Trunk	Non Secure SIP Profile

Cisco Unified Communications Manager-SME SIP trunk to Service Provider via ASR CUBE



Route Class Signaling Enabled* | Default
Use Trusted Relay Point* | Default

☑ PSTN Access

☐ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile | < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain | < None >

Call Routing Information

☐ Remote-Party-Id

☑ Asserted-Identity

Asserted-Type* | PAI
SIP Privacy* | Default

Inbound Calls

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

☑ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings | Default Prefix Settings

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

Connected Party Settings

Connected Party Transformation CSS | < None >

☑ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS | < None >

☑ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS | < None >

☑ Use Device Pool Calling Party Transformation CSS

Calling Party Selection* | Originator
Calling Line ID Presentation* | Default
Calling Name Presentation* | Default
Calling and Connected Party Info Format* | Deliver DN only in connected party

☑ Redirecting Diversion Header Delivery - Outbound

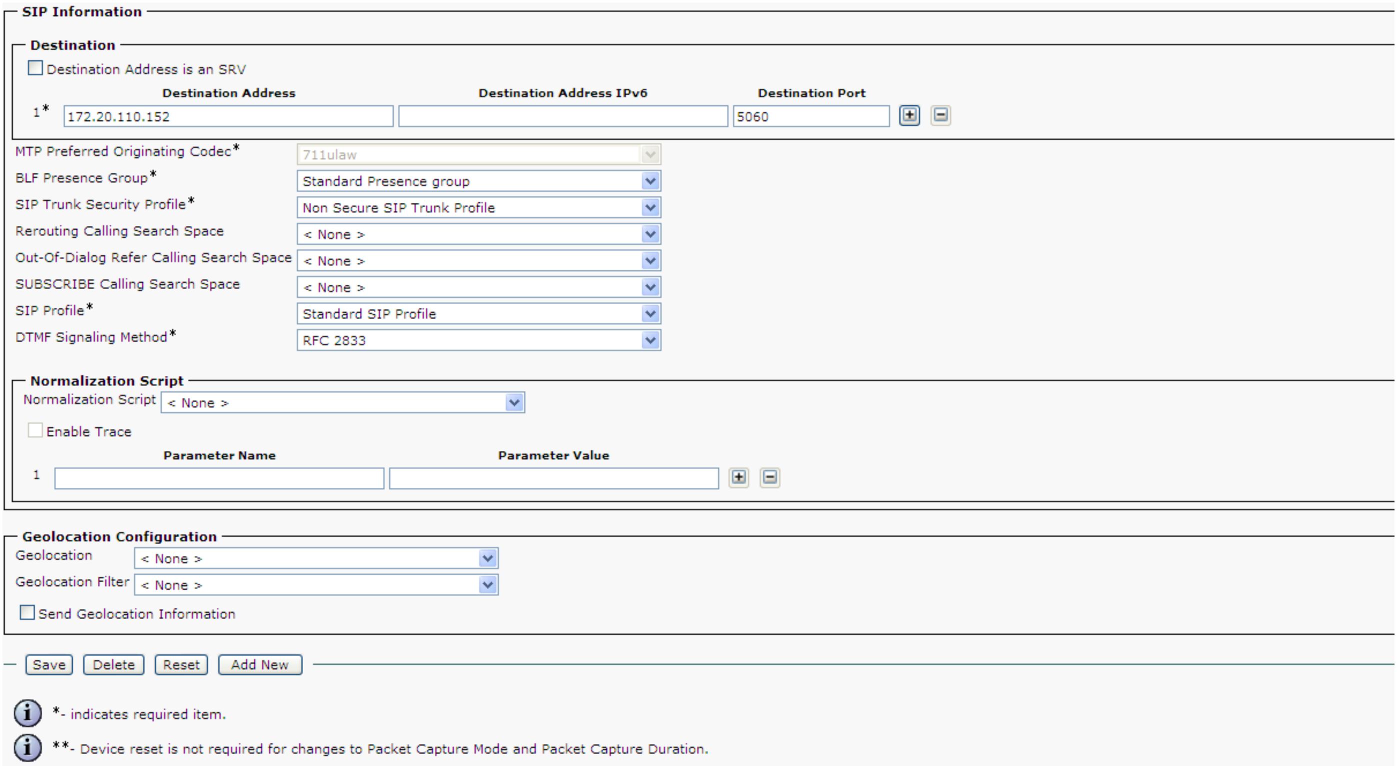
Redirecting Party Transformation CSS | < None >

☑ Use Device Pool Redirecting Party Transformation CSS

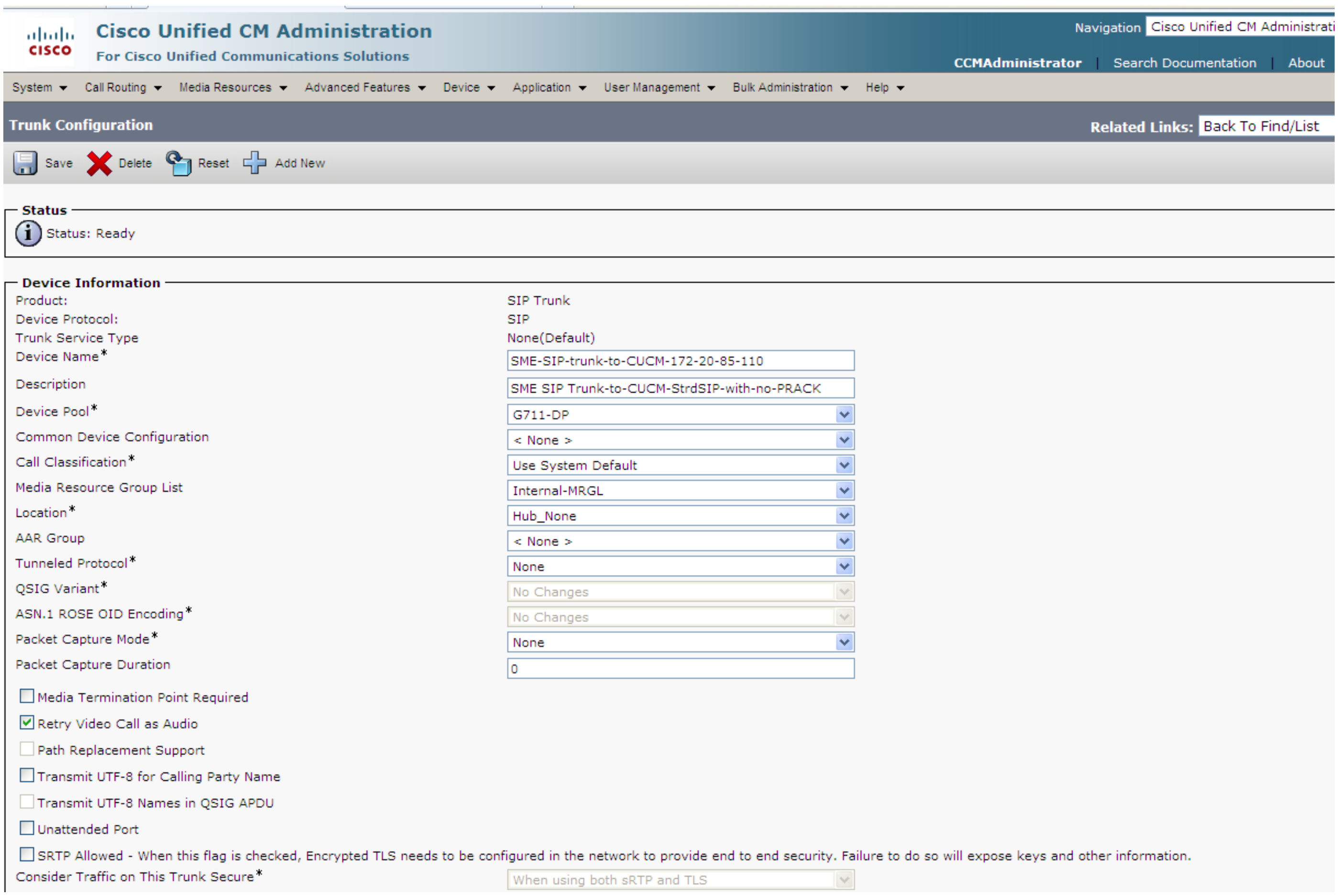
Caller Information

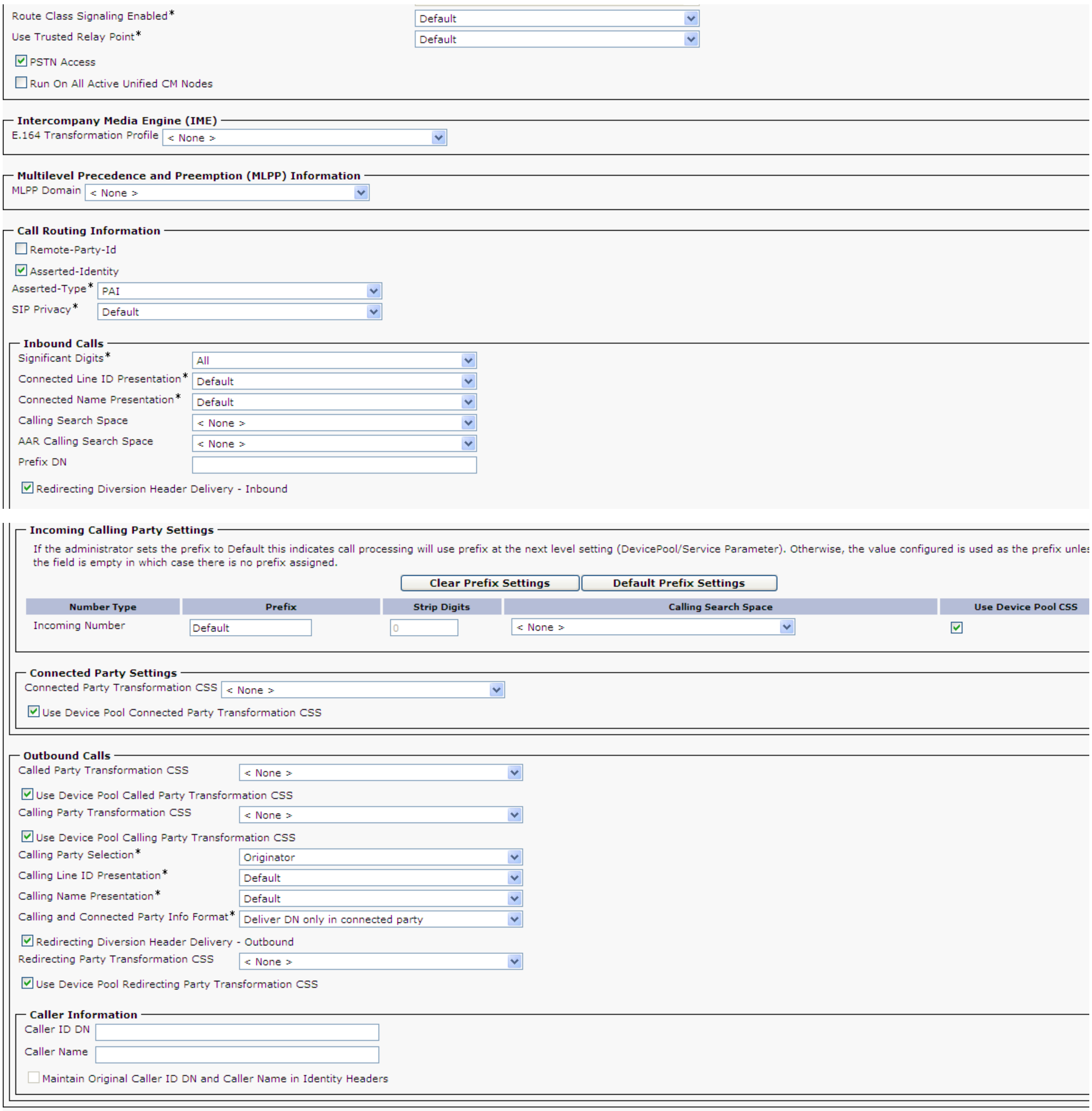
Caller ID DN |
Caller Name |

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


SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1*
172.20.110.152
5060
MTP Preferred Originating Codec*
711ulaw
BLF Presence Group*
Standard Presence group
SIP Trunk Security Profile*
Non Secure SIP Trunk Profile
Rerouting Calling Search Space
< None >
Out-Of-Dialog Refer Calling Search Space
< None >
SUBSCRIBE Calling Search Space
< None >
SIP Profile*
Standard SIP Profile
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
Save
Delete
Reset
Add New
*- indicates required item.
**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Cisco Unified Communications Manager-SME SIP trunk to Cisco Unified Communications Manager





Route Class Signaling Enabled* Default
Use Trusted Relay Point* Default
☑ PSTN Access
☐ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)
E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain < None >

Call Routing Information
☐ Remote-Party-Id
☑ Asserted-Identity
Asserted-Type* PAI
SIP Privacy* Default

Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN
☑ Redirecting Diversion Header Delivery - Inbound

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

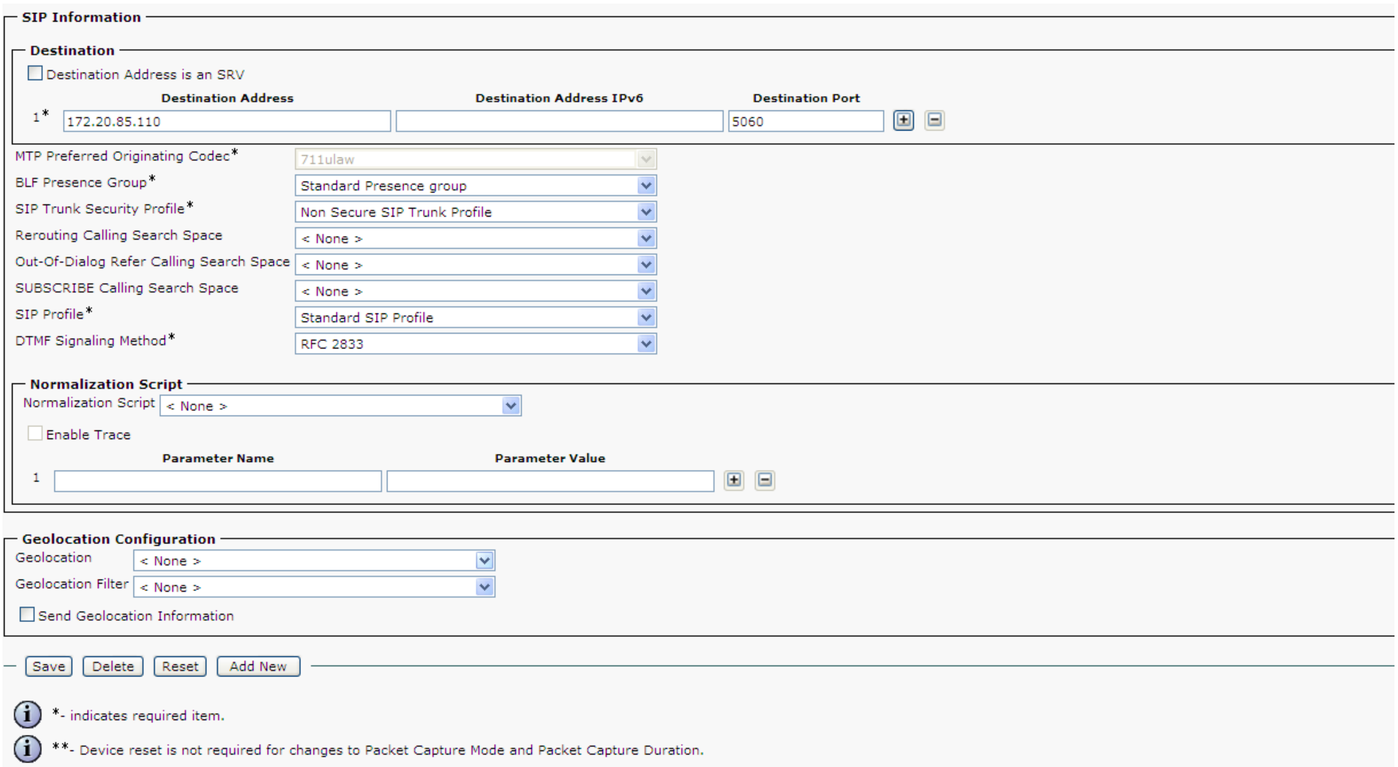
Clear Prefix Settings | Default Prefix Settings

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

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

Outbound Calls
Called Party Transformation CSS < None >
☑ Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS < None >
☑ Use Device Pool Calling Party Transformation CSS
Calling Party Selection* Originator
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling and Connected Party Info Format* Deliver DN only in connected party
☑ Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS < None >
☑ Use Device Pool Redirecting Party Transformation CSS

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



SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	172.20.85.110		5060

MTP Preferred Originating Codec*	711ulaw
BLF Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	RFC 2833

Normalization Script

Normalization Script < None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Geolocation Configuration

Geolocation	< None >
Geolocation Filter	< None >

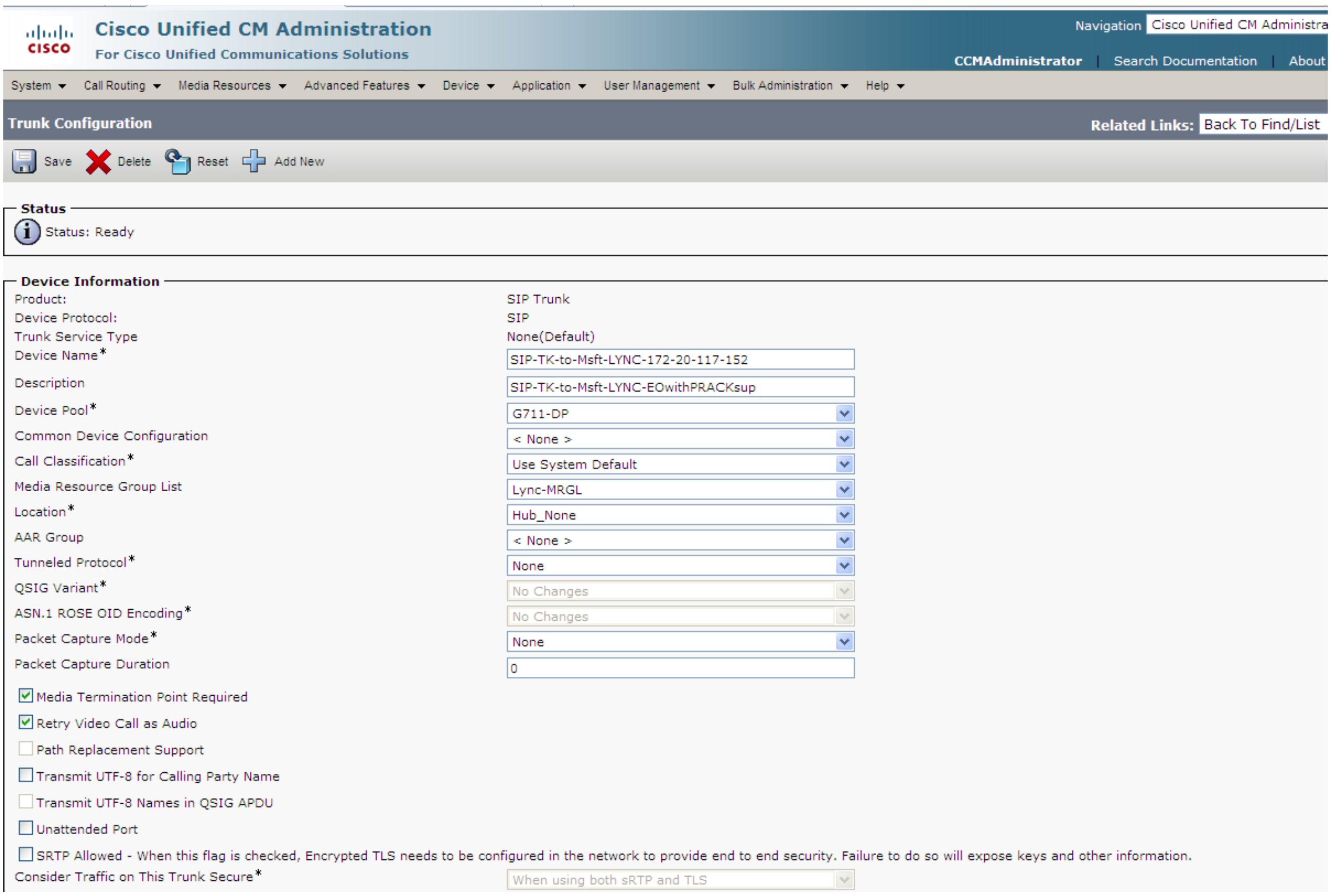
☐ Send Geolocation Information

Save Delete Reset Add New

*- indicates required item.

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


SIP-TK-to-Msft-LYNC-172-20-117-152
SIP-TK-to-Msft-LYNC-EOwithPRACKsup
G711-DP
\+1415XXXXXXX
SIP Trunk
Non Secure S
Profile


Cisco Unified CM Administration
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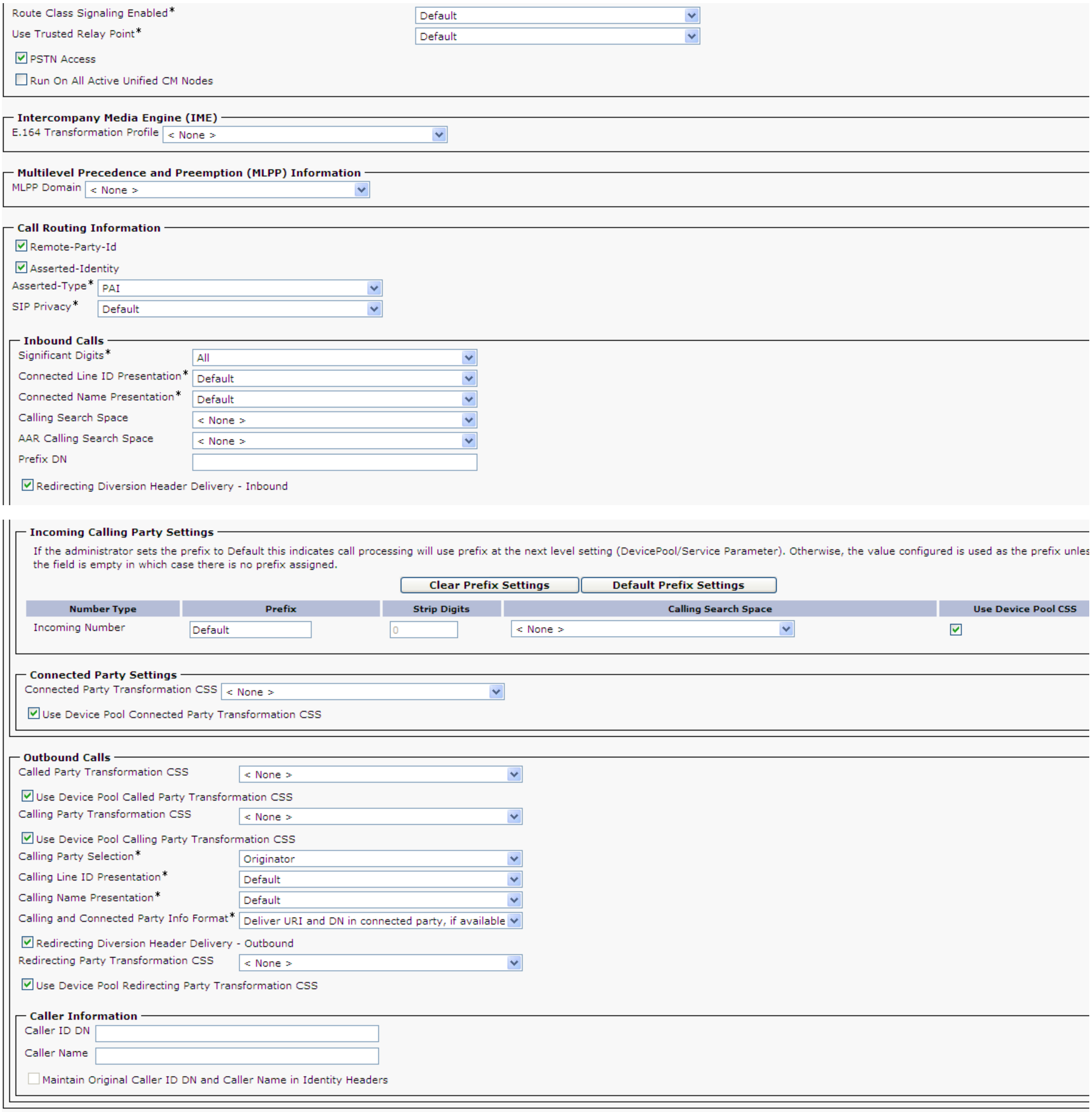
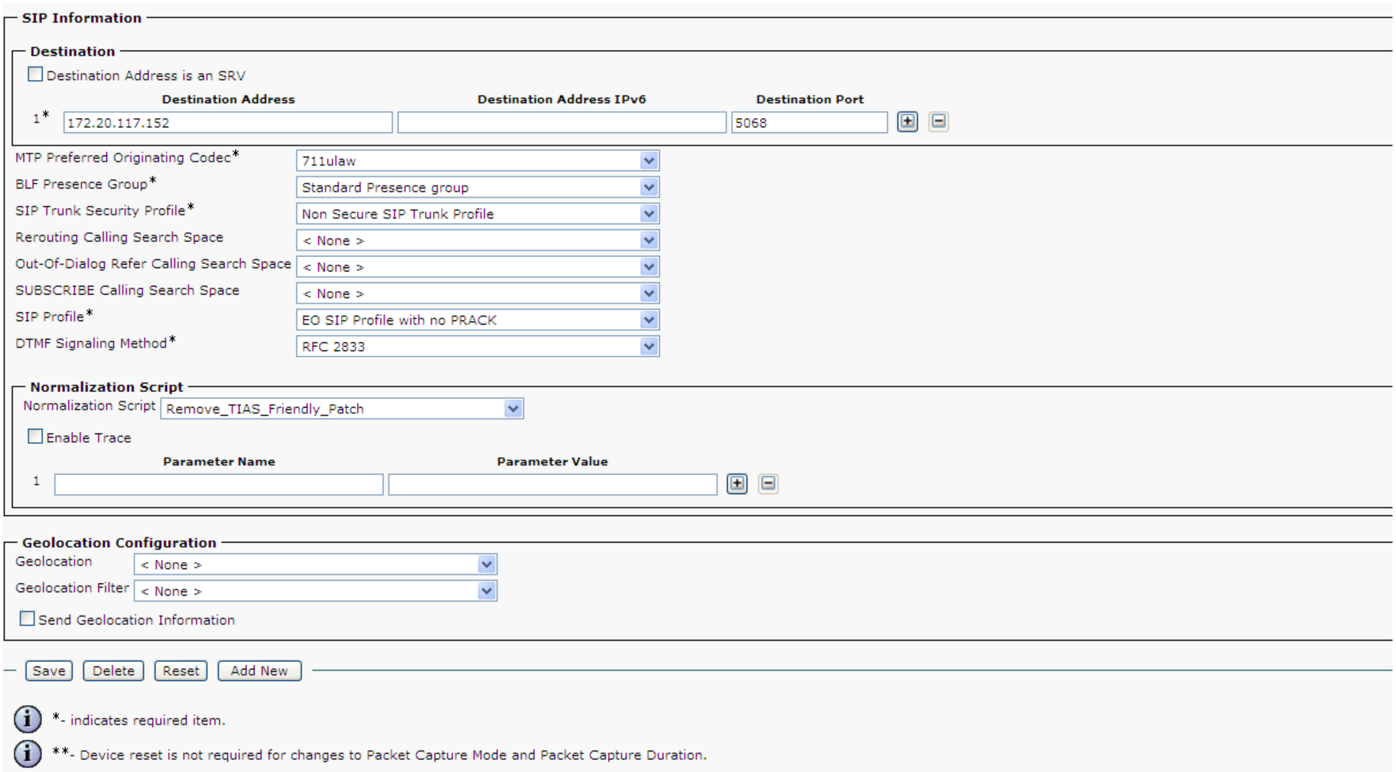
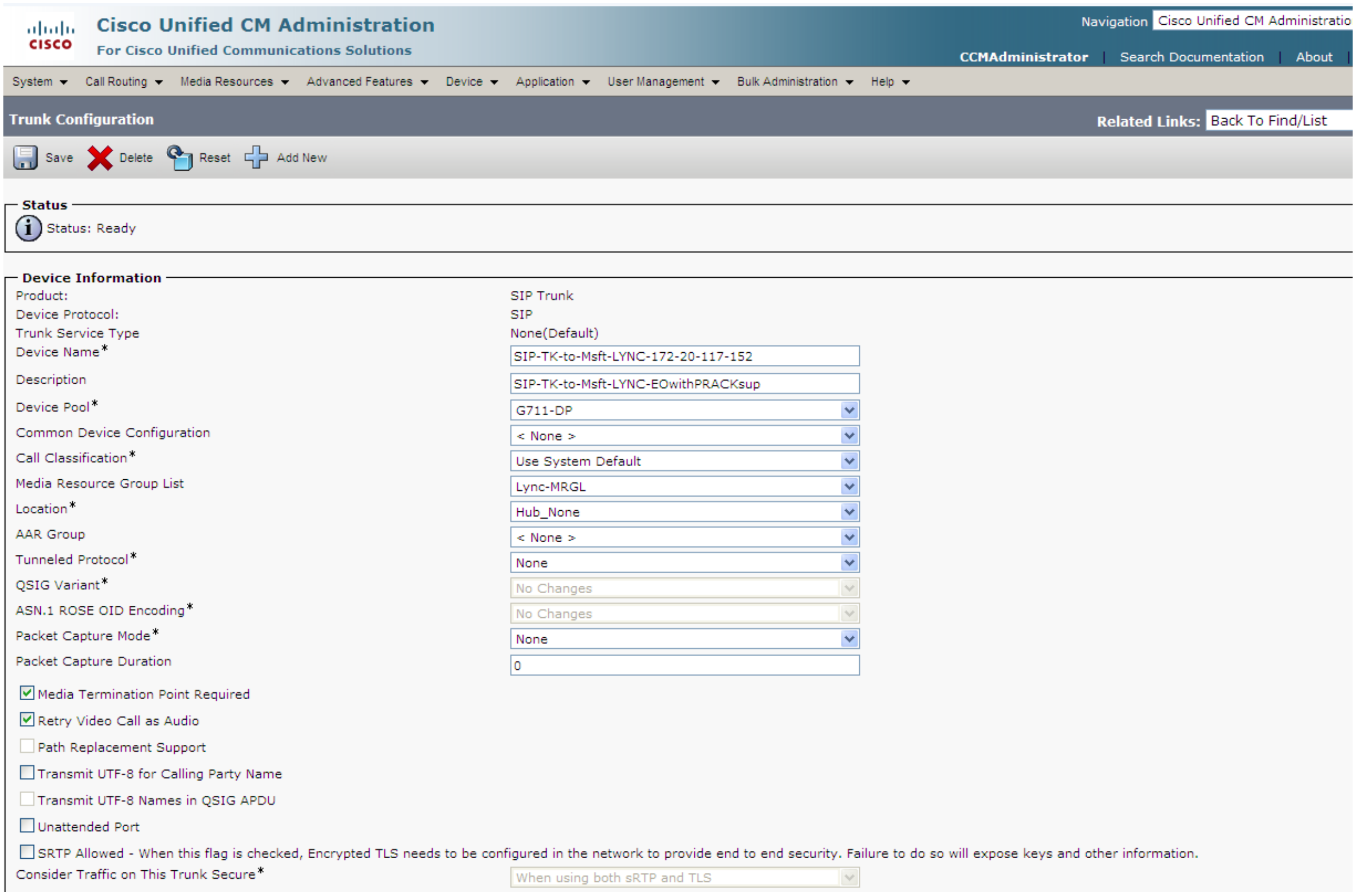
Navigation Cisco Unified CM Administra
CCMAdministrator | Search Documentation | About
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Trunk Configuration
Related Links: Back To Find/List
Save Delete Reset Add New
Status
Status: Ready
Device Information
Product: SIP Trunk
Device Protocol: SIP
Trunk Service Type None(Default)
Device Name* SIP-TK-to-Msft-LYNC-172-20-117-152
Description SIP-TK-to-Msft-LYNC-EOwithPRACKsup
Device Pool* G711-DP
Common Device Configuration < None >
Call Classification* Use System Default
Media Resource Group List Lync-MRGL
Location* Hub_None
AAR Group < None >
Tunneled Protocol* None
QSIG Variant* No Changes
ASN.1 ROSE OID Encoding* No Changes
Packet Capture Mode* None
Packet Capture Duration 0
Media Termination Point Required
Retry Video Call as Audio
Path Replacement Support
Transmit UTF-8 for Calling Party Name
Transmit UTF-8 Names in QSIG APDU
Unattended Port
SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
Consider Traffic on This Trunk Secure* When using both sRTP and TLS


Route Class Signaling Enabled* Default
Use Trusted Relay Point* Default
PSTN Access
Run On All Active Unified CM Nodes
Intercompany Media Engine (IME)
E.164 Transformation Profile < None >
Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain < None >
Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type* PAI
SIP Privacy* Default
Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN
Redirecting Diversion Header Delivery - Inbound
Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unles the field is empty in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS
Incoming Number | Default | 0 | < None >
Connected Party Settings
Connected Party Transformation CSS < None >
Use Device Pool Connected Party Transformation CSS
Outbound Calls
Called Party Transformation CSS < None >
Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS < None >
Use Device Pool Calling Party Transformation CSS
Calling Party Selection* Originator
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling and Connected Party Info Format* Deliver URI and DN in connected party, if available
Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS < None >
Use Device Pool Redirecting Party Transformation CSS
Caller Information
Caller ID DN
Caller Name
Maintain Original Caller ID DN and Caller Name in Identity Headers


SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1*
172.20.117.152
5068
MTP Preferred Originating Codec*
711ulaw
BLF Presence Group*
Standard Presence group
SIP Trunk Security Profile*
Non Secure SIP Trunk Profile
Rerouting Calling Search Space
< None >
Out-Of-Dialog Refer Calling Search Space
< None >
SUBSCRIBE Calling Search Space
< None >
SIP Profile*
EO SIP Profile with no PRACK
DTMF Signaling Method*
RFC 2833
Normalization Script
Normalization Script
Remove_TIAS_Friendly_Patch
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.

Cisco Unified Communications Manager-SME SIP trunk to Microsoft Lync 2010 Server Standard Edition →This SIP trunk will be used when the call is made from the endpoint behind CUCM using URI dialing.



Route Class Signaling Enabled* Default

Use Trusted Relay Point* Default

☑ PSTN Access

☐ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

Call Routing Information

☑ Remote-Party-Id

☑ Asserted-Identity

Asserted-Type* PAI

SIP Privacy* Default

Inbound Calls

Significant Digits* All

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☑ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings | Default Prefix Settings

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

Connected Party Settings

Connected Party Transformation CSS < None >

☑ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS < None >

☑ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

☑ Use Device Pool Calling Party Transformation CSS

Calling Party Selection* Originator

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Calling and Connected Party Info Format* Deliver URI and DN in connected party, if available

☑ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

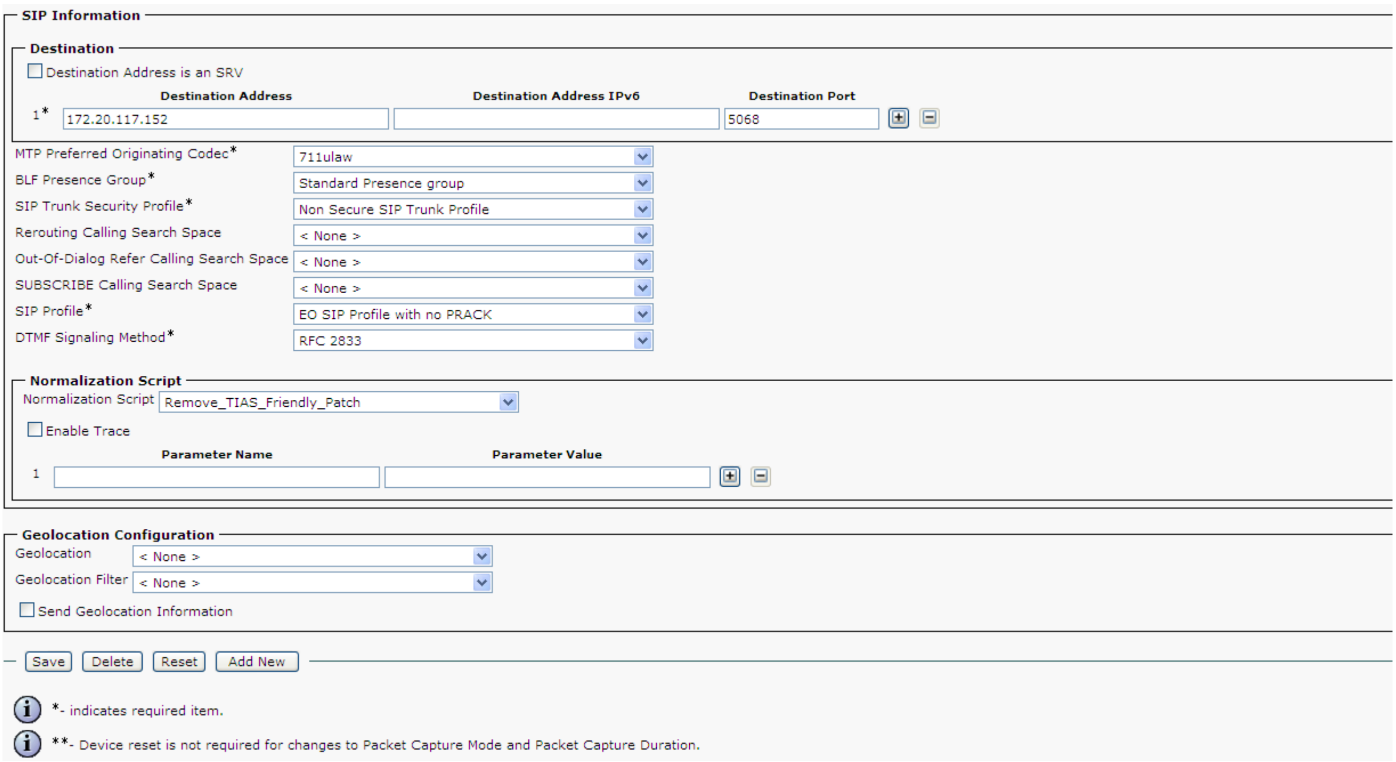
☑ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

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


SIP Information

Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	172.20.117.152		5068

MTP Preferred Originating Codec*	711ulaw
BLF Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	EO SIP Profile with no PRACK
DTMF Signaling Method*	RFC 2833

Normalization Script

Normalization Script: Remove_TIAS_Friendly_Patch

Enable Trace

	Parameter Name	Parameter Value
1		

Geolocation Configuration

Geolocation	< None >
Geolocation Filter	< None >

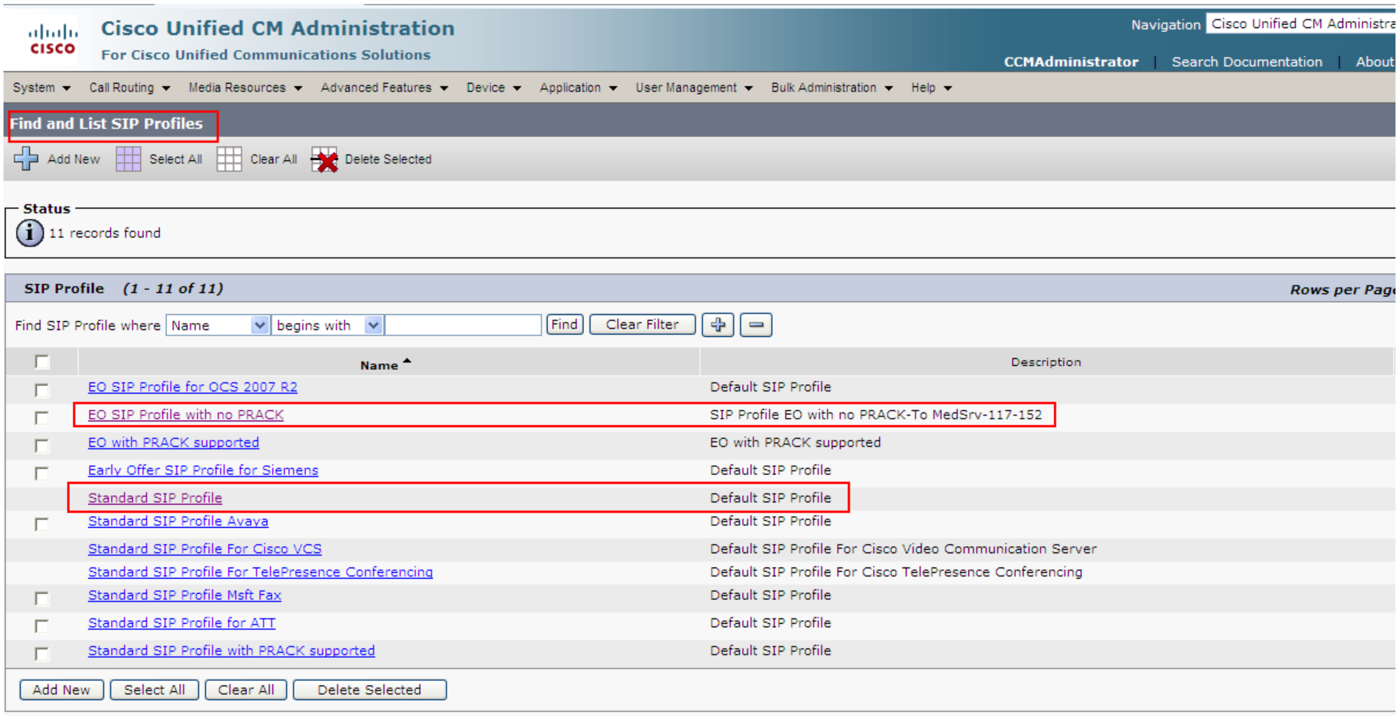
Send Geolocation Information

Save | Delete | Reset | Add New

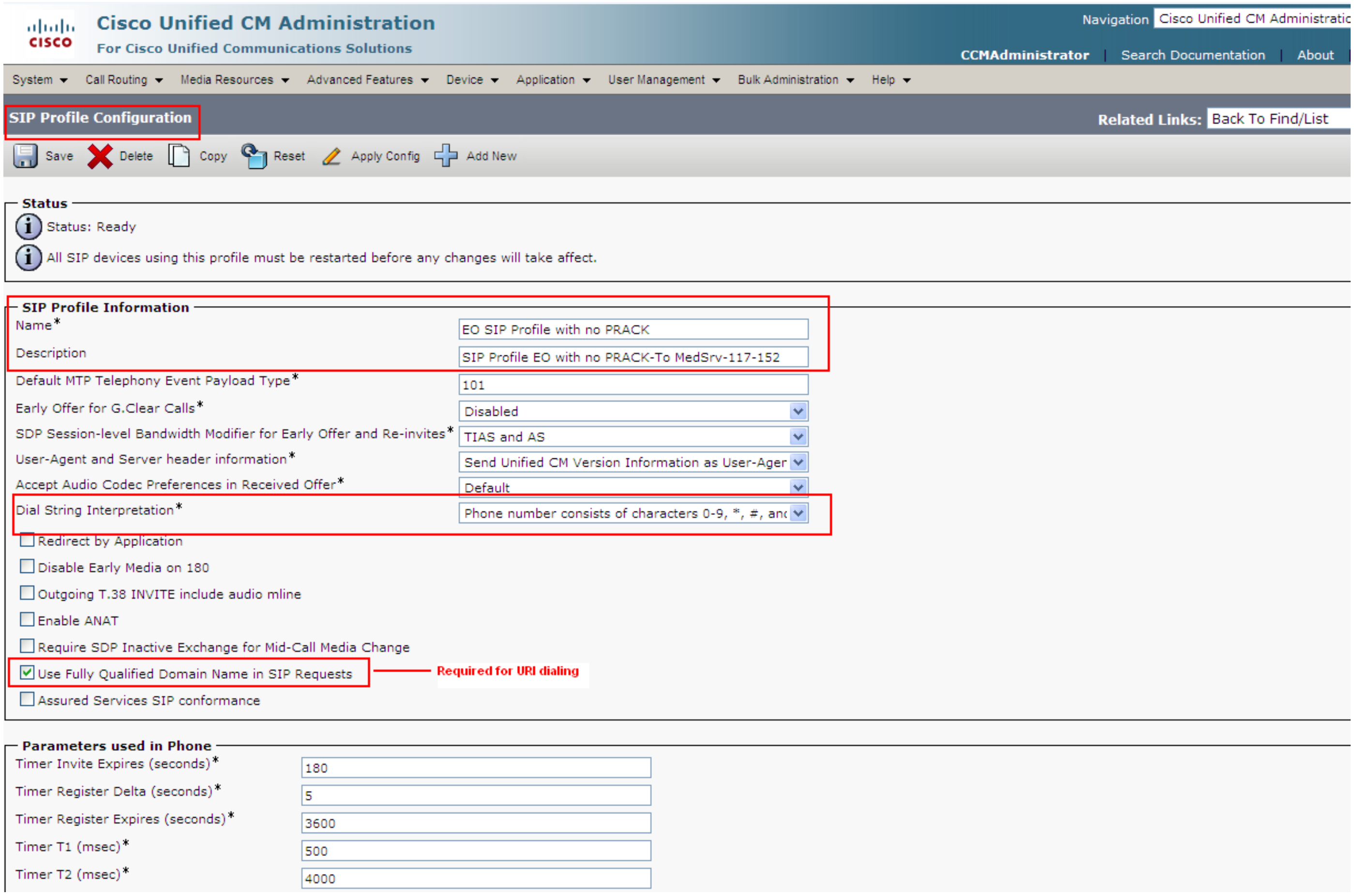
*- indicates required item.

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

Cisco Unified Communications Manager-SME SIP Profile Information



SIP Profile used with Cisco UCM-SME SIP trunk to Microsoft Lync Server 2010



Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD

- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script: Remove_TIAS_Friendly_Patch

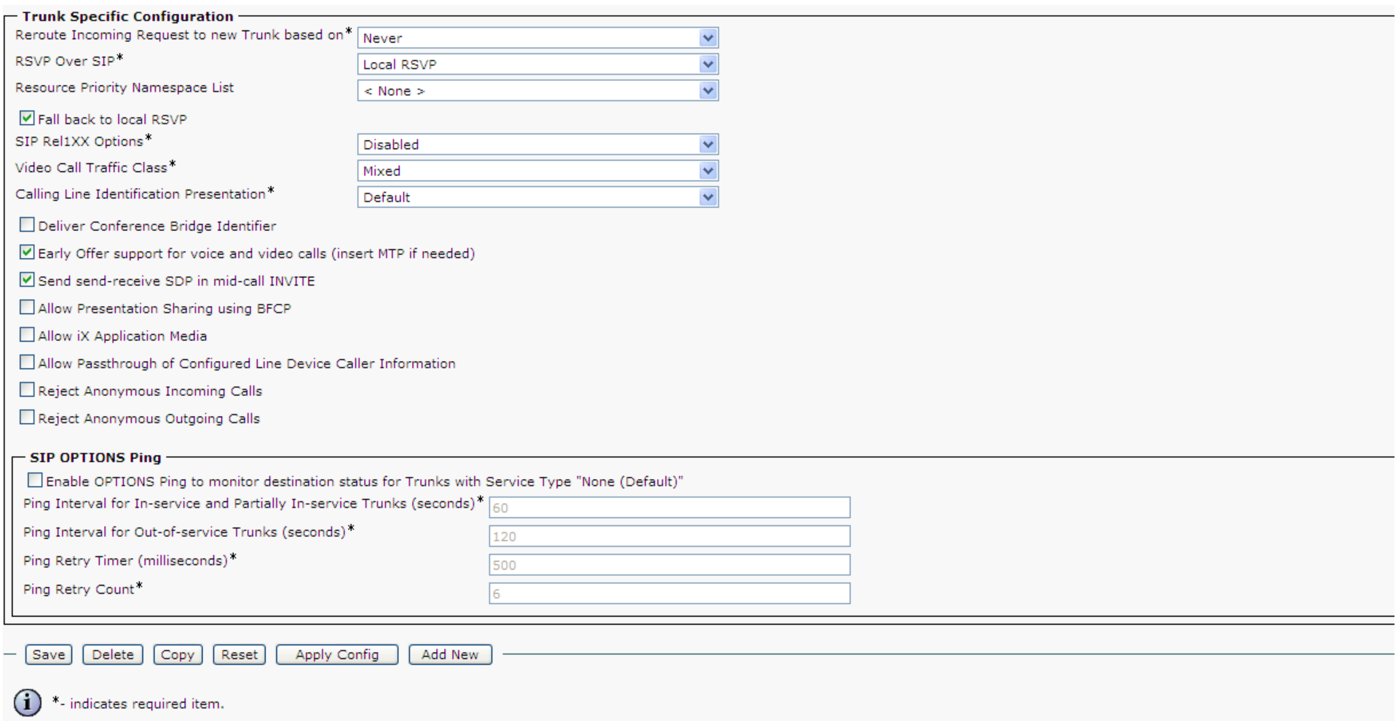
- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

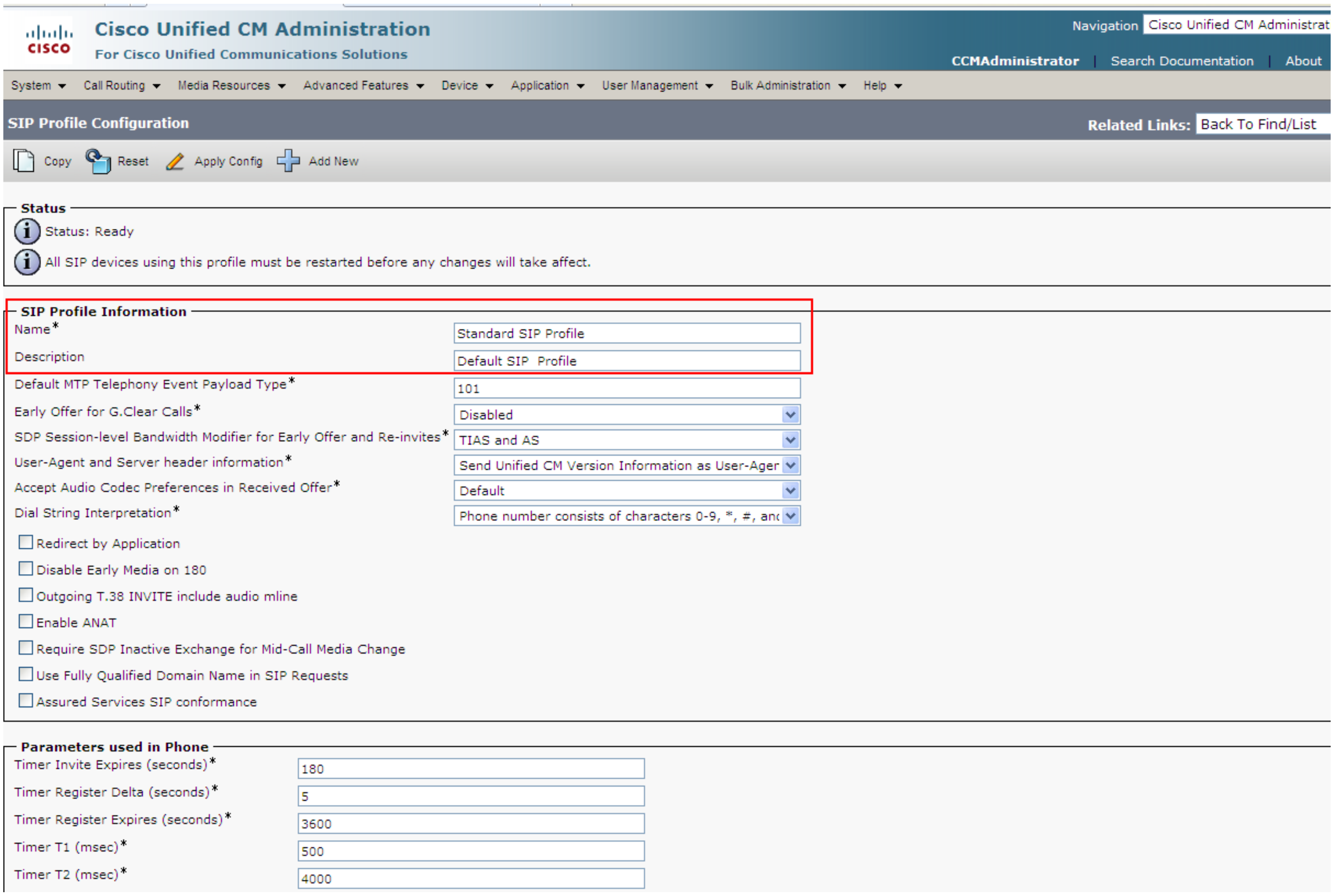
Incoming Requests FROM URI Settings

Caller ID DN

Caller Name


Trunk Specific Configuration
Reroute Incoming Request to new Trunk based on*
Never
RSVP Over SIP*
Local RSVP
Resource Priority Namespace List
< None >
Fall back to local RSVP
SIP Rel1XX Options*
Disabled
Video Call Traffic Class*
Mixed
Calling Line Identification Presentation*
Default
Deliver Conference Bridge Identifier
Early Offer support for voice and video calls (insert MTP if needed)
Send send-receive SDP in mid-call INVITE
Allow Presentation Sharing using BFCP
Allow iX Application Media
Allow Passthrough of Configured Line Device Caller Information
Reject Anonymous Incoming Calls
Reject Anonymous Outgoing Calls
SIP OPTIONS Ping
Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"
Ping Interval for In-service and Partially In-service Trunks (seconds)*
60
Ping Interval for Out-of-service Trunks (seconds)*
120
Ping Retry Timer (milliseconds)*
500
Ping Retry Count*
6
Save
Delete
Copy
Reset
Apply Config
Add New
*- indicates required item.

SIP Profile used with Cisco UCM-SME SIP trunk to Cisco Unified Communications Manager



Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script: < None >

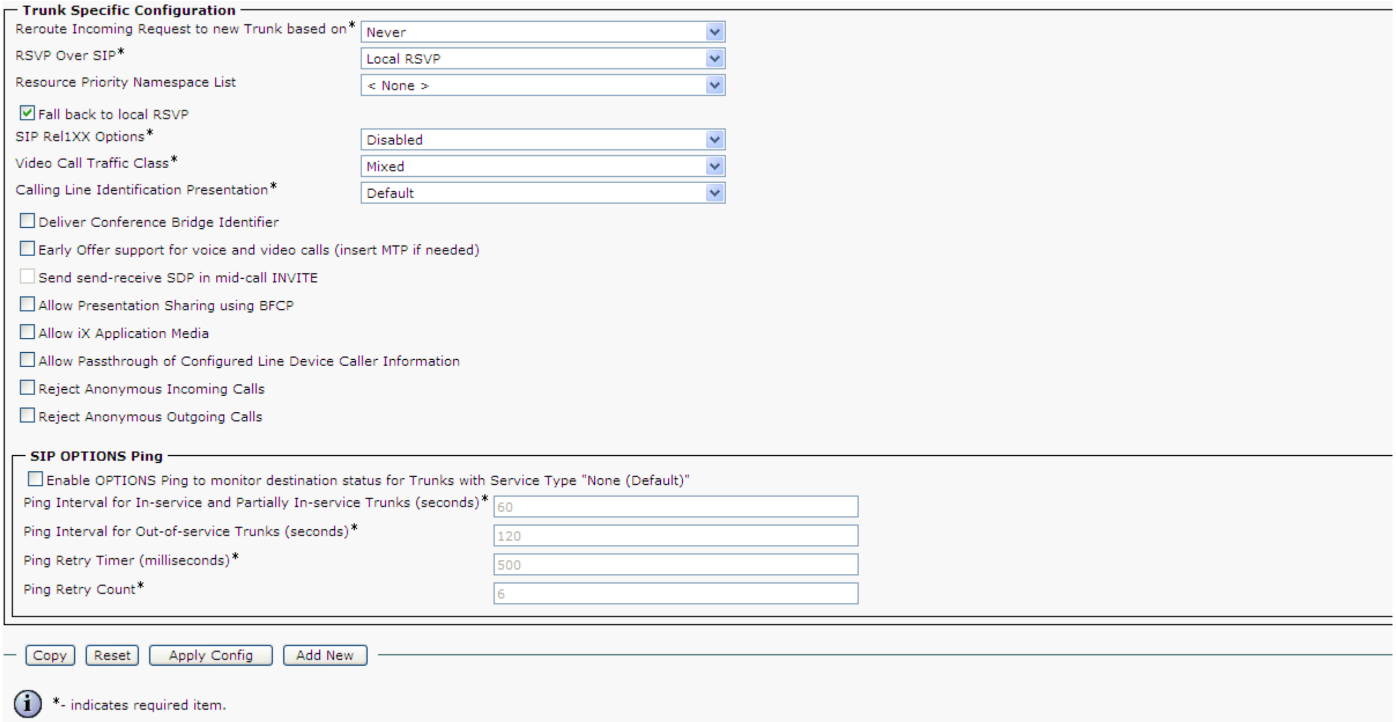
- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name



Trunk Specific Configuration

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

☑ Fall back to local RSVP

SIP Rel1XX Options*	Disabled
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default

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

SIP OPTIONS Ping

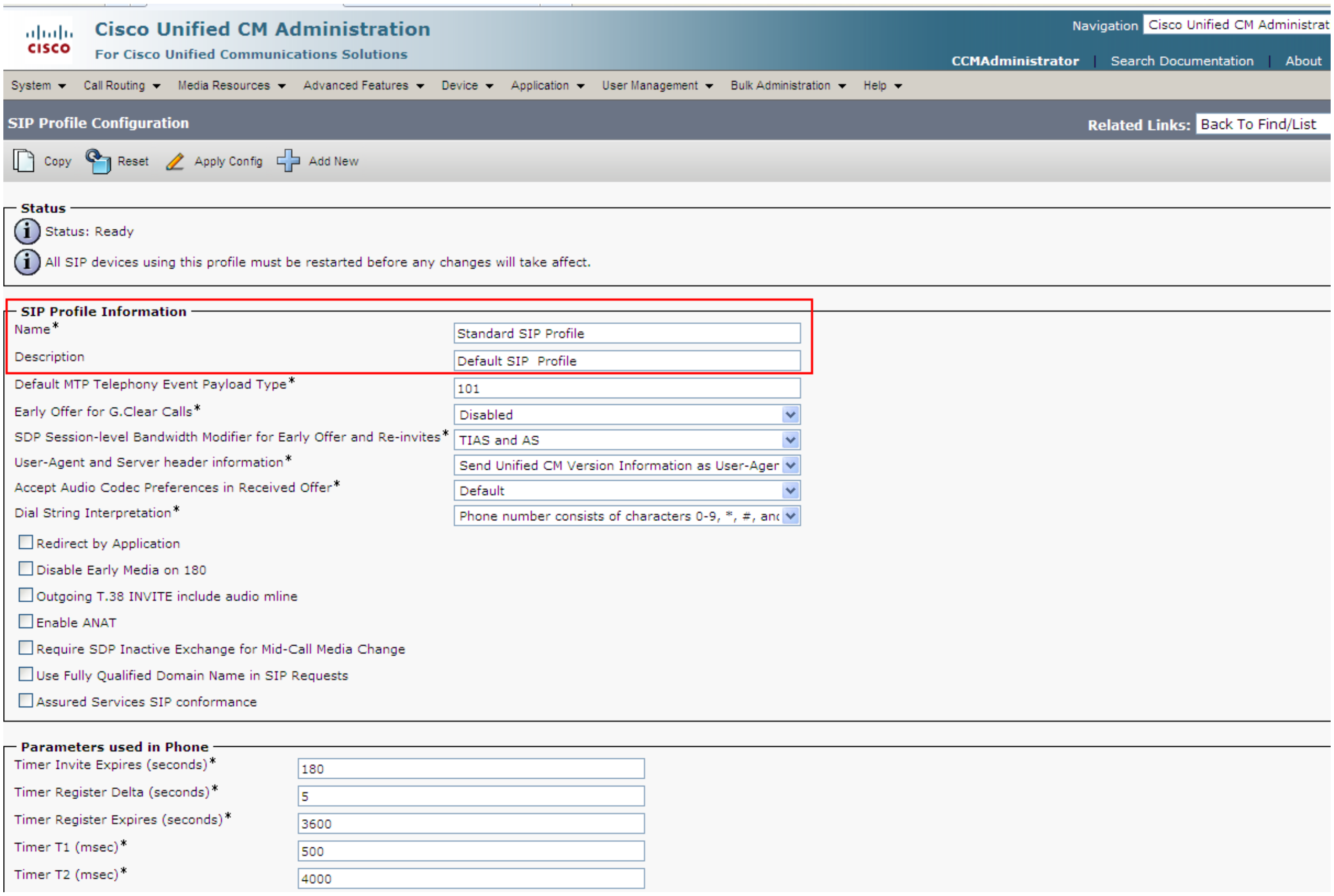
☐ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

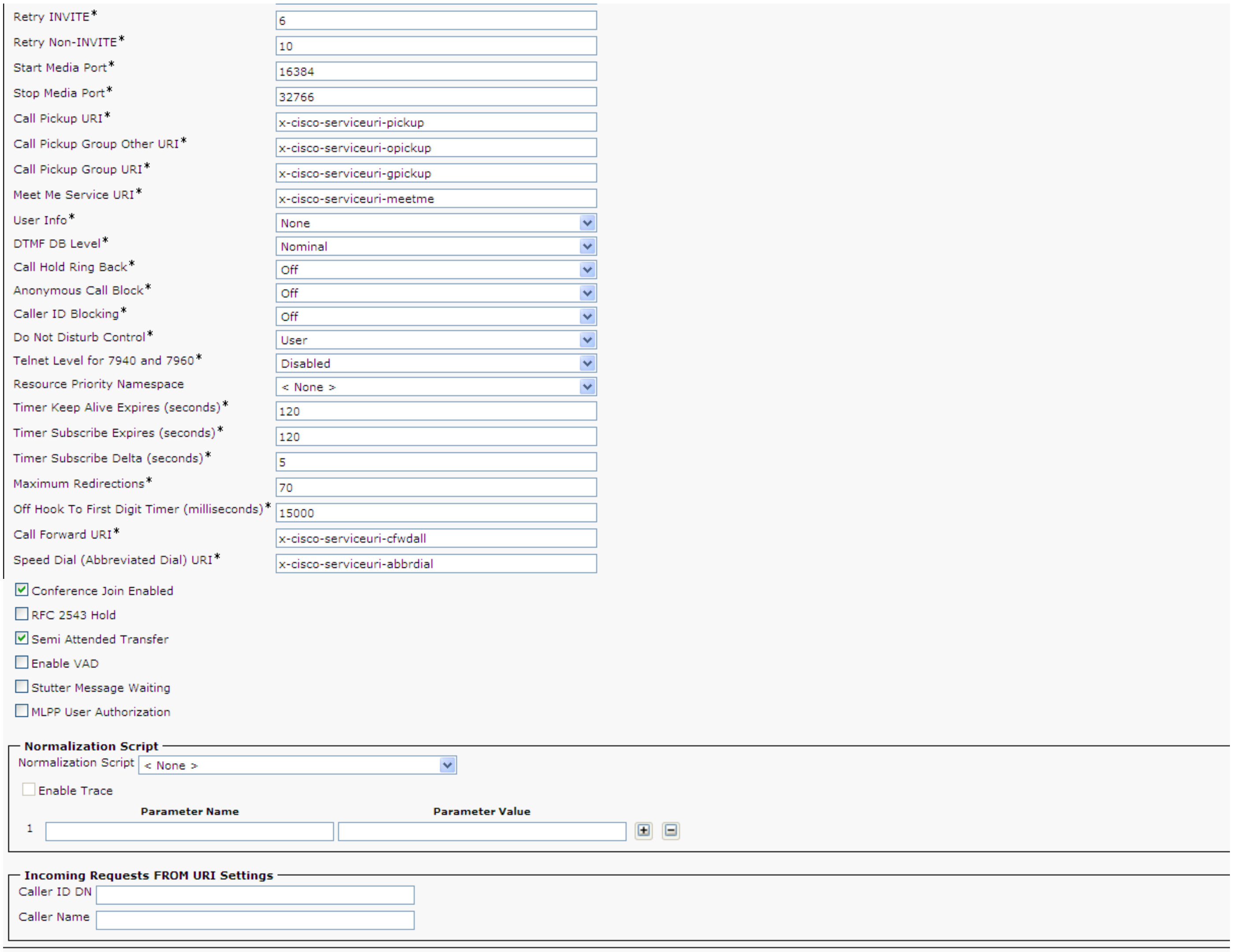
Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

Copy | Reset | Apply Config | Add New

*- indicates required item.

SIP Profile used with Cisco UCM-SME SIP trunk to Service Provider





Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script: < None >

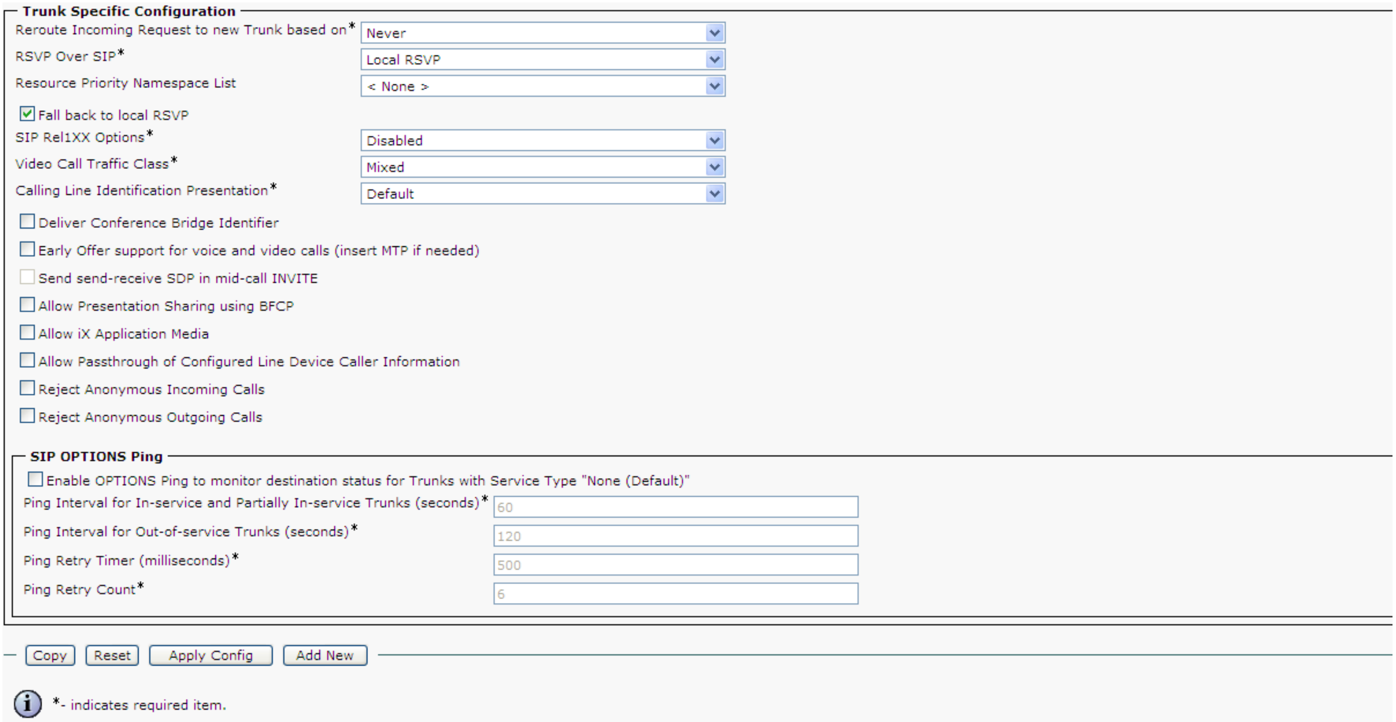
- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

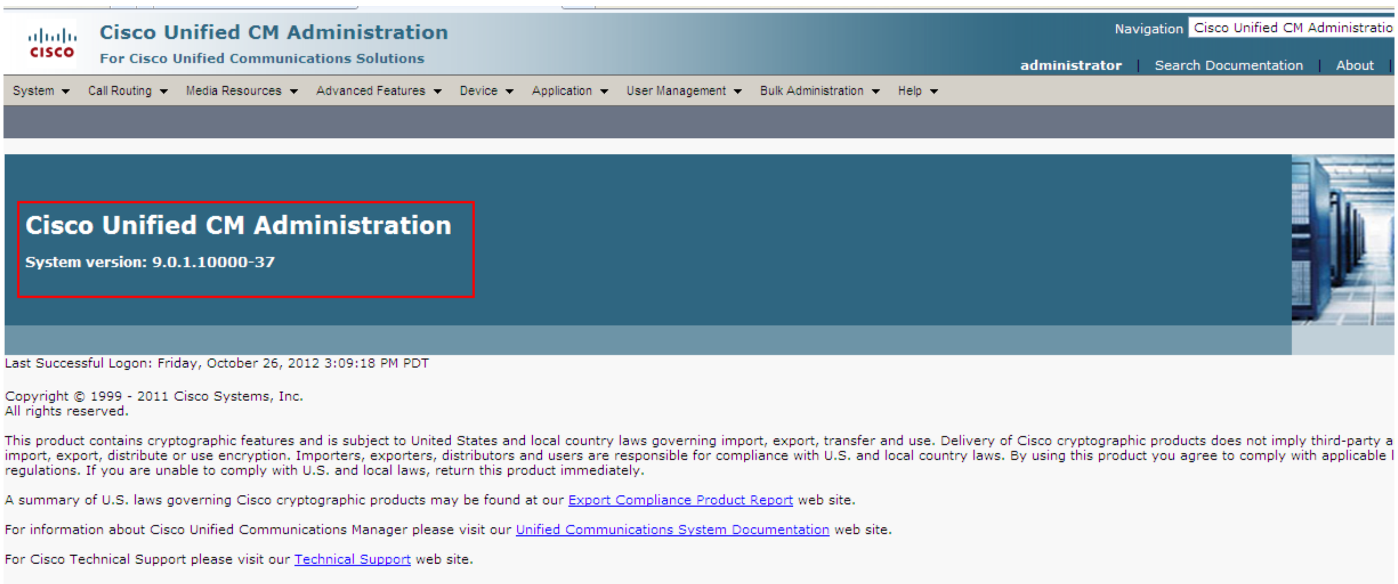
Caller ID DN

Caller Name

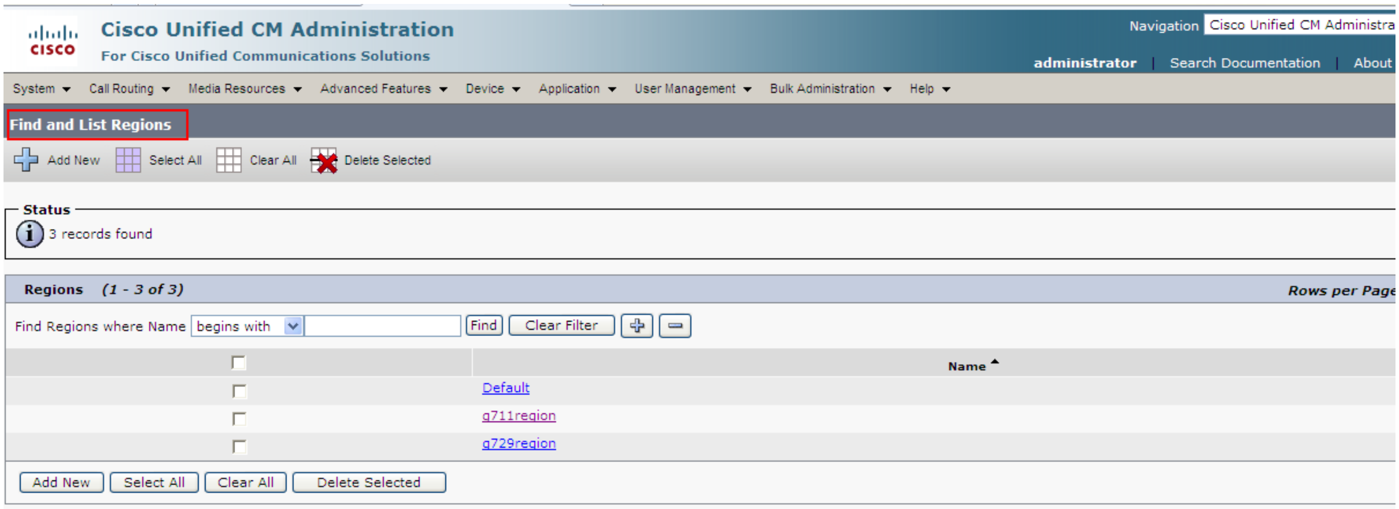

Trunk Specific Configuration
Reroute Incoming Request to new Trunk based on* Never
RSVP Over SIP* Local RSVP
Resource Priority Namespace List < None >
Fall back to local RSVP
SIP Rel1XX Options* Disabled
Video Call Traffic Class* Mixed
Calling Line Identification Presentation* Default
Deliver Conference Bridge Identifier
Early Offer support for voice and video calls (insert MTP if needed)
Send send-receive SDP in mid-call INVITE
Allow Presentation Sharing using BFCP
Allow iX Application Media
Allow Passthrough of Configured Line Device Caller Information
Reject Anonymous Incoming Calls
Reject Anonymous Outgoing Calls
SIP OPTIONS Ping
Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"
Ping Interval for In-service and Partially In-service Trunks (seconds)* 60
Ping Interval for Out-of-service Trunks (seconds)* 120
Ping Retry Timer (milliseconds)* 500
Ping Retry Count* 6
Copy
Reset
Apply Config
Add New
*- indicates required item.

Configuring the Cisco Unified Communications Manager

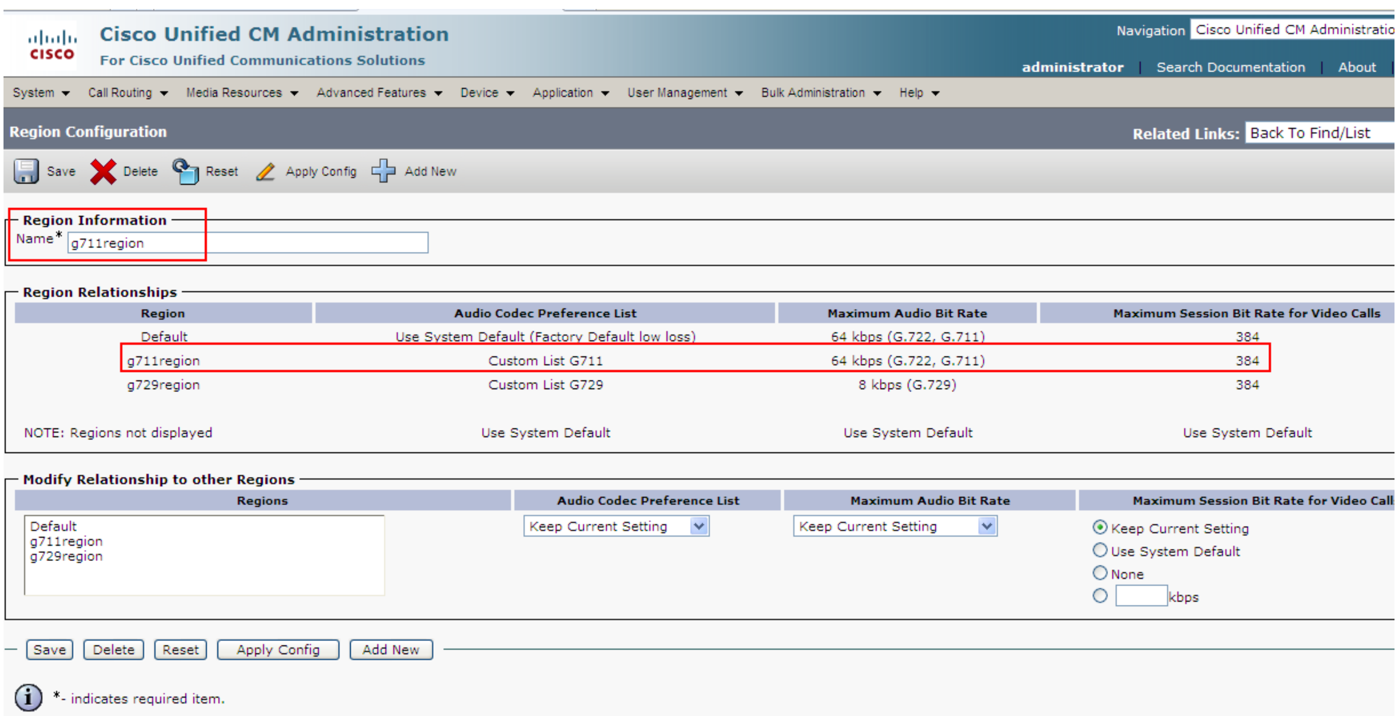
Cisco Unified Communications Manager-Software release



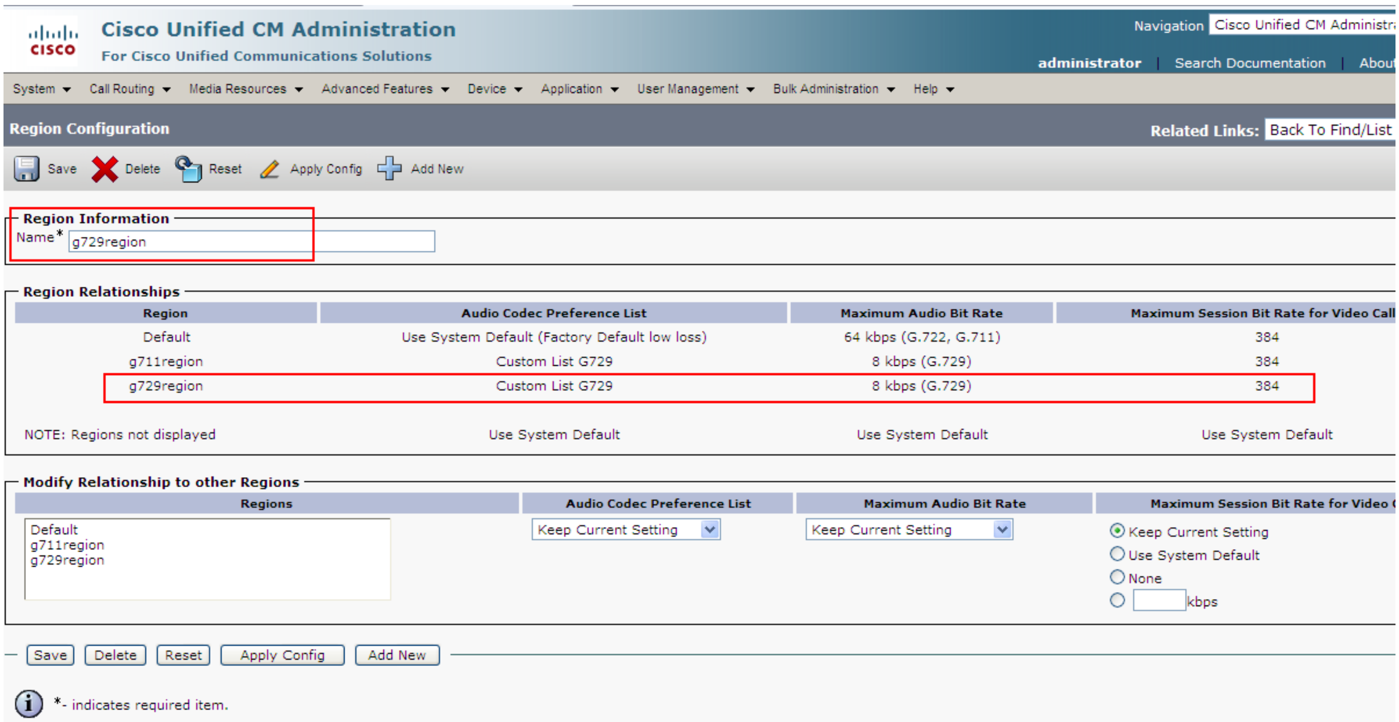
Cisco Unified Communications Manager-Region Configuration



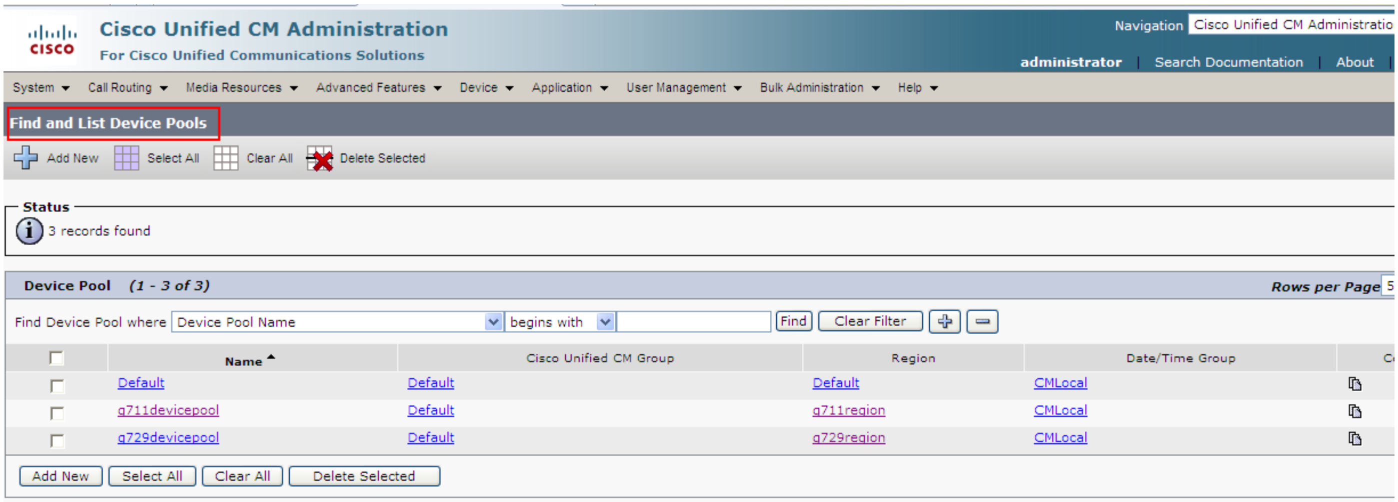
Cisco UCM Regions Configuration – G711 Region



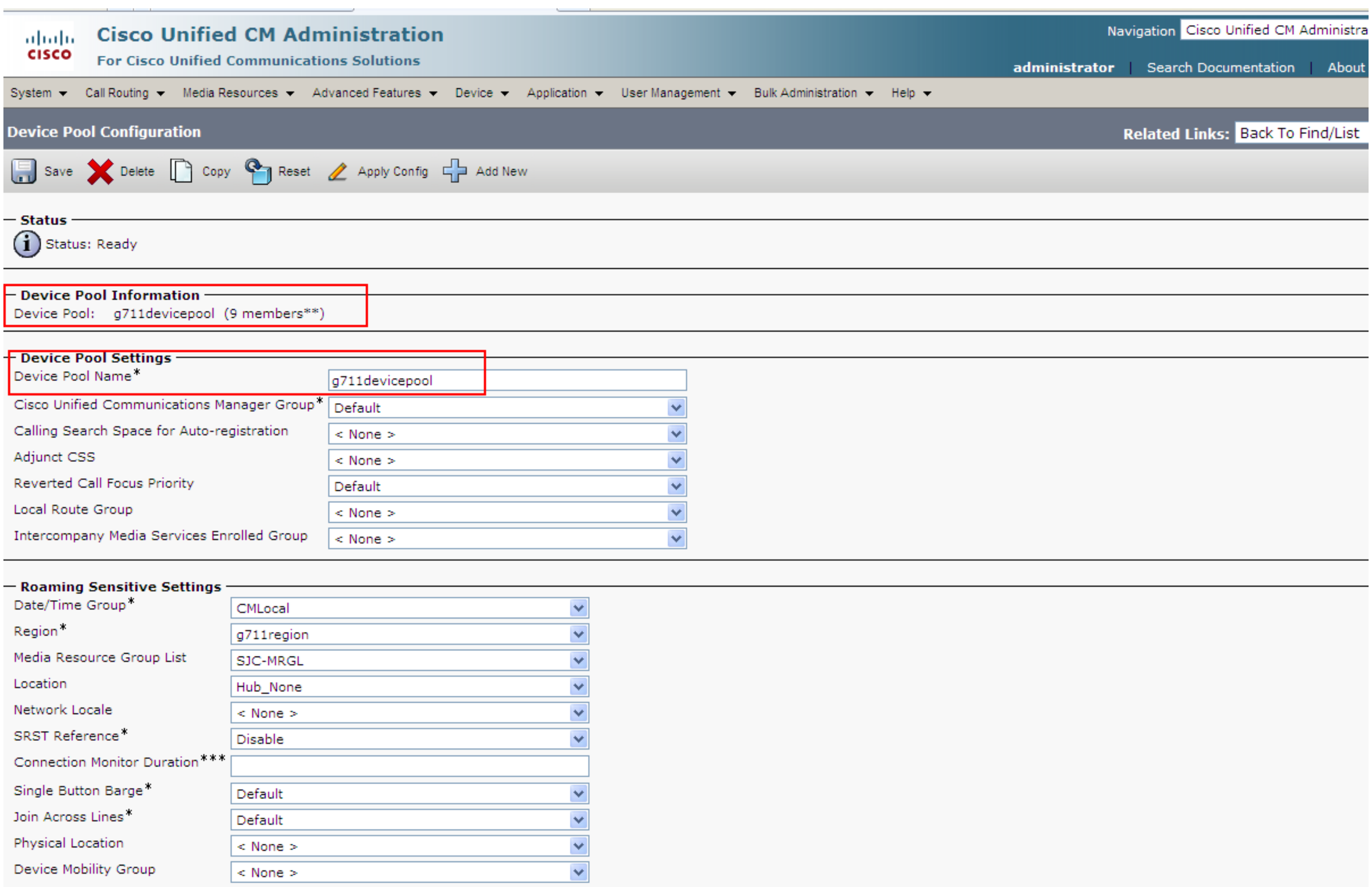
Cisco UCM Regions Configuration – G729 Region

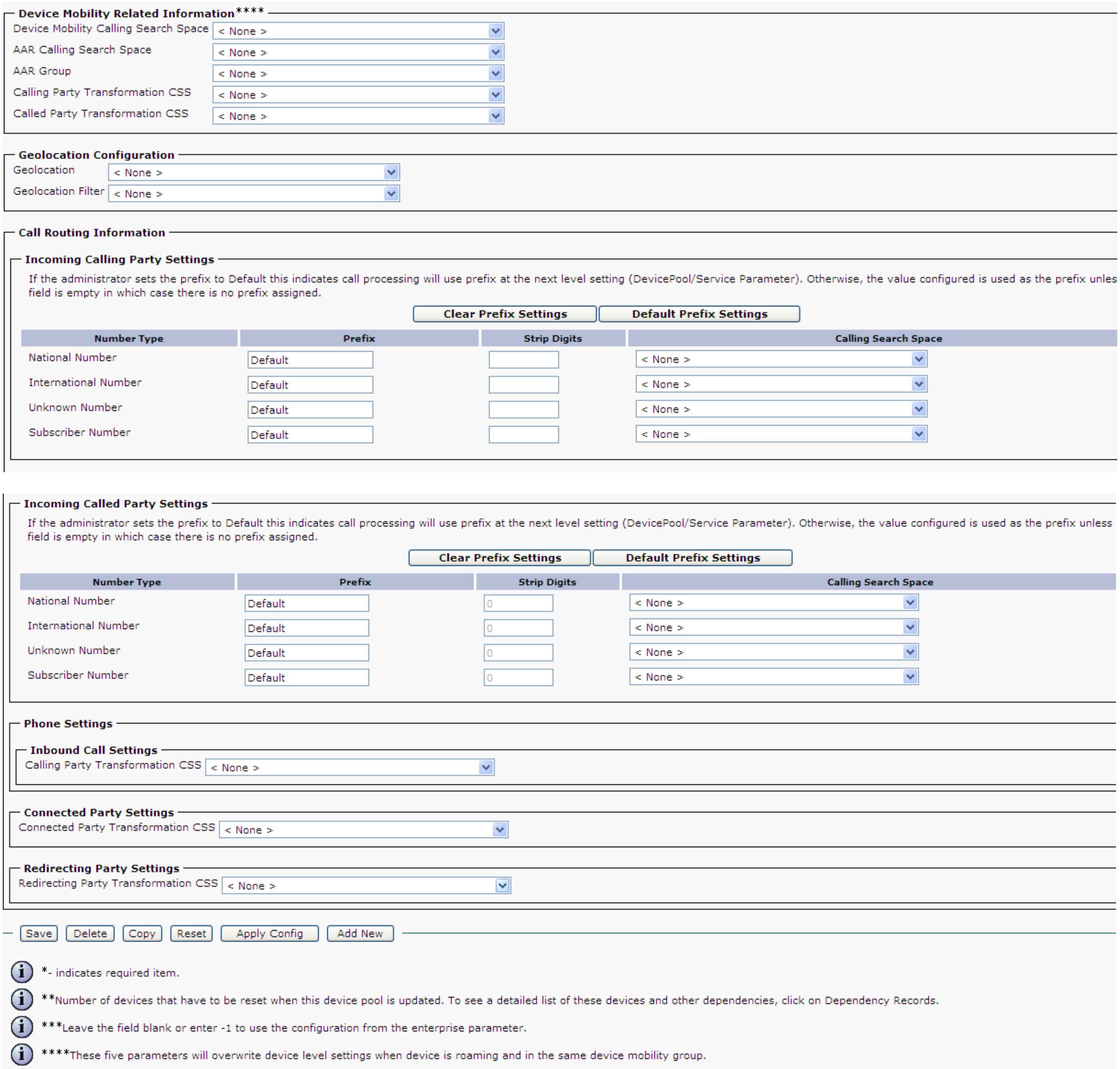


Cisco Unified Communications Manager-Device Pool configuration

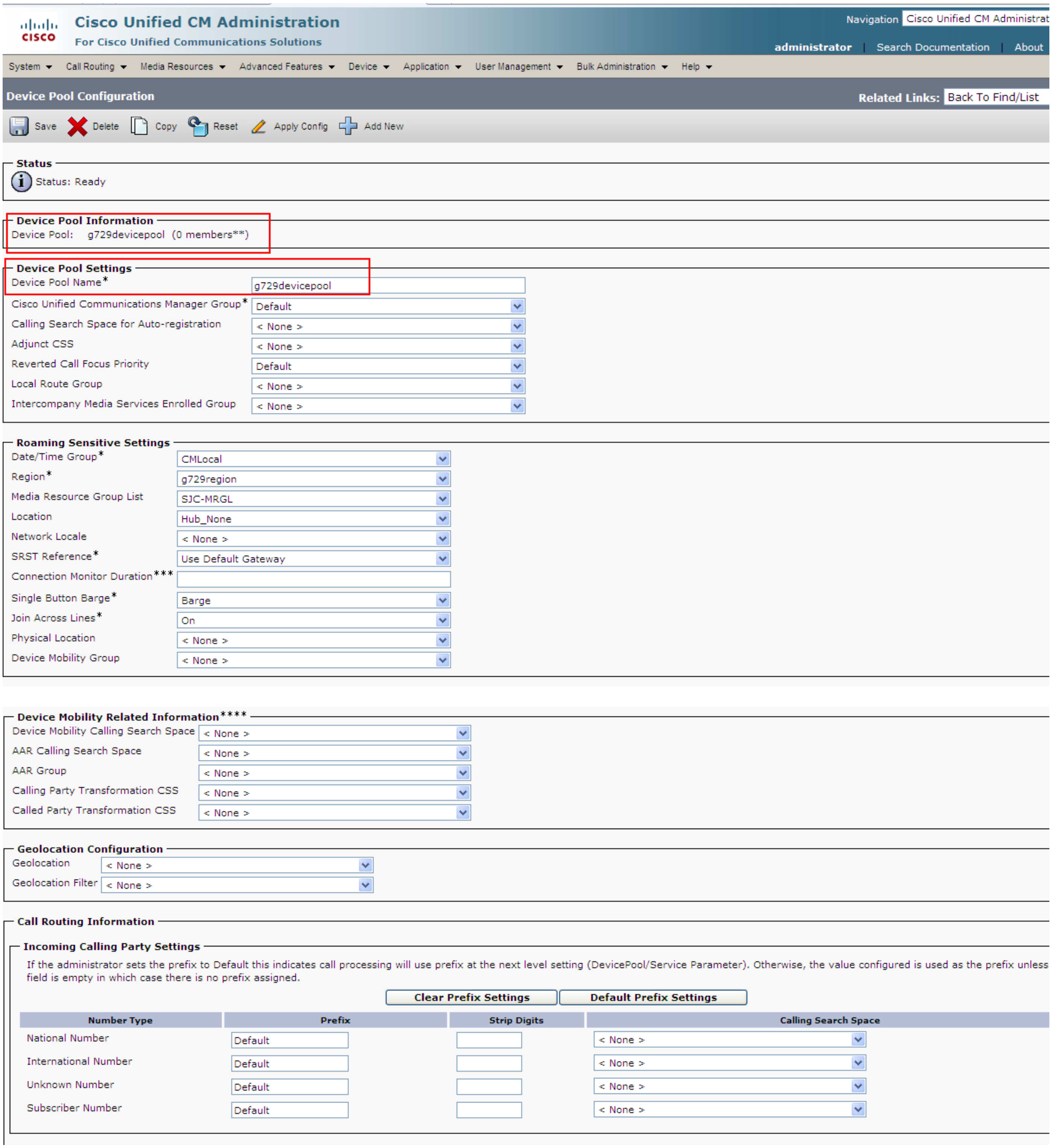


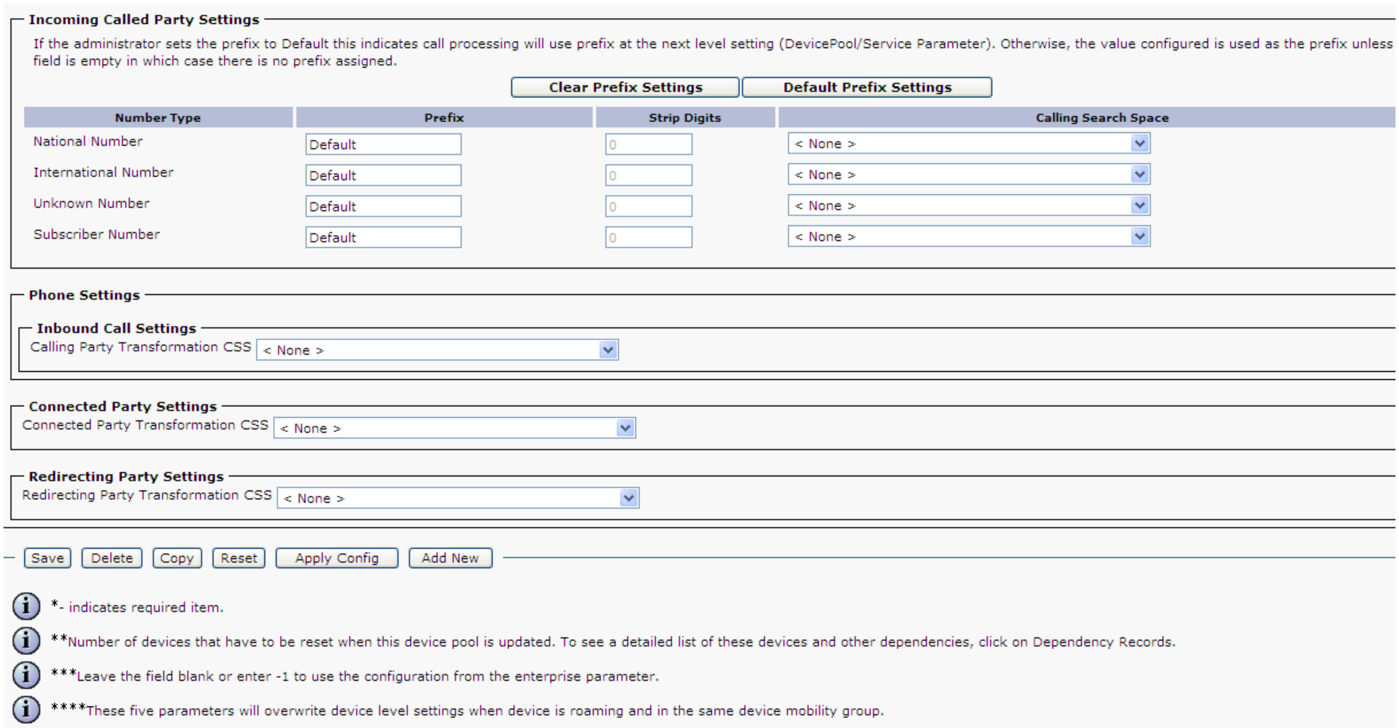
Cisco UCM Device Pool for G711




Device Mobility Related Information****
Device Mobility Calling Search Space
< None >
AAR Calling Search Space
< None >
AAR Group
< None >
Calling Party Transformation CSS
< None >
Called Party Transformation CSS
< None >
Geolocation Configuration
Geolocation
< None >
Geolocation Filter
< None >
Call Routing Information
Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unles
field is empty in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
National Number
Default
< None >
International Number
Default
< None >
Unknown Number
Default
< None >
Subscriber Number
Default
< None >
Incoming Called Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless
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
0
< None >
International Number
Default
0
< None >
Unknown Number
Default
0
< None >
Subscriber Number
Default
0
< None >
Phone Settings
Inbound Call Settings
Calling Party Transformation CSS
< None >
Connected Party Settings
Connected Party Transformation CSS
< None >
Redirecting Party Settings
Redirecting Party Transformation CSS
< None >
Save
Delete
Copy
Reset
Apply Config
Add New
*- indicates required item.
**Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.
***Leave the field blank or enter -1 to use the configuration from the enterprise parameter.
****These five parameters will overwrite device level settings when device is roaming and in the same device mobility group.

Cisco UCM Device Pool for G729





Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless 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	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >

Phone Settings

Inbound Call Settings

Calling Party Transformation CSS < None >

Connected Party Settings

Connected Party Transformation CSS < None >

Redirecting Party Settings

Redirecting Party Transformation CSS < None >

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

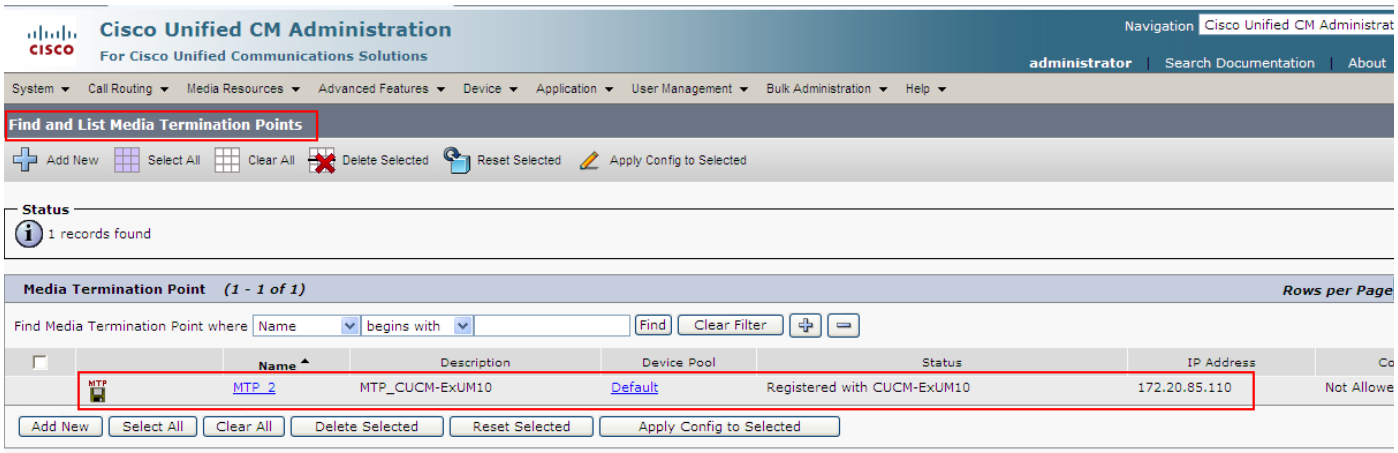
*- indicates required item.

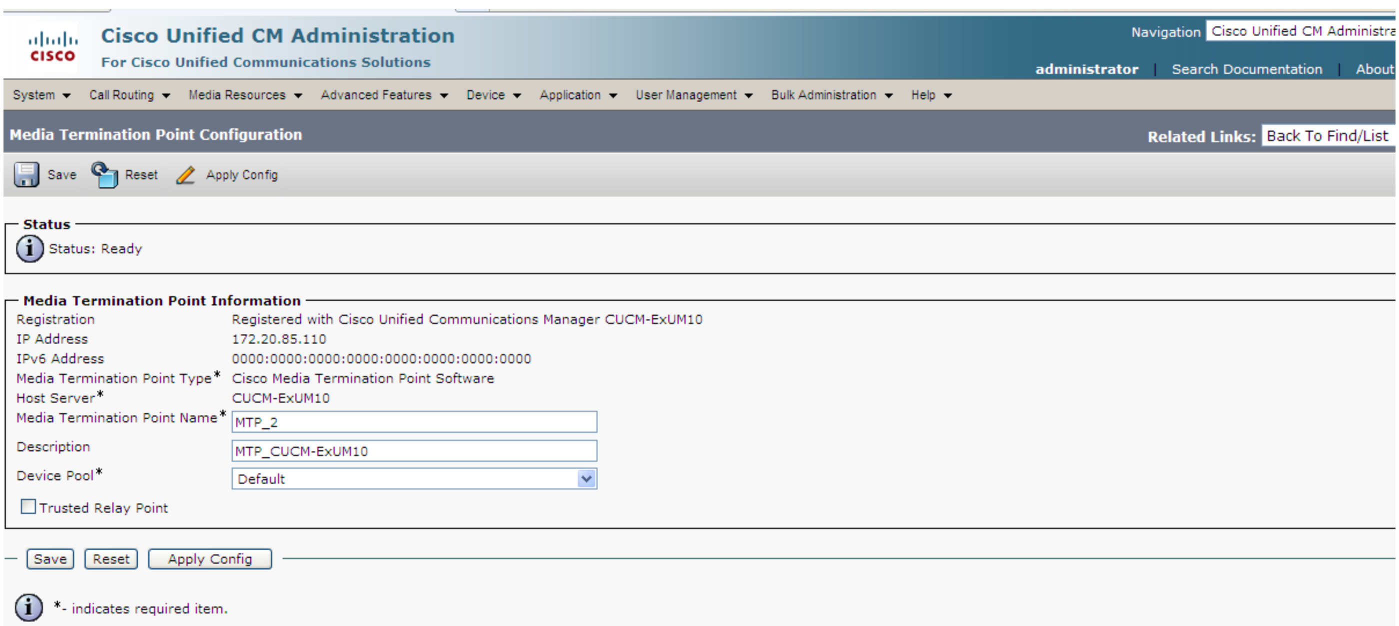
**Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.

***Leave the field blank or enter -1 to use the configuration from the enterprise parameter.

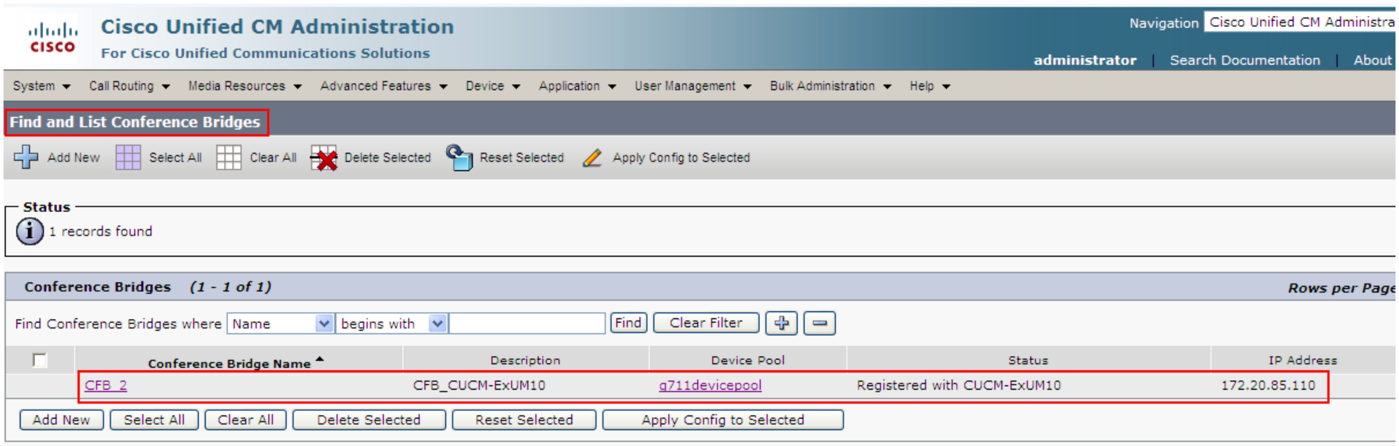
****These five parameters will overwrite device level settings when device is roaming and in the same device mobility group.

Cisco Unified Communications Manager-Media Termination Point Configuration

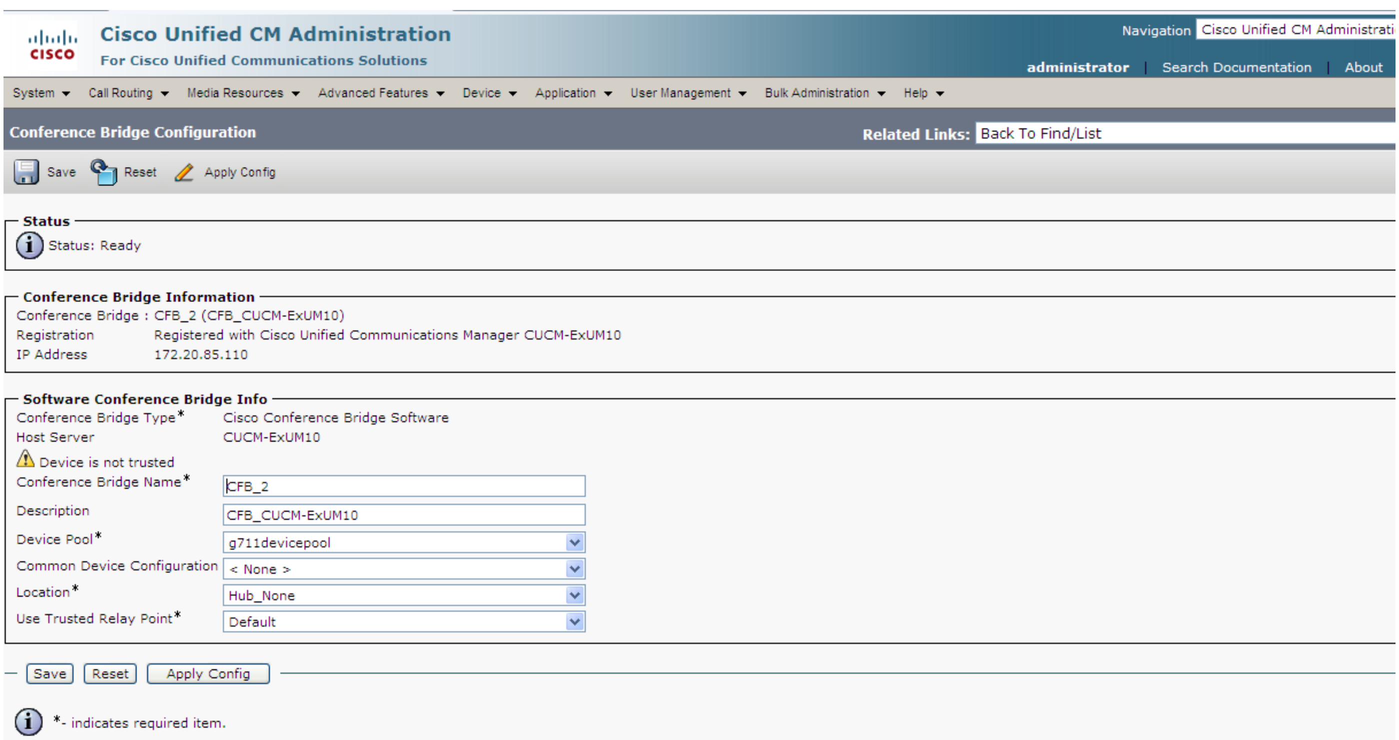




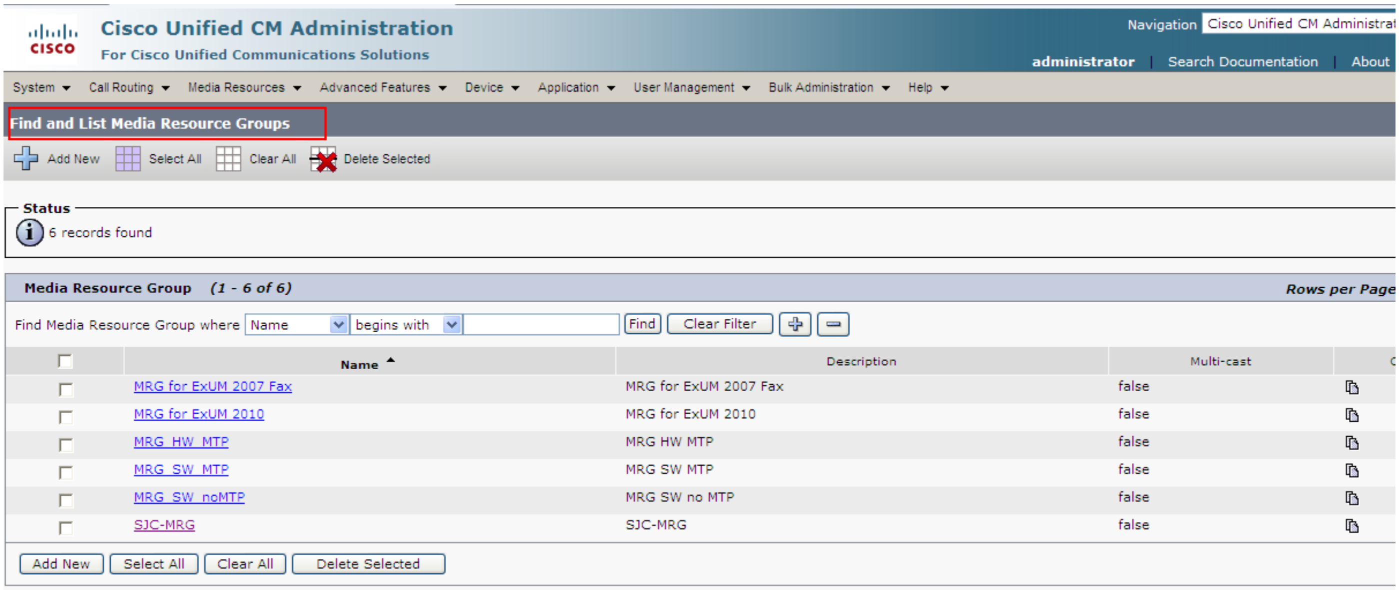
Cisco Unified Communications Manager-Conference Bridge Configuration

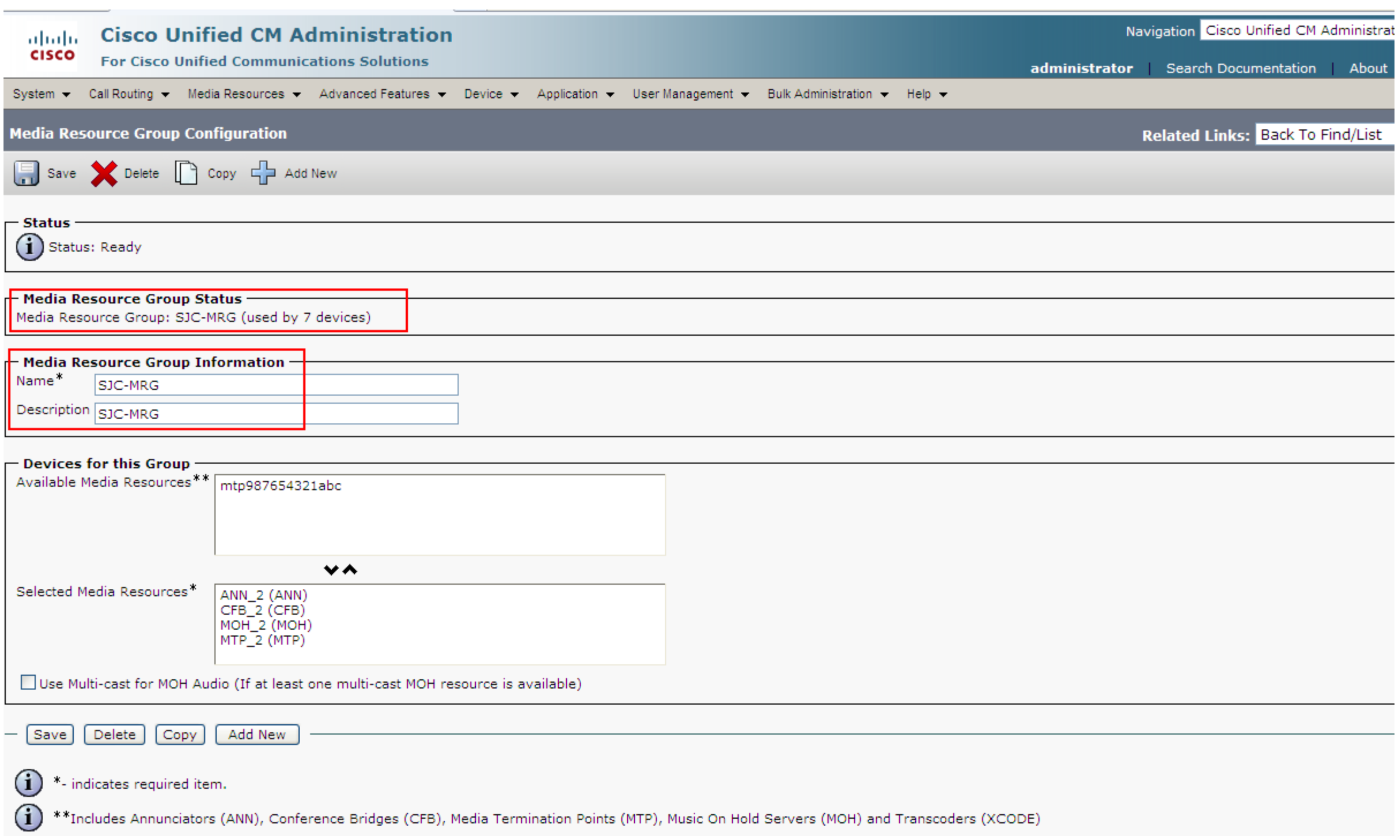


Software Conference Bridge

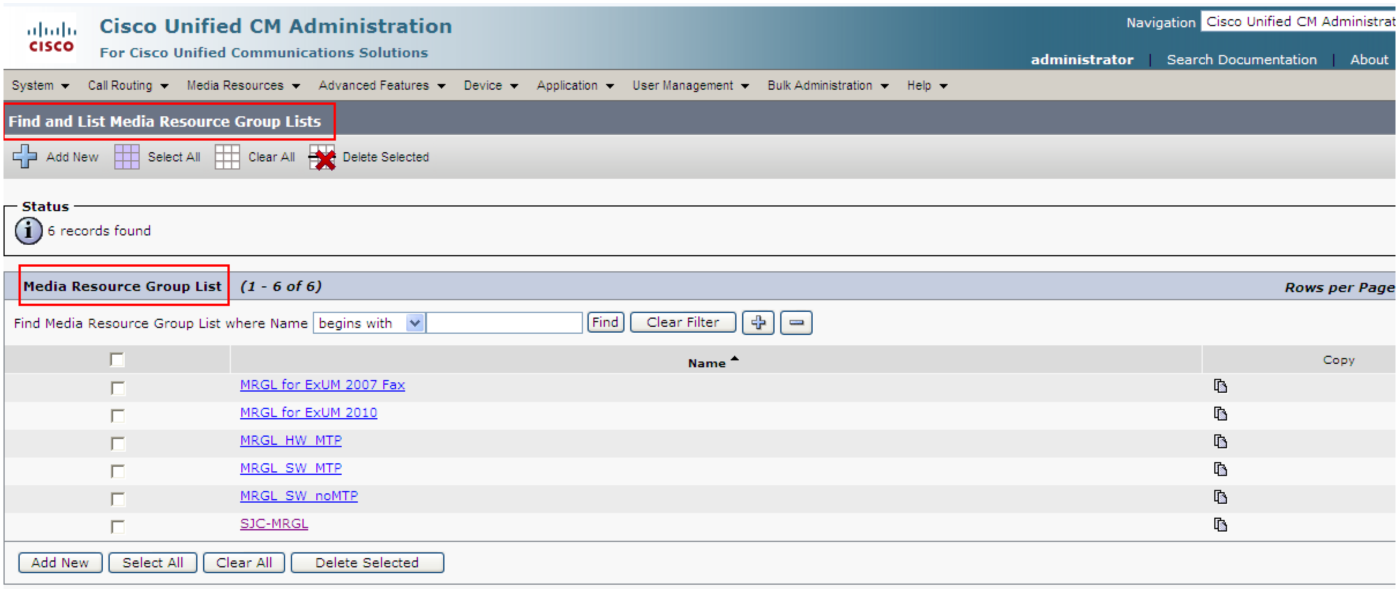


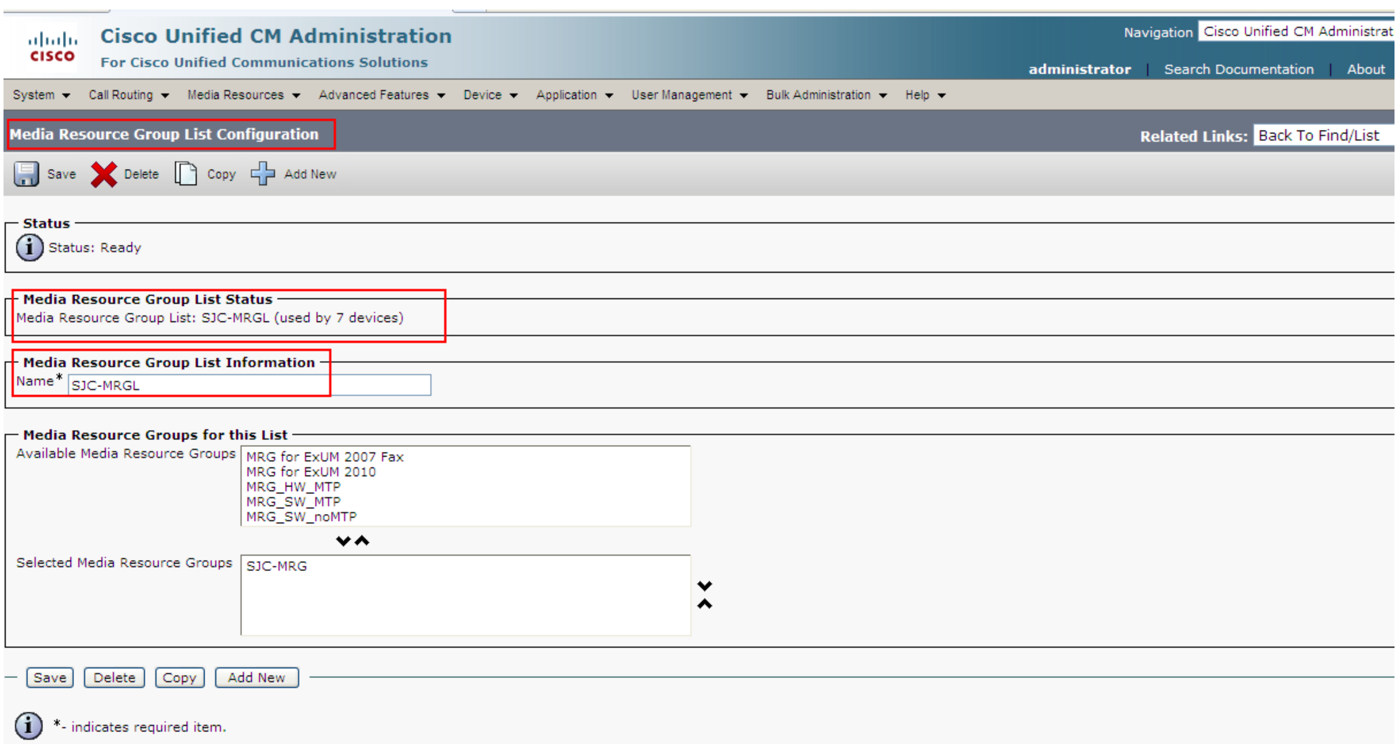
Cisco Unified Communications Manager-Media Resource Group configuration

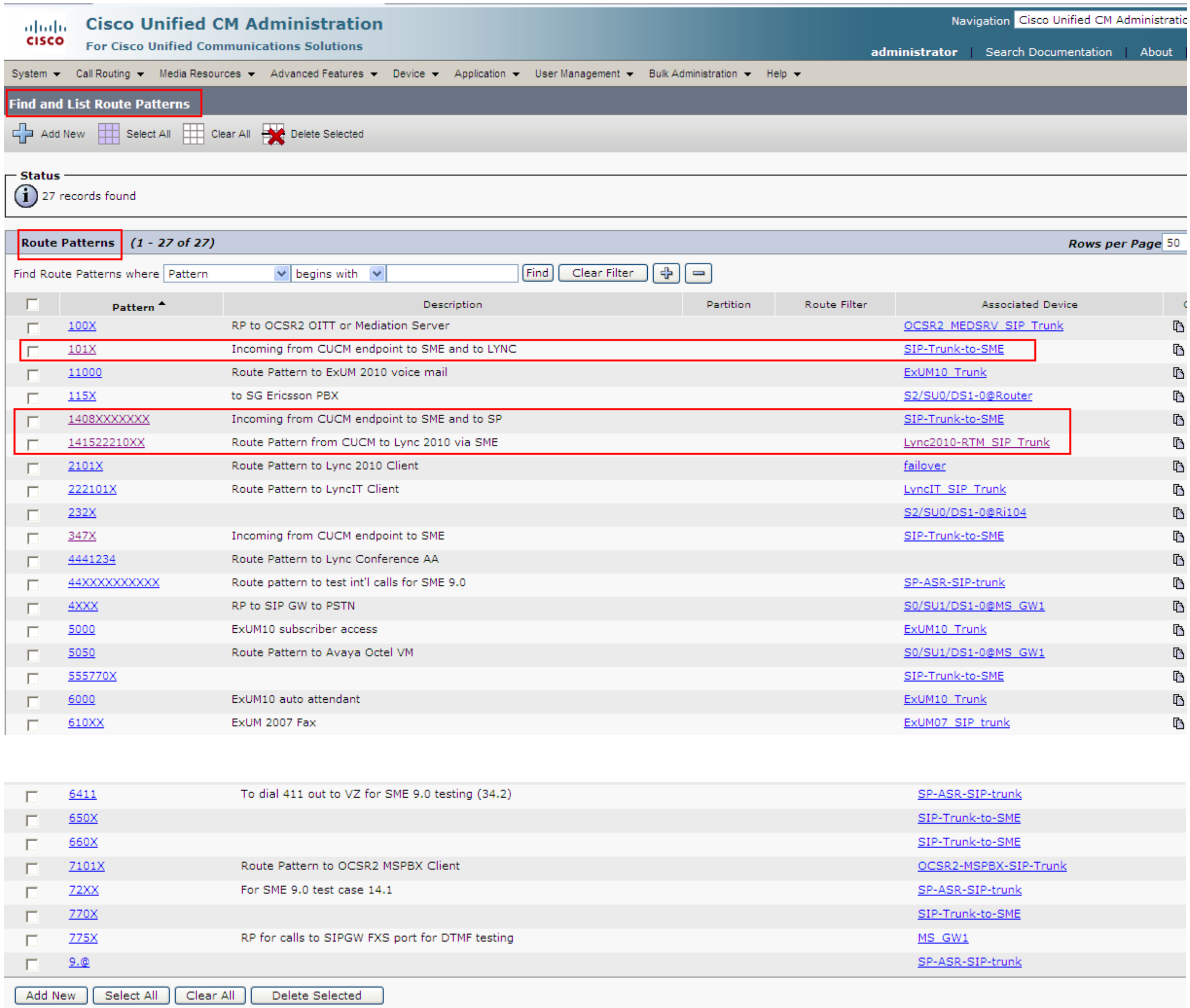




Cisco Unified Communications Manager-Media Resource Group List configuration

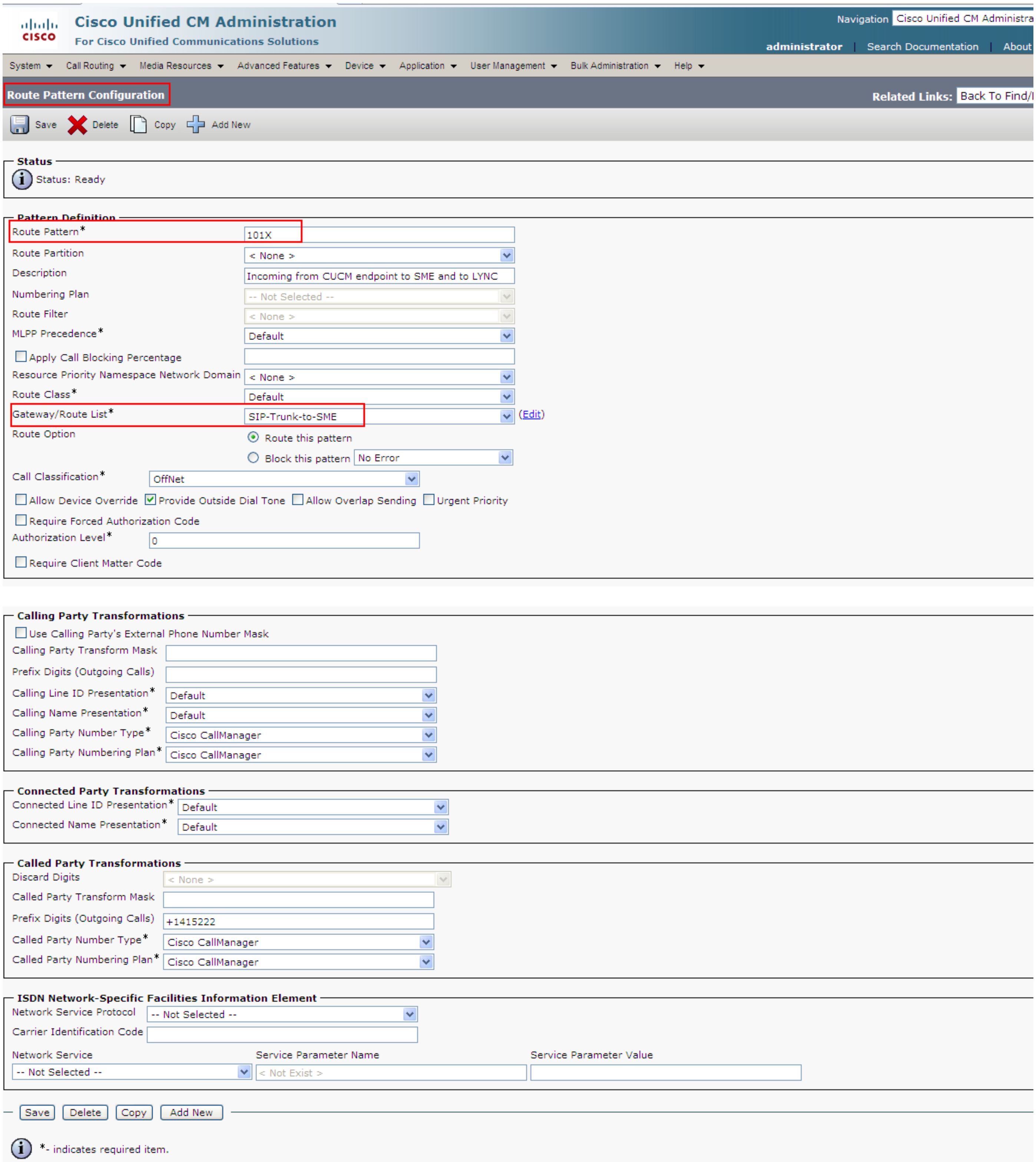




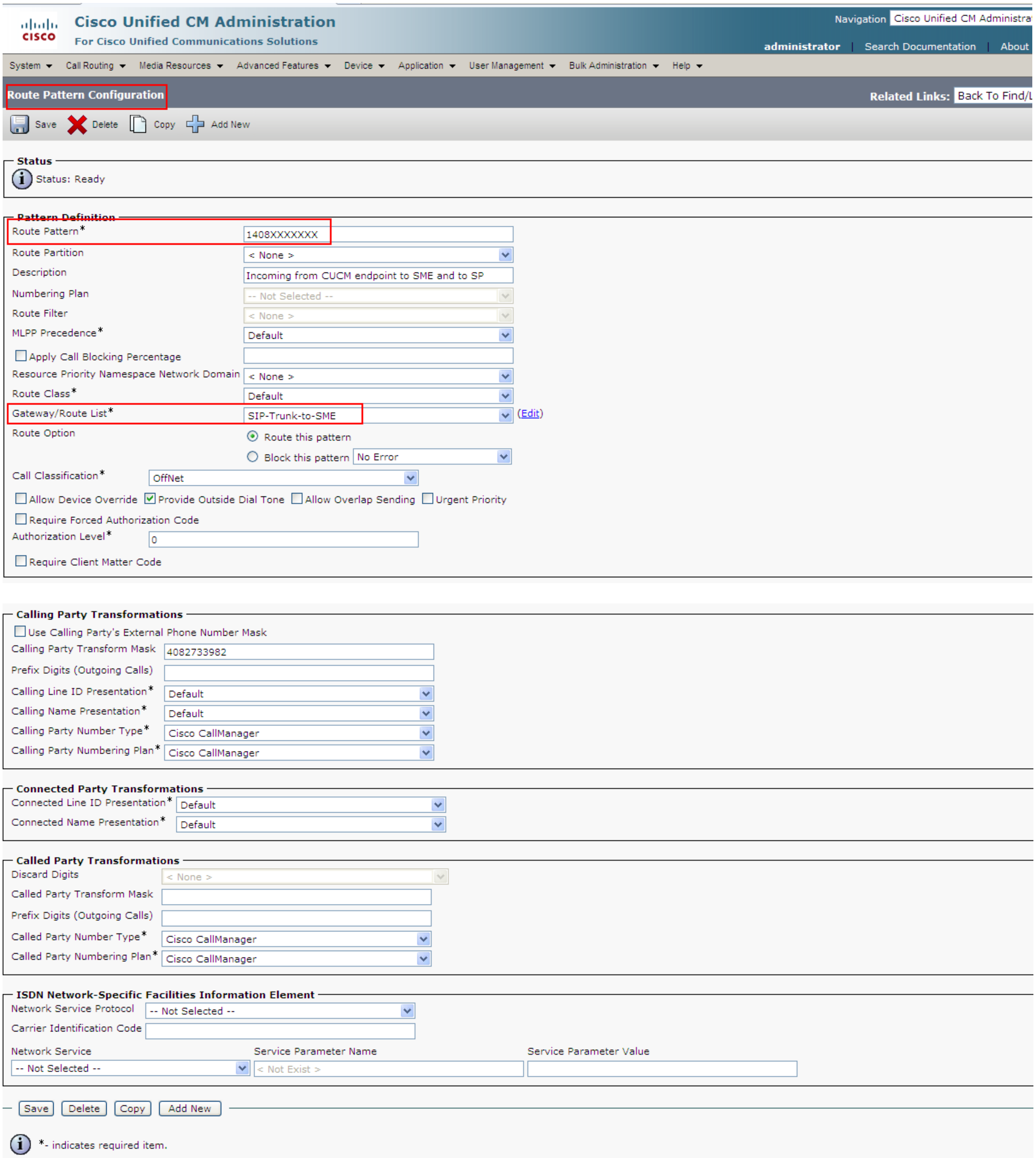


Pattern	Description	Partition	Route Filter	Associated Device
100X	RP to OCSR2 OITT or Mediation Server			OCSR2_MEDSRV_SIP_Trunk
101X	Incoming from CUCM endpoint to SME and to LYNC			SIP-Trunk-to-SME
11000	Route Pattern to ExUM 2010 voice mail			ExUM10_Trunk
115X	to SG Ericsson PBX			S2/SU0/DS1-0@Router
1408XXXXXXX	Incoming from CUCM endpoint to SME and to SP			SIP-Trunk-to-SME
141522210XX	Route Pattern from CUCM to Lync 2010 via SME			Lync2010-RTM_SIP_Trunk
2101X	Route Pattern to Lync 2010 Client			failover
222101X	Route Pattern to LyncIT Client			LyncIT_SIP_Trunk
232X				S2/SU0/DS1-0@Ri104
347X	Incoming from CUCM endpoint to SME			SIP-Trunk-to-SME
4441234	Route Pattern to Lync Conference AA			
44XXXXXXXXXX	Route pattern to test int'l calls for SME 9.0			SP-ASR-SIP-trunk
4XXX	RP to SIP GW to PSTN			S0/SU1/DS1-0@MS_GW1
5000	ExUM10 subscriber access			ExUM10_Trunk
5050	Route Pattern to Avaya Octel VM			S0/SU1/DS1-0@MS_GW1
555770X				SIP-Trunk-to-SME
6000	ExUM10 auto attendant			ExUM10_Trunk
610XX	ExUM 2007 Fax			ExUM07_SIP_trunk
6411	To dial 411 out to VZ for SME 9.0 testing (34.2)			SP-ASR-SIP-trunk
650X				SIP-Trunk-to-SME
660X				SIP-Trunk-to-SME
7101X	Route Pattern to OCSR2 MSPBX Client			OCSR2-MSPBX-SIP-Trunk
72XX	For SME 9.0 test case 14.1			SP-ASR-SIP-trunk
770X				SIP-Trunk-to-SME
775X	RP for calls to SIPGW FXS port for DTMF testing			MS_GW1
9.@				SP-ASR-SIP-trunk

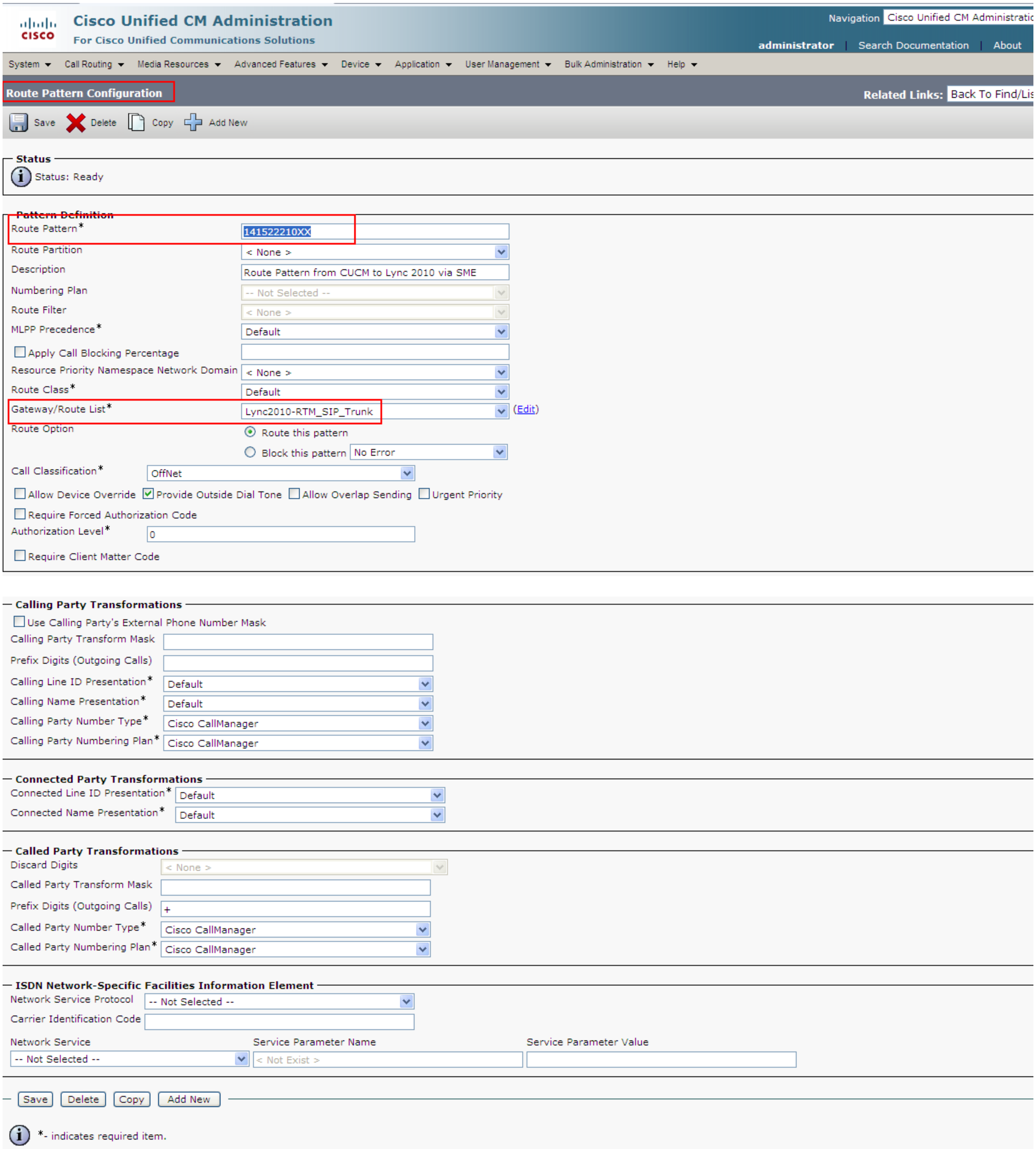
Route Pattern-101X



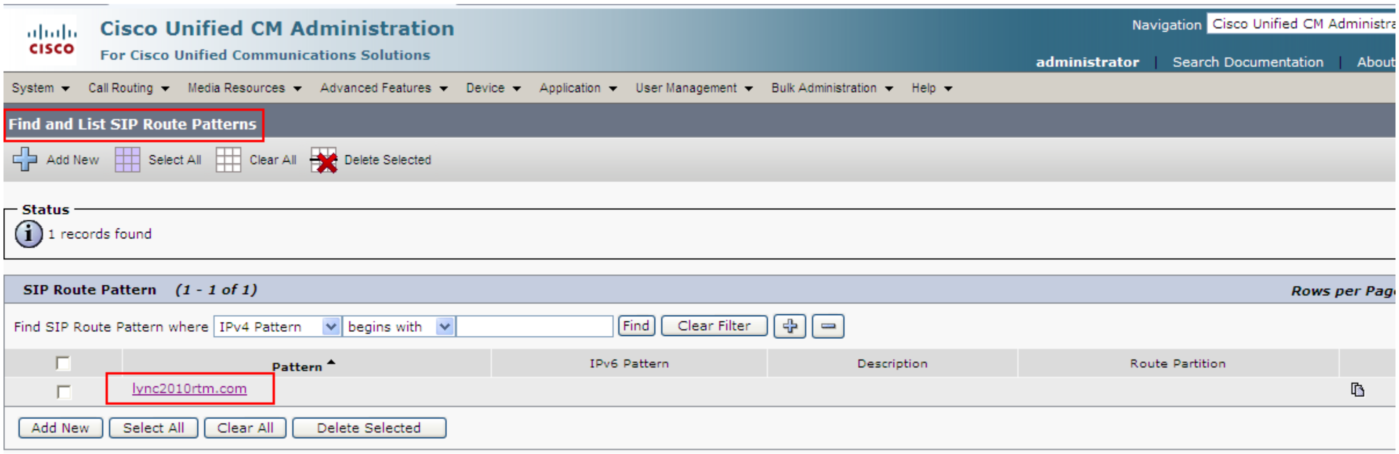
Route Pattern-1408XXXXXXX

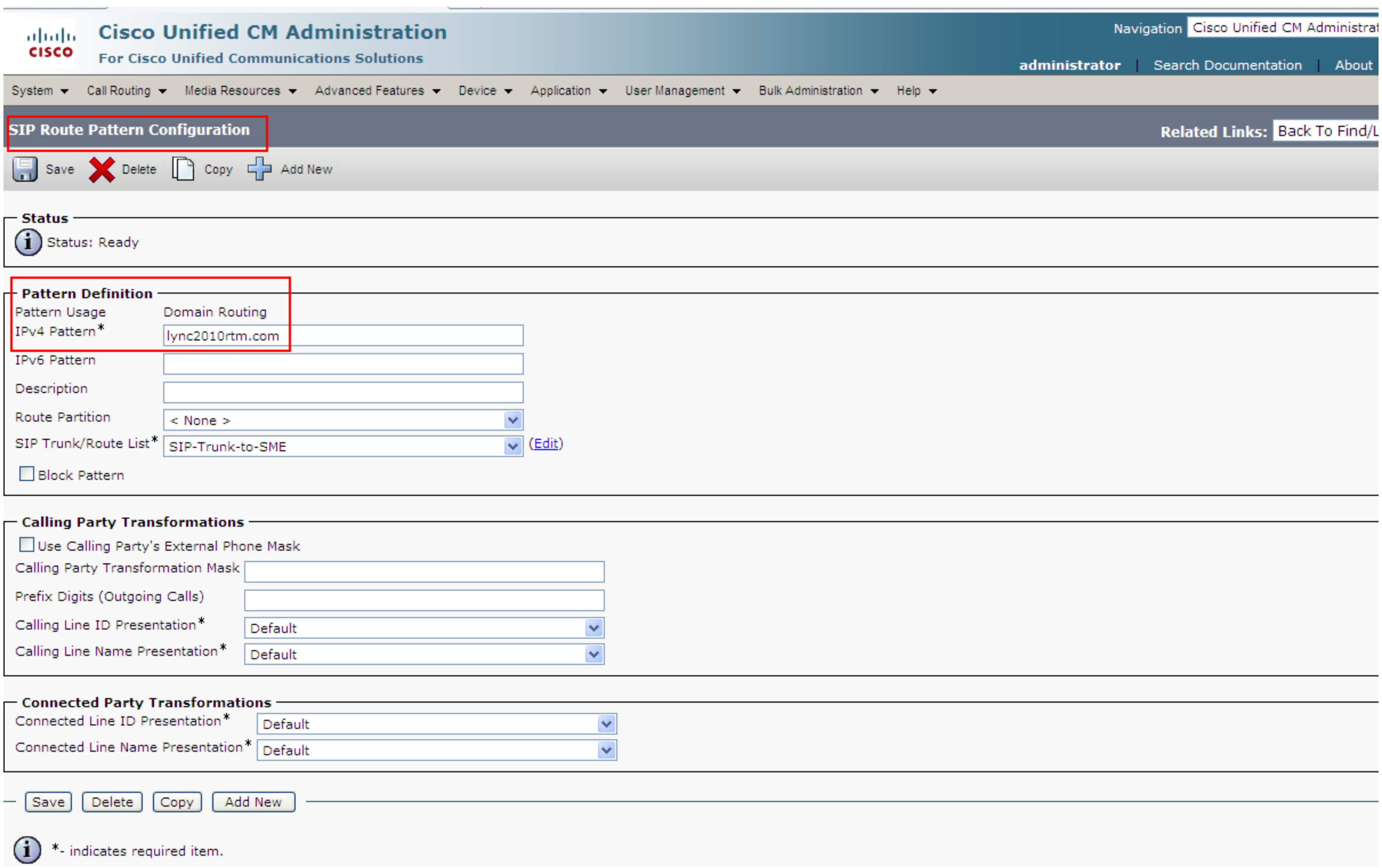


Route Pattern- 141522210XX



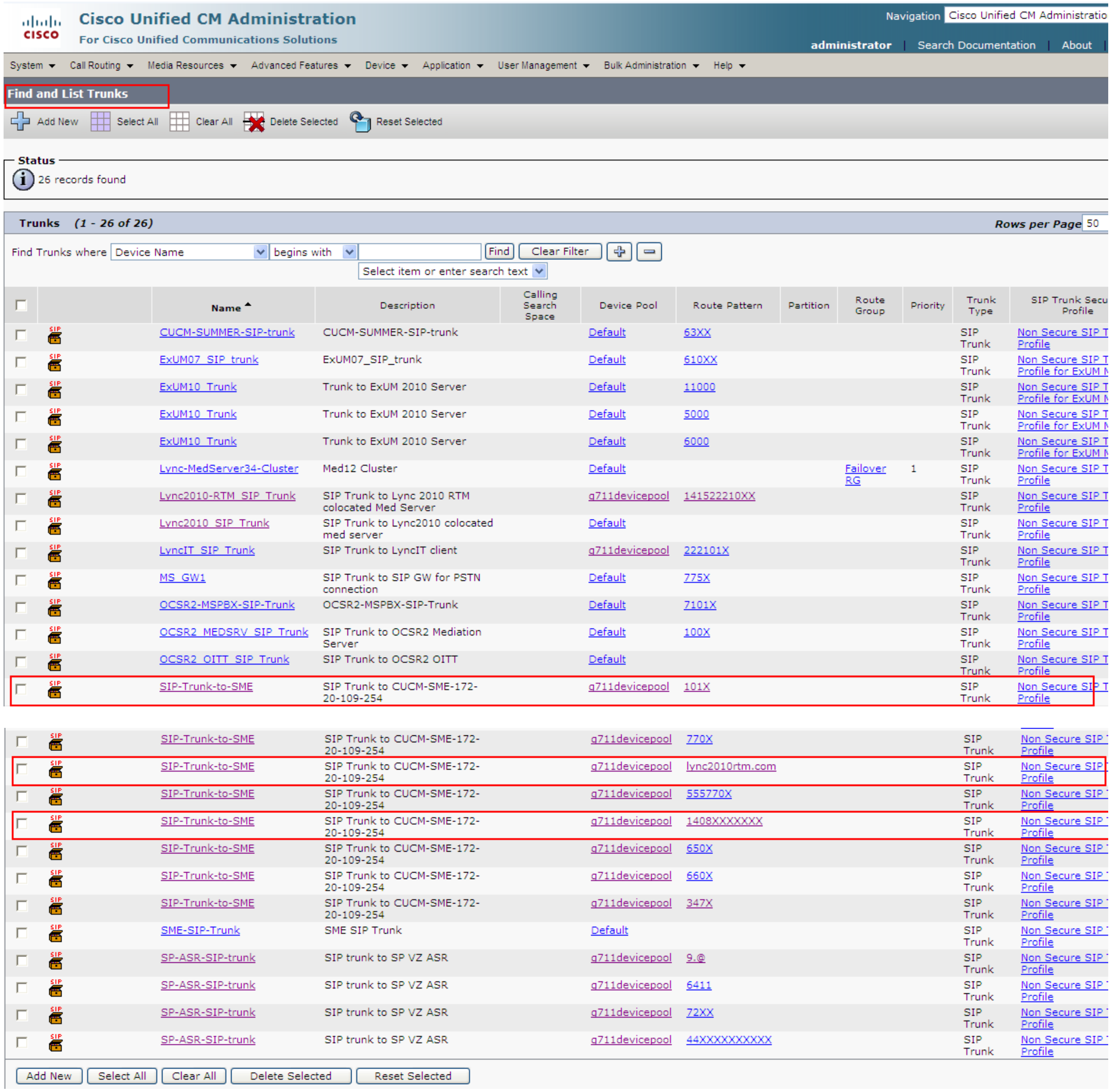
SIP Route Pattern*





*This configuration is needed for the calls initiated from CUCM using URI dialing. There is a SIP trunk to CUCM-SME using this route pattern.

Cisco Unified Communications Manager-SIP Trunk configuration


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administratio
administrator | Search Documentation | About

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Find and List Trunks

Add New | Select All | Clear All | Delete Selected | Reset Selected

Status

26 records found

Trunks (1 - 26 of 26) Rows per Page 50

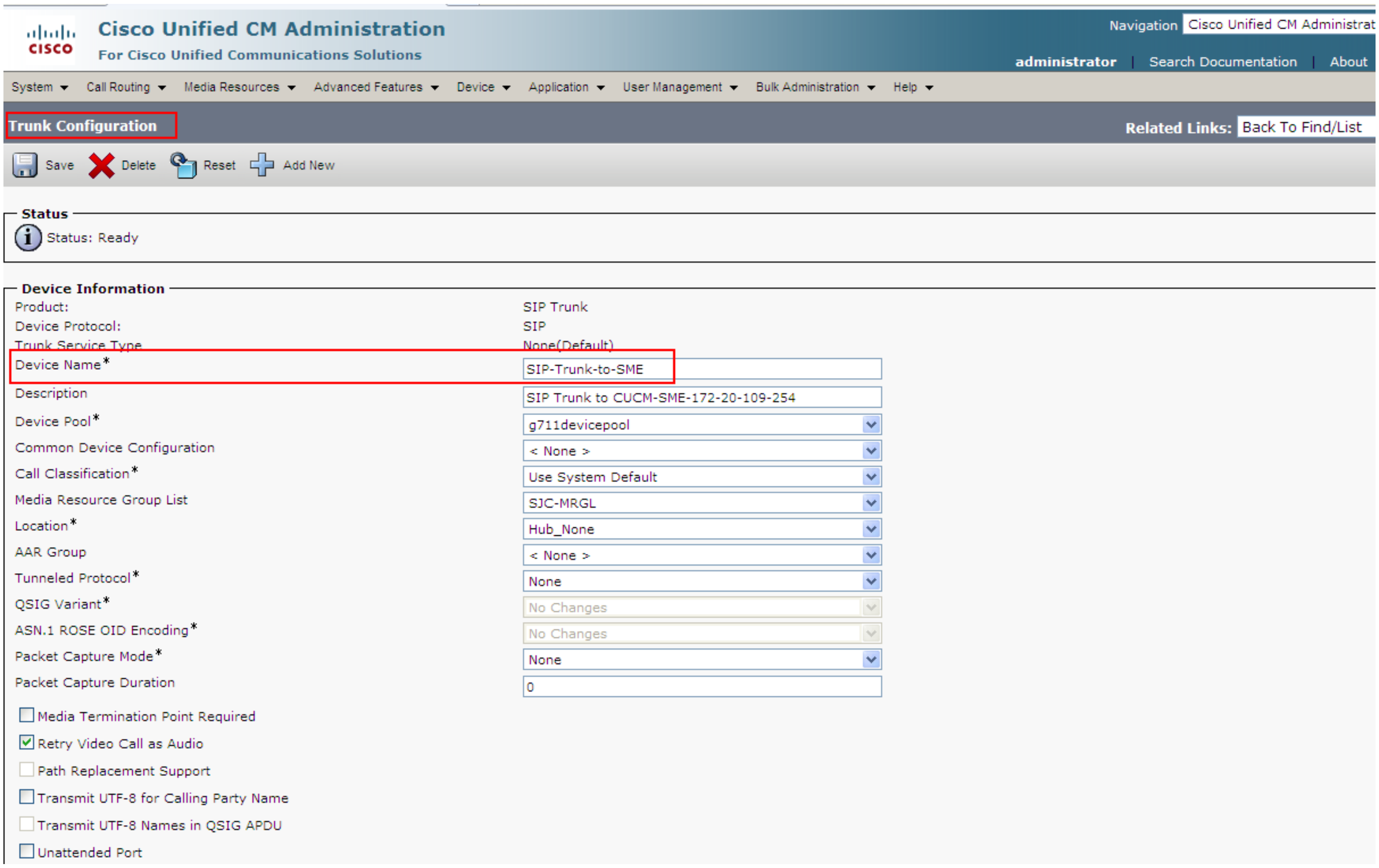
Find Trunks where Device Name begins with Find Clear Filter

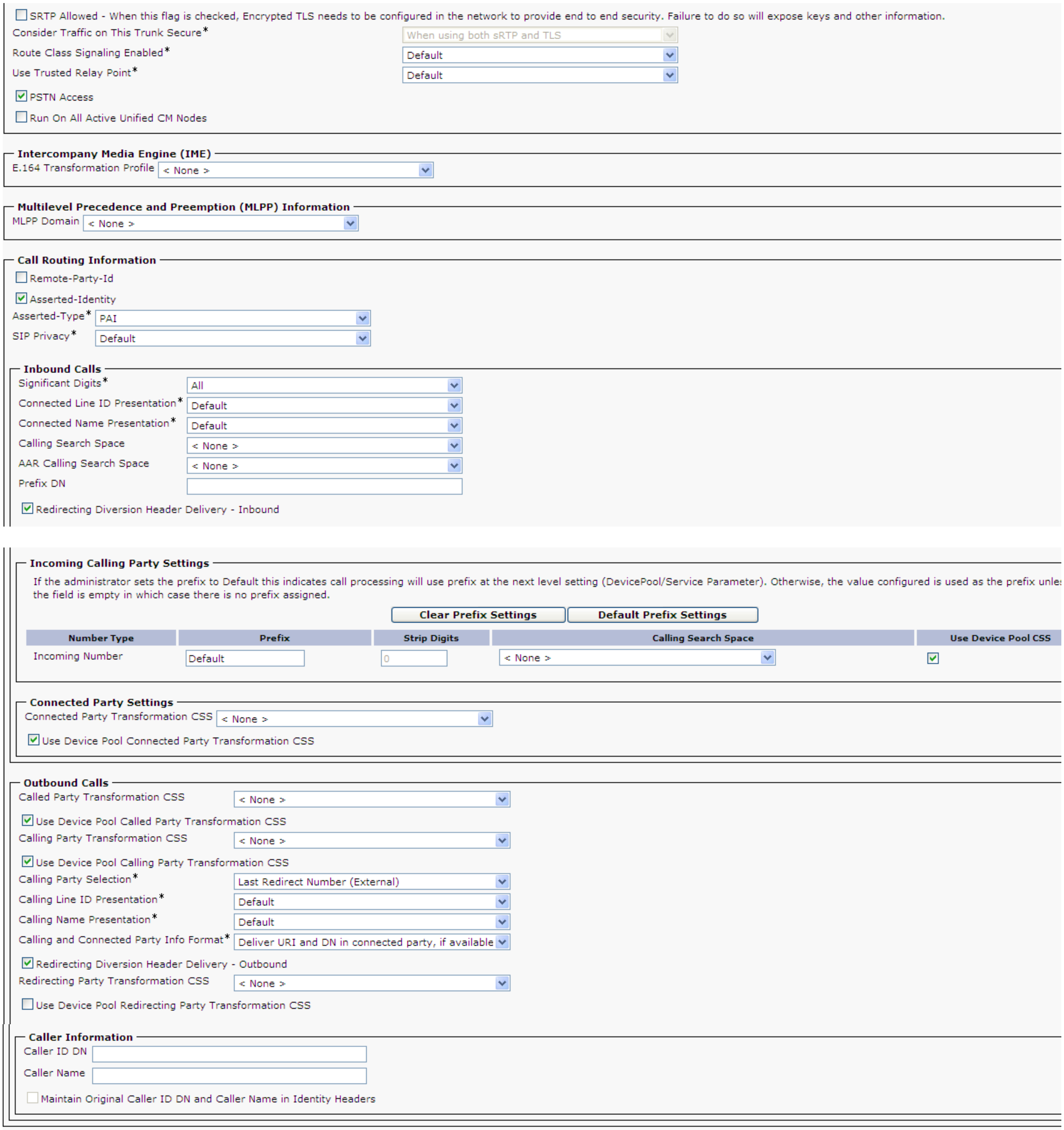
Select item or enter search text

Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Secu Profile
CUCM-SUMMER-SIP-trunk	CUCM-SUMMER-SIP-trunk		Default	63XX				SIP Trunk	Non Secure SIP T Profile
ExUM07_SIP_trunk	ExUM07_SIP_trunk		Default	610XX				SIP Trunk	Non Secure SIP T Profile for ExUM N
ExUM10_Trunk	Trunk to ExUM 2010 Server		Default	11000				SIP Trunk	Non Secure SIP T Profile for ExUM N
ExUM10_Trunk	Trunk to ExUM 2010 Server		Default	5000				SIP Trunk	Non Secure SIP T Profile for ExUM N
ExUM10_Trunk	Trunk to ExUM 2010 Server		Default	6000				SIP Trunk	Non Secure SIP T Profile for ExUM N
Lync-MedServer34-Cluster	Med12 Cluster		Default			Failover RG	1	SIP Trunk	Non Secure SIP T Profile
Lync2010-RTM_SIP_Trunk	SIP Trunk to Lync 2010 RTM colocated Med Server		g711devicepool	141522210XX				SIP Trunk	Non Secure SIP T Profile
Lync2010_SIP_Trunk	SIP Trunk to Lync2010 colocated med server		Default					SIP Trunk	Non Secure SIP T Profile
LyncIT_SIP_Trunk	SIP Trunk to LyncIT client		g711devicepool	222101X				SIP Trunk	Non Secure SIP T Profile
MS_GW1	SIP Trunk to SIP GW for PSTN connection		Default	775X				SIP Trunk	Non Secure SIP T Profile
OCSR2-MSPBX-SIP-Trunk	OCSR2-MSPBX-SIP-Trunk		Default	7101X				SIP Trunk	Non Secure SIP T Profile
OCSR2_MEDSRV_SIP_Trunk	SIP Trunk to OCSR2 Mediation Server		Default	100X				SIP Trunk	Non Secure SIP T Profile
OCSR2_OITT_SIP_Trunk	SIP Trunk to OCSR2 OITT		Default					SIP Trunk	Non Secure SIP T Profile
SIP-Trunk-to-SME	SIP Trunk to CUCM-SME-172-20-109-254		g711devicepool	101X				SIP Trunk	Non Secure SIP T Profile
SIP-Trunk-to-SME	SIP Trunk to CUCM-SME-172-20-109-254		g711devicepool	770X				SIP Trunk	Non Secure SIP Profile
SIP-Trunk-to-SME	SIP Trunk to CUCM-SME-172-20-109-254		g711devicepool	lync2010rtm.com				SIP Trunk	Non Secure SIP Profile
SIP-Trunk-to-SME	SIP Trunk to CUCM-SME-172-20-109-254		g711devicepool	555770X				SIP Trunk	Non Secure SIP Profile
SIP-Trunk-to-SME	SIP Trunk to CUCM-SME-172-20-109-254		g711devicepool	1408XXXXXXX				SIP Trunk	Non Secure SIP Profile
SIP-Trunk-to-SME	SIP Trunk to CUCM-SME-172-20-109-254		g711devicepool	650X				SIP Trunk	Non Secure SIP Profile
SIP-Trunk-to-SME	SIP Trunk to CUCM-SME-172-20-109-254		g711devicepool	660X				SIP Trunk	Non Secure SIP Profile
SIP-Trunk-to-SME	SIP Trunk to CUCM-SME-172-20-109-254		g711devicepool	347X				SIP Trunk	Non Secure SIP Profile
SME-SIP-Trunk	SME SIP Trunk		Default					SIP Trunk	Non Secure SIP Profile
SP-ASR-SIP-trunk	SIP trunk to SP VZ ASR		g711devicepool	9.@				SIP Trunk	Non Secure SIP Profile
SP-ASR-SIP-trunk	SIP trunk to SP VZ ASR		g711devicepool	6411				SIP Trunk	Non Secure SIP Profile
SP-ASR-SIP-trunk	SIP trunk to SP VZ ASR		g711devicepool	72XX				SIP Trunk	Non Secure SIP Profile
SP-ASR-SIP-trunk	SIP trunk to SP VZ ASR		g711devicepool	44XXXXXXXXXX				SIP Trunk	Non Secure SIP Profile

Add New | Select All | Clear All | Delete Selected | Reset Selected

SIP trunk to SME with Route pattern 101X





SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
Consider Traffic on This Trunk Secure*
When using both sRTP and TLS
Route Class Signaling Enabled*
Default
Use Trusted Relay Point*
Default
PSTN Access
Run On All Active Unified CM Nodes
Intercompany Media Engine (IME)
E.164 Transformation Profile
< None >
Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain
< None >
Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type*
PAI
SIP Privacy*
Default
Inbound Calls
Significant Digits*
All
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Calling Search Space
< None >
AAR Calling Search Space
< None >
Prefix DN
Redirecting Diversion Header Delivery - Inbound
Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unle
the field is empty in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >
Connected Party Settings
Connected Party Transformation CSS
< None >
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*
Last Redirect Number (External)
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling and Connected Party Info Format*
Deliver URI and DN in connected party, if available
Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS
< None >
Use Device Pool Redirecting Party Transformation CSS
Caller Information
Caller ID DN
Caller Name
Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	172.20.109.254		5060

MTP Preferred Originating Codec*	711ulaw
BLF Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile with FQDN in SIP Requests
DTMF Signaling Method*	RFC 2833

Normalization Script

Normalization Script: remove-tias

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Geolocation Configuration

Geolocation	< None >
Geolocation Filter	< None >

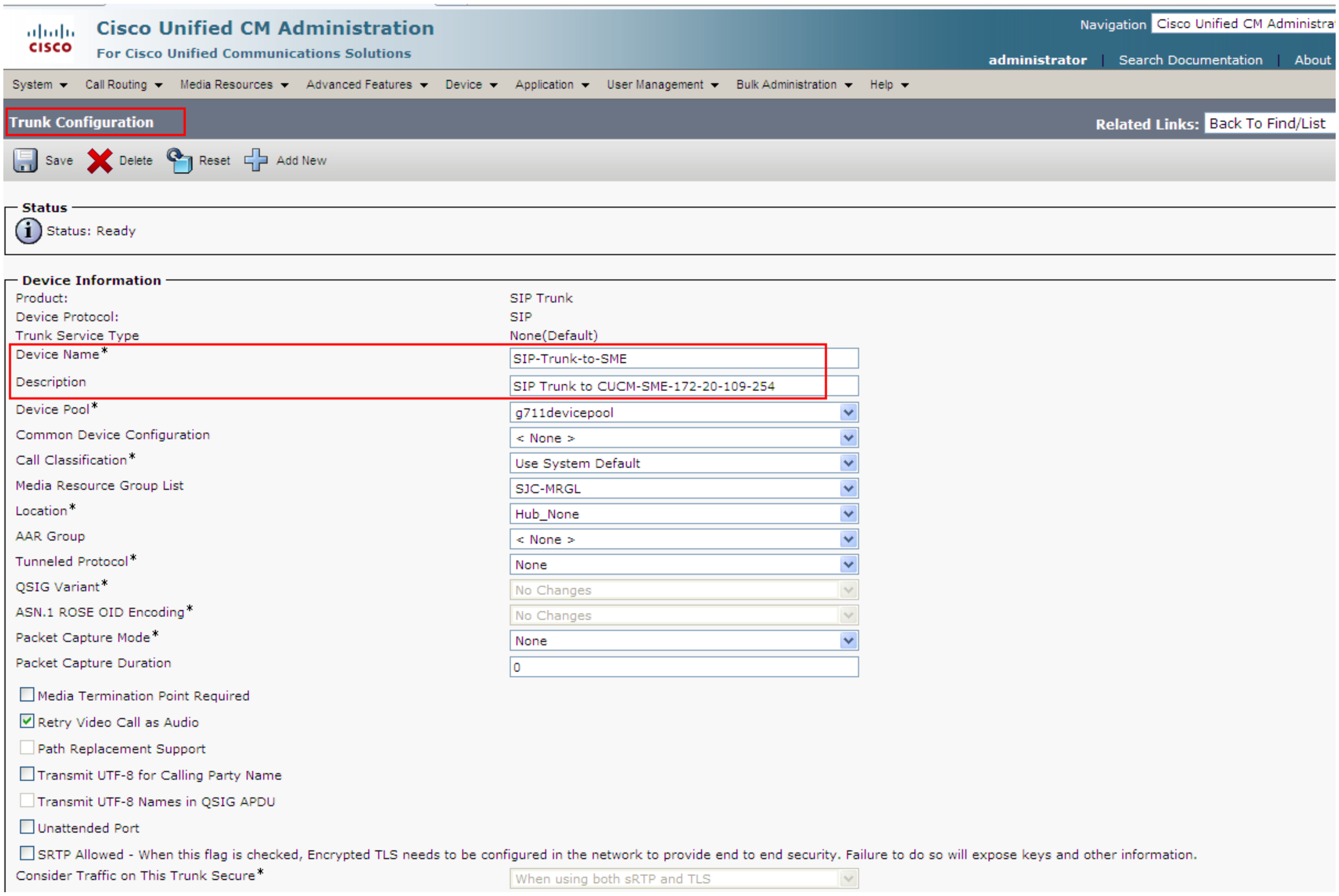
☐ Send Geolocation Information

Save | Delete | Reset | Add New

*- indicates required item.

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

SIP trunk to SME with Route Pattern lync2010rtm.com



Route Class Signaling Enabled* — Default

Use Trusted Relay Point* — Default

- [x] PSTN Access
- [ ] Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile — < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain — < None >

Call Routing Information

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

Asserted-Type* — PAI

SIP Privacy* — Default

Inbound Calls

Significant Digits* — All

Connected Line ID Presentation* — Default

Connected Name Presentation* — Default

Calling Search Space — < None >

AAR Calling Search Space — < None >

Prefix DN —

- [x] Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings | Default Prefix Settings

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

Connected Party Settings

Connected Party Transformation CSS — < None >

- [x] Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS — < None >

- [x] Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS — < None >

- [x] Use Device Pool Calling Party Transformation CSS

Calling Party Selection* — Last Redirect Number (External)

Calling Line ID Presentation* — Default

Calling Name Presentation* — Default

Calling and Connected Party Info Format* — Deliver URI and DN in connected party, if available

- [x] Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS — < None >

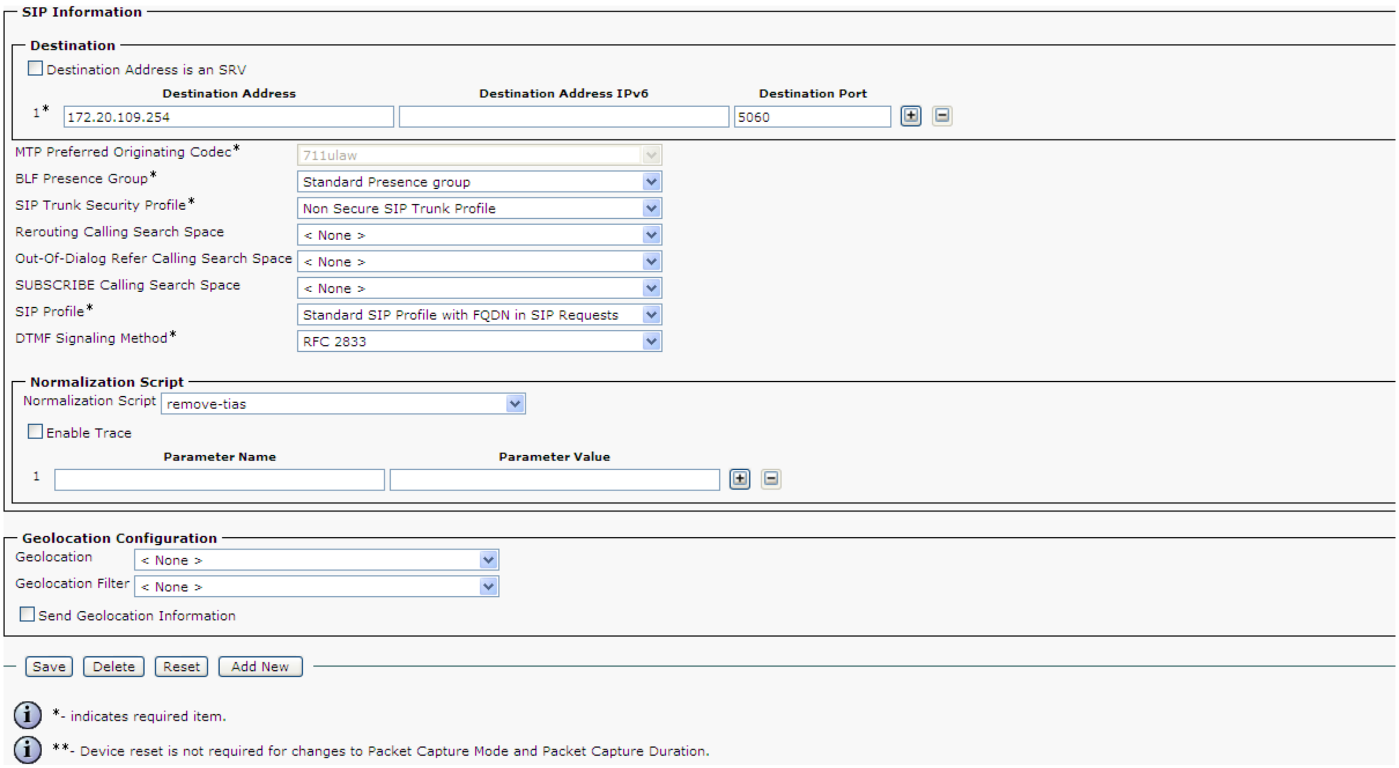
- [ ] Use Device Pool Redirecting Party Transformation CSS

Caller Information

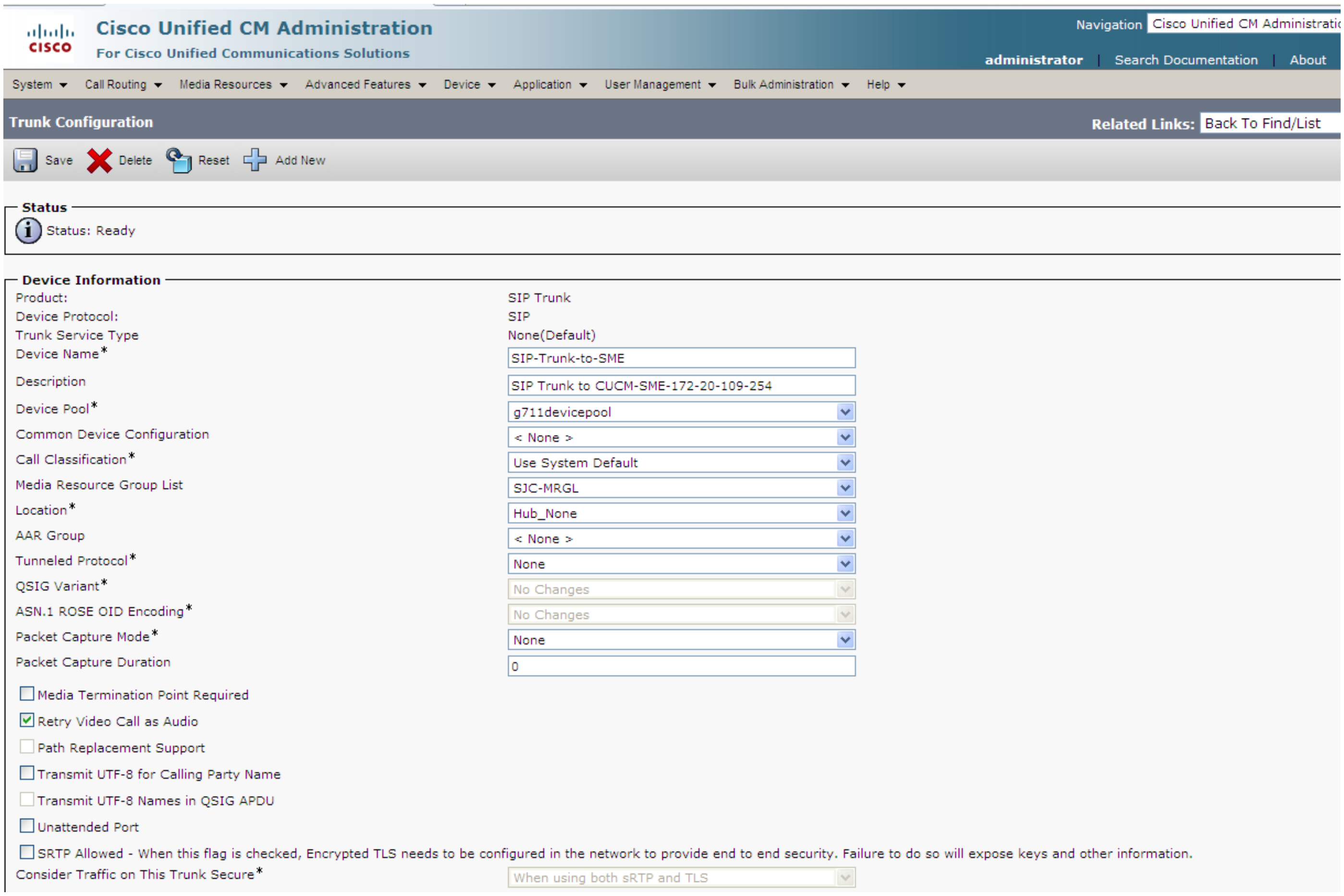
Caller ID DN —

Caller Name —

- [ ] Maintain Original Caller ID DN and Caller Name in Identity Headers


SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1*
172.20.109.254
5060
MTP Preferred Originating Codec*
711ulaw
BLF Presence Group*
Standard Presence group
SIP Trunk Security Profile*
Non Secure SIP Trunk Profile
Rerouting Calling Search Space
< None >
Out-Of-Dialog Refer Calling Search Space
< None >
SUBSCRIBE Calling Search Space
< None >
SIP Profile*
Standard SIP Profile with FQDN in SIP Requests
DTMF Signaling Method*
RFC 2833
Normalization Script
Normalization Script
remove-tias
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.

SIP trunk to SME with Route Pattern 1408XXXXXXX



Route Class Signaling Enabled* Default
Use Trusted Relay Point* Default
☑ PSTN Access
☐ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)
E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain < None >

Call Routing Information
☐ Remote-Party-Id
☑ Asserted-Identity
Asserted-Type* PAI
SIP Privacy* Default

Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN
☑ Redirecting Diversion Header Delivery - Inbound

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

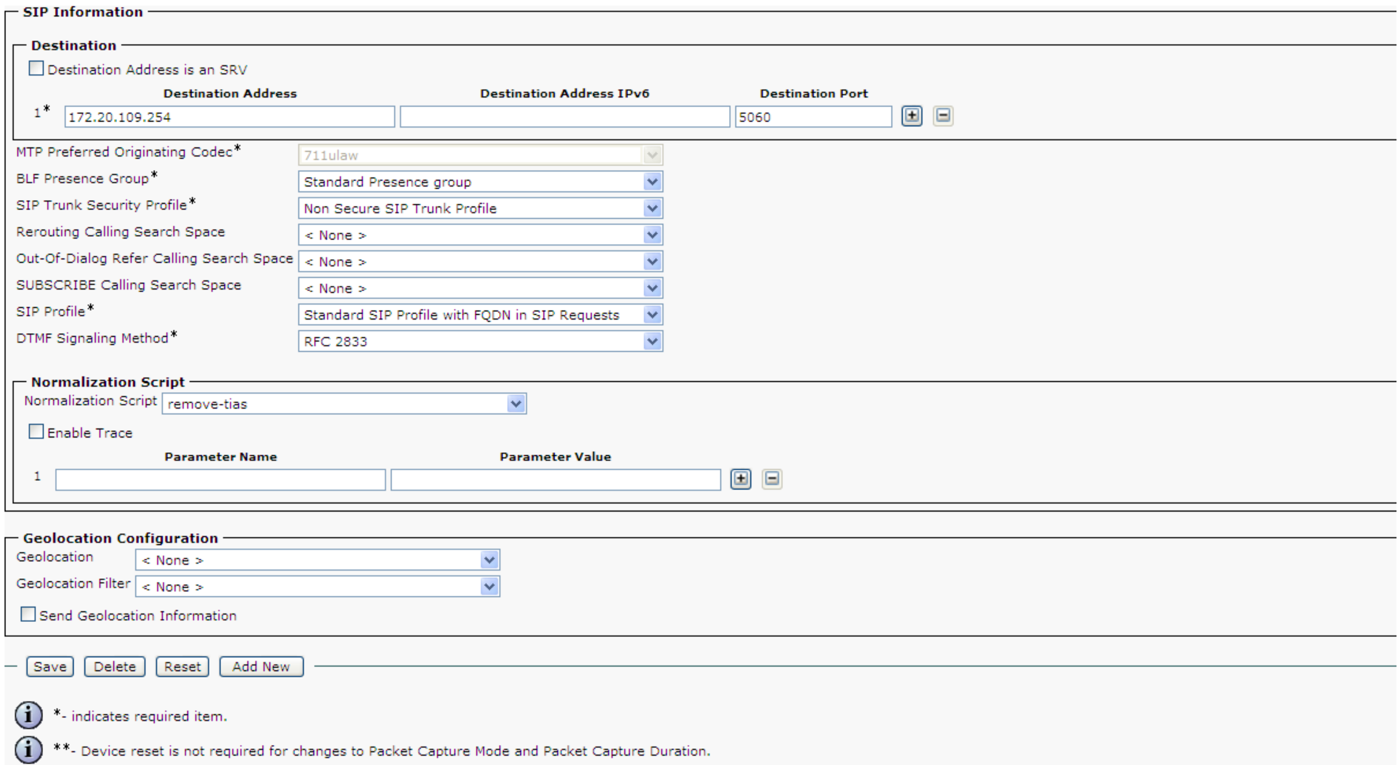
Clear Prefix Settings | Default Prefix Settings

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

Connected Party Settings
Connected Party Transformation CSS < None >
☑ 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* Last Redirect Number (External)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling and Connected Party Info Format* Deliver URI and DN in connected party, if available
☑ Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS < None >
☐ Use Device Pool Redirecting Party Transformation CSS

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



SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	172.20.109.254		5060

MTP Preferred Originating Codec*	711ulaw
BLF Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile with FQDN in SIP Requests
DTMF Signaling Method*	RFC 2833

Normalization Script

Normalization Script: remove-tias

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Geolocation Configuration

Geolocation	< None >
Geolocation Filter	< None >

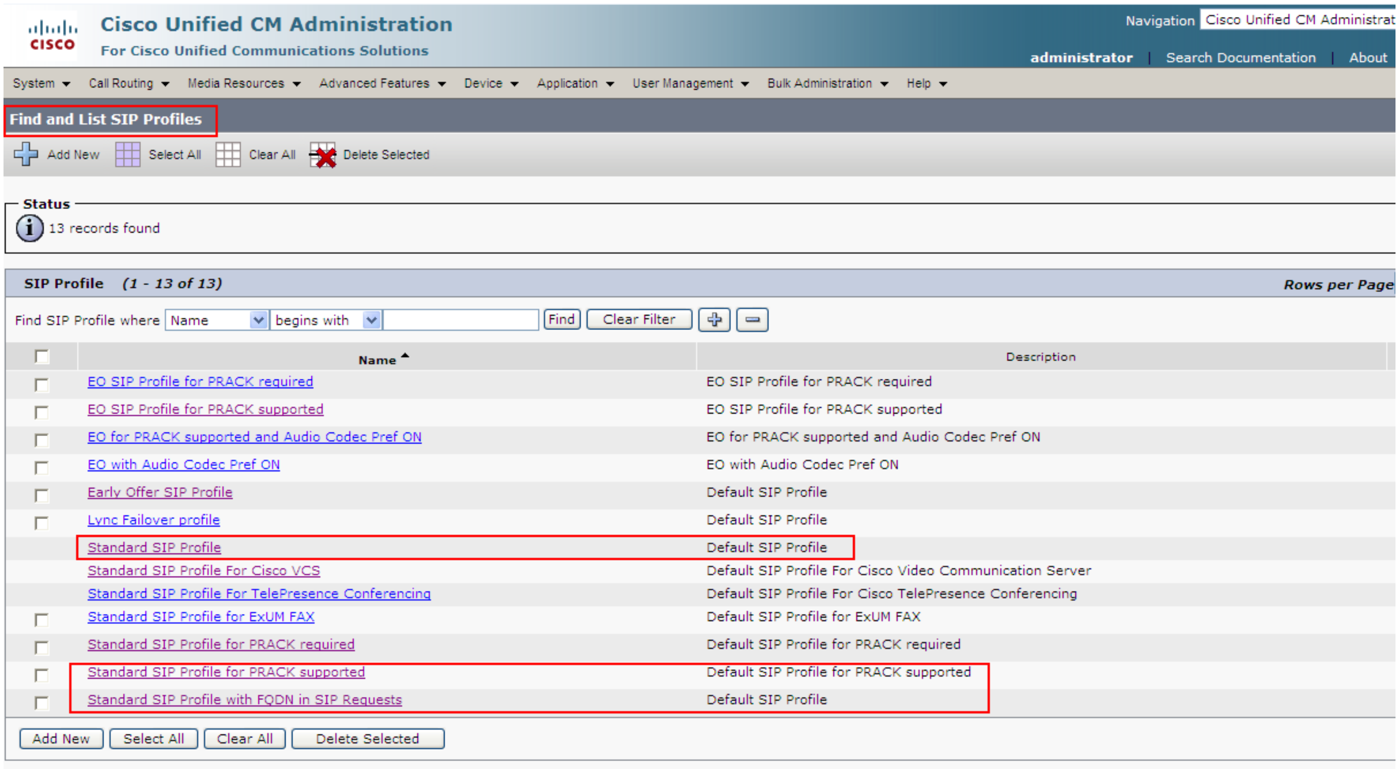
☐ Send Geolocation Information

Save | Delete | Reset | Add New

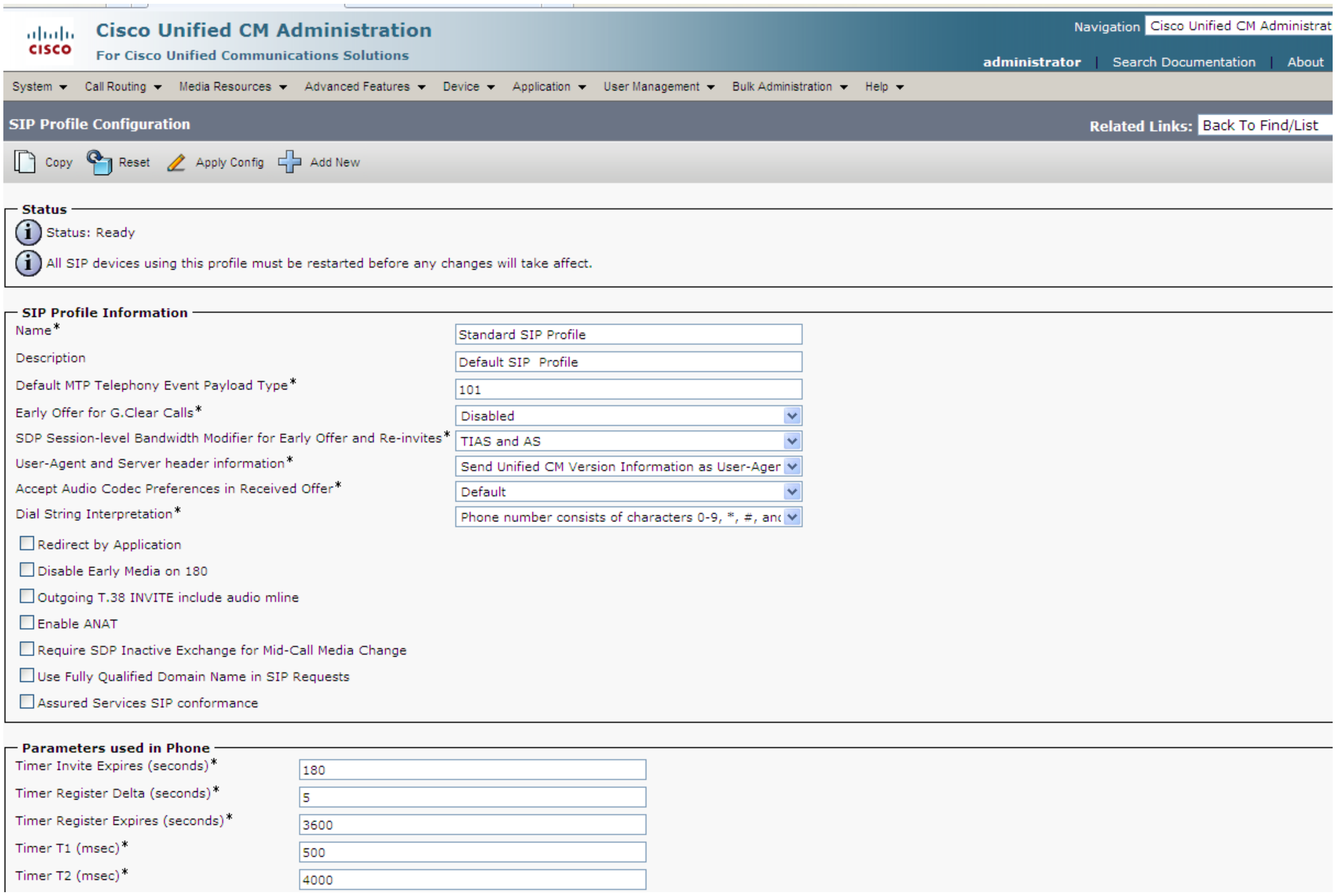
*- indicates required item.

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

Cisco Unified Communications Manager-SIP Profile Information



Standard SIP Profile



Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script: < None >

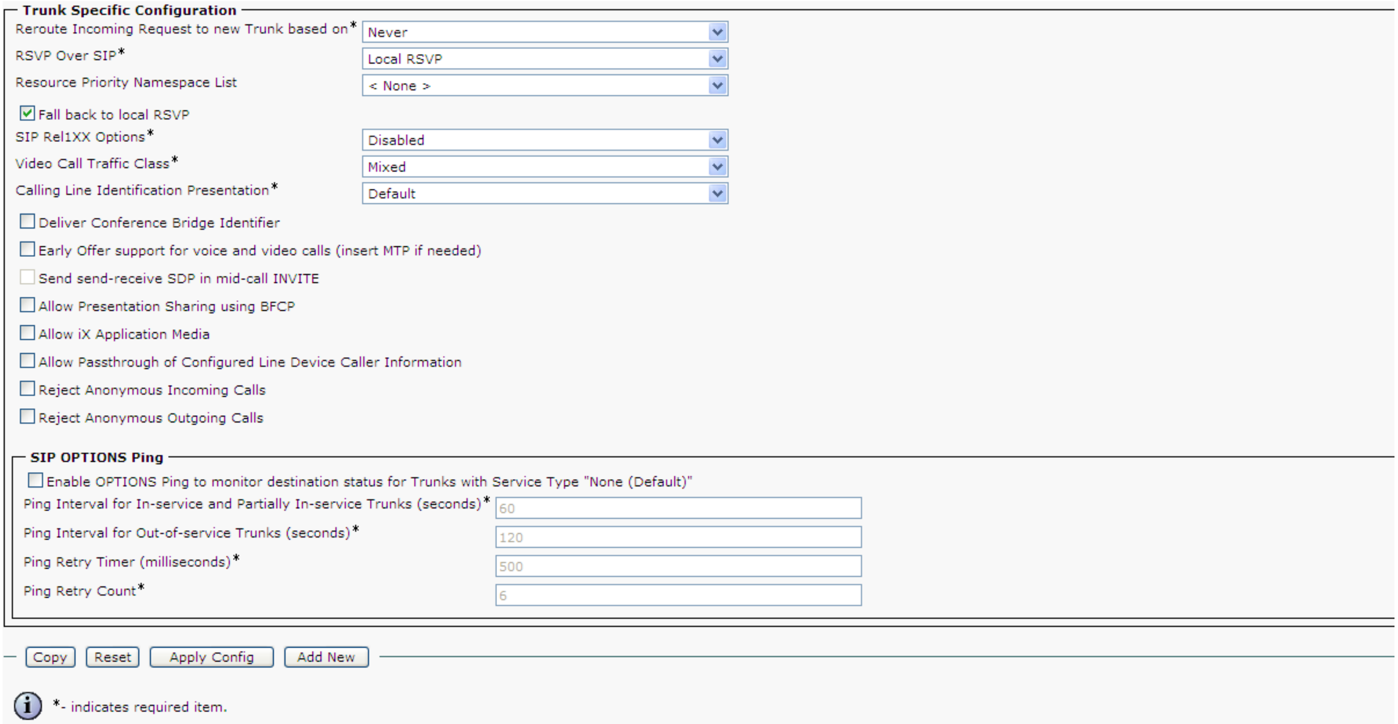
- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

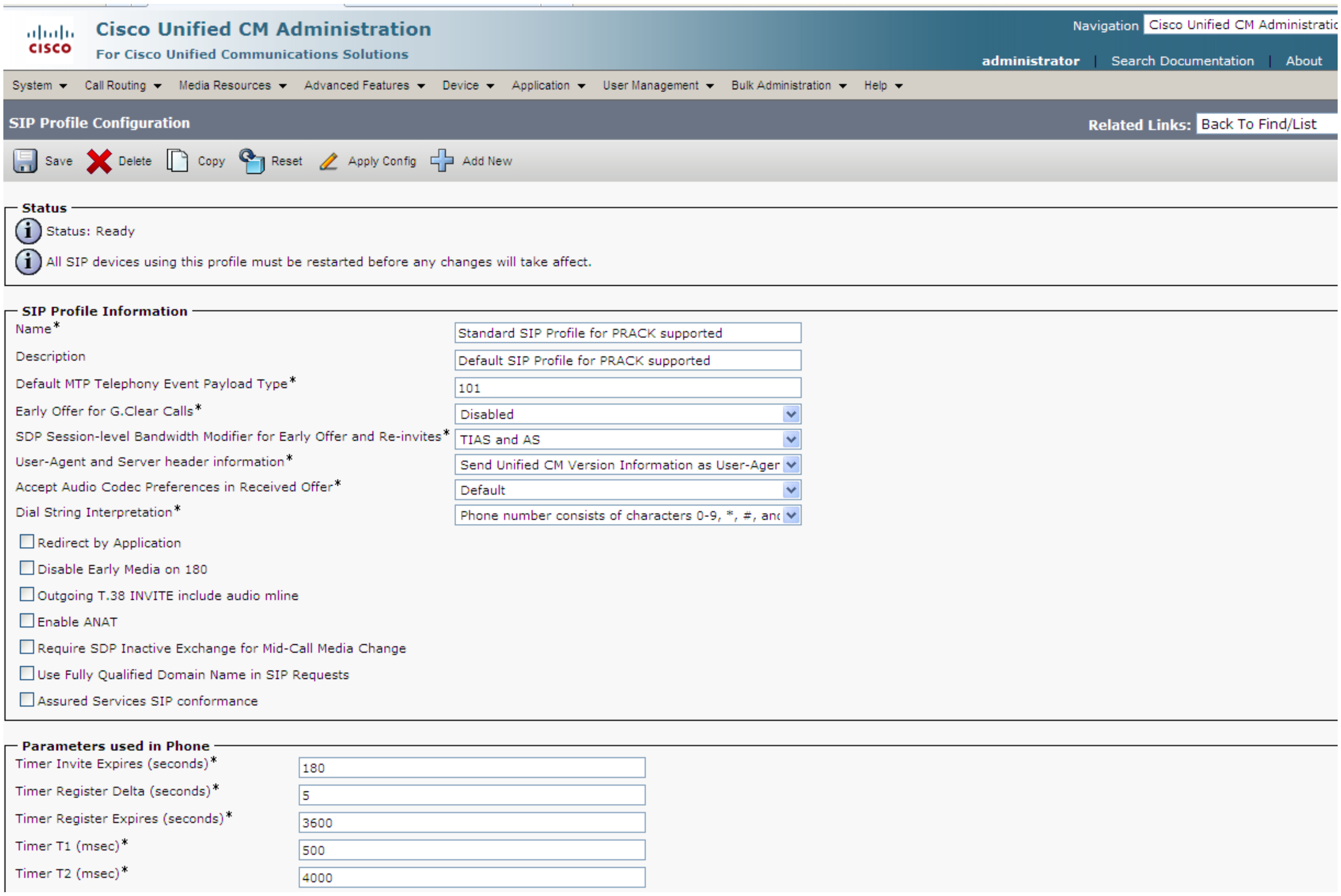
Incoming Requests FROM URI Settings

Caller ID DN

Caller Name


Trunk Specific Configuration
Reroute Incoming Request to new Trunk based on*
Never
RSVP Over SIP*
Local RSVP
Resource Priority Namespace List
< None >
Fall back to local RSVP
SIP Rel1XX Options*
Disabled
Video Call Traffic Class*
Mixed
Calling Line Identification Presentation*
Default
Deliver Conference Bridge Identifier
Early Offer support for voice and video calls (insert MTP if needed)
Send send-receive SDP in mid-call INVITE
Allow Presentation Sharing using BFCP
Allow iX Application Media
Allow Passthrough of Configured Line Device Caller Information
Reject Anonymous Incoming Calls
Reject Anonymous Outgoing Calls
SIP OPTIONS Ping
Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"
Ping Interval for In-service and Partially In-service Trunks (seconds)*
60
Ping Interval for Out-of-service Trunks (seconds)*
120
Ping Retry Timer (milliseconds)*
500
Ping Retry Count*
6
Copy
Reset
Apply Config
Add New
*- indicates required item.

Standard SIP Profile for PRACK supported



Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script < None >

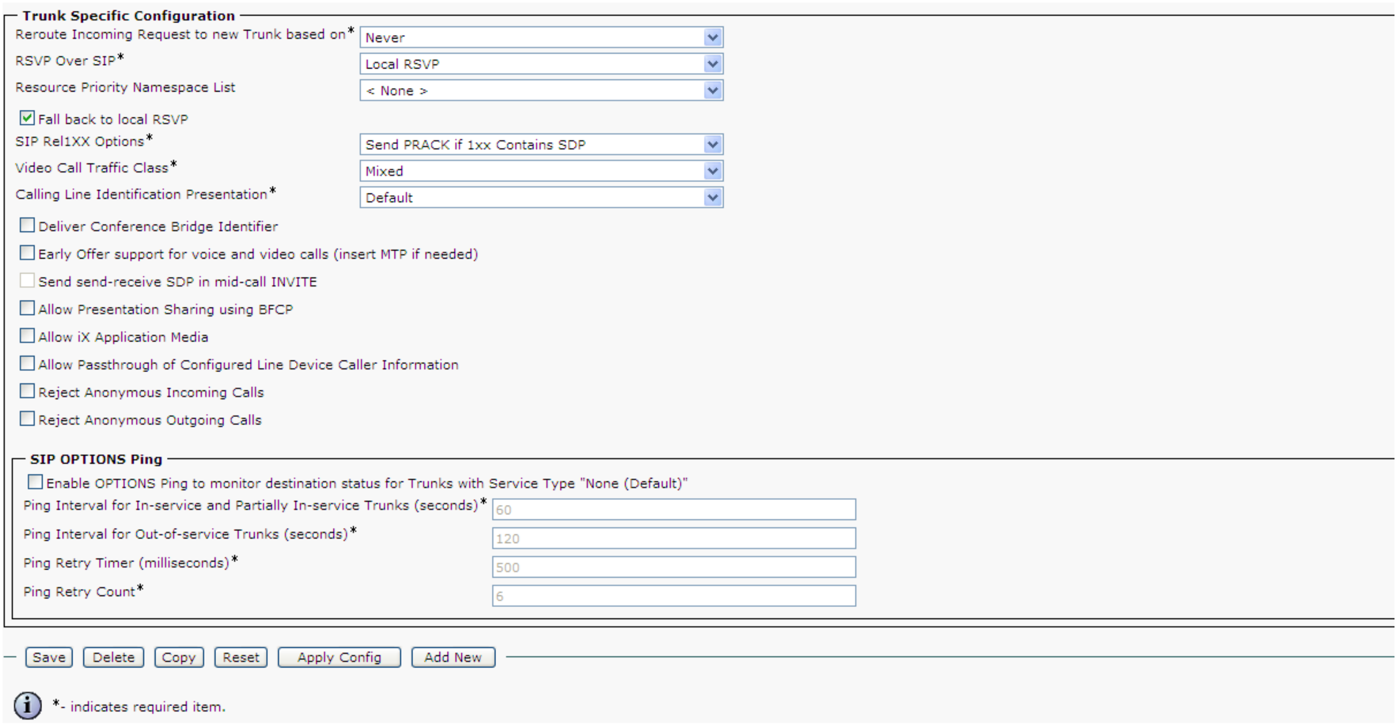
- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

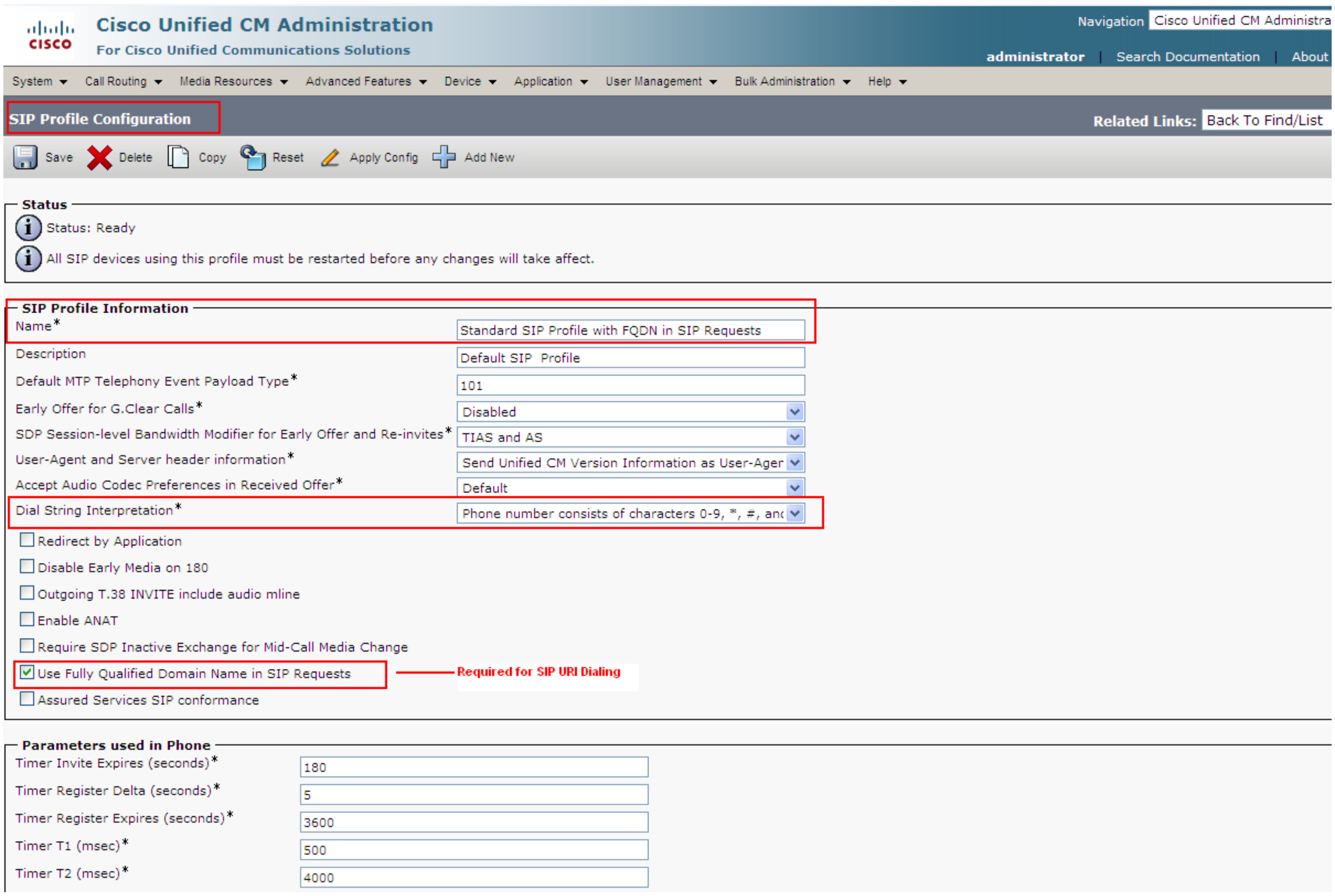
Incoming Requests FROM URI Settings

Caller ID DN

Caller Name


Trunk Specific Configuration
Reroute Incoming Request to new Trunk based on*
Never
RSVP Over SIP*
Local RSVP
Resource Priority Namespace List
< None >
Fall back to local RSVP
SIP Rel1XX Options*
Send PRACK if 1xx Contains SDP
Video Call Traffic Class*
Mixed
Calling Line Identification Presentation*
Default
Deliver Conference Bridge Identifier
Early Offer support for voice and video calls (insert MTP if needed)
Send send-receive SDP in mid-call INVITE
Allow Presentation Sharing using BFCP
Allow iX Application Media
Allow Passthrough of Configured Line Device Caller Information
Reject Anonymous Incoming Calls
Reject Anonymous Outgoing Calls
SIP OPTIONS Ping
Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"
Ping Interval for In-service and Partially In-service Trunks (seconds)*
60
Ping Interval for Out-of-service Trunks (seconds)*
120
Ping Retry Timer (milliseconds)*
500
Ping Retry Count*
6
Save
Delete
Copy
Reset
Apply Config
Add New
*- indicates required item.

Standard SIP Profile with FQDN in SIP requests



Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script: < None >

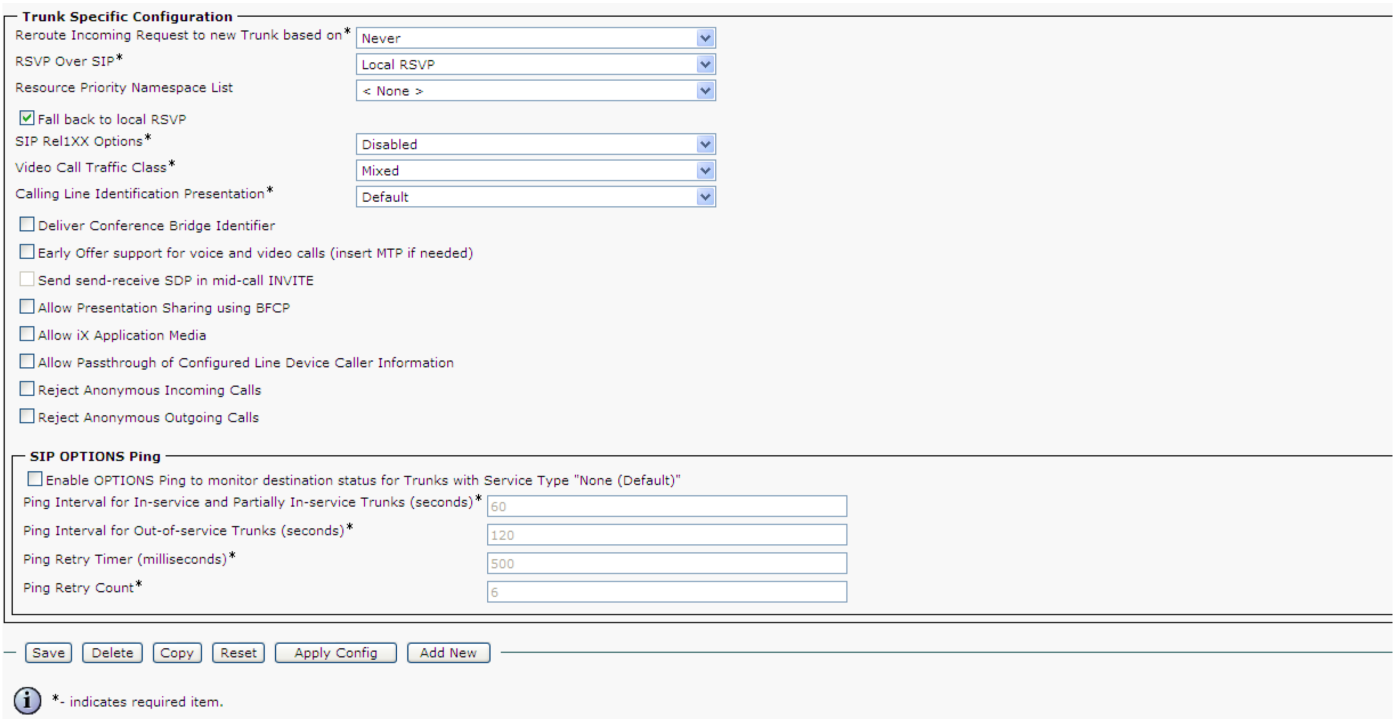
- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

Caller ID DN:

Caller Name:



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on* Never

RSVP Over SIP* Local RSVP

Resource Priority Namespace List < None >

☑ Fall back to local RSVP

SIP Rel1XX Options* Disabled

Video Call Traffic Class* Mixed

Calling Line Identification Presentation* Default

☐ Deliver Conference Bridge Identifier

☐ Early Offer support for voice and video calls (insert MTP if needed)

☐ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

SIP OPTIONS Ping

☐ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)* 60

Ping Interval for Out-of-service Trunks (seconds)* 120

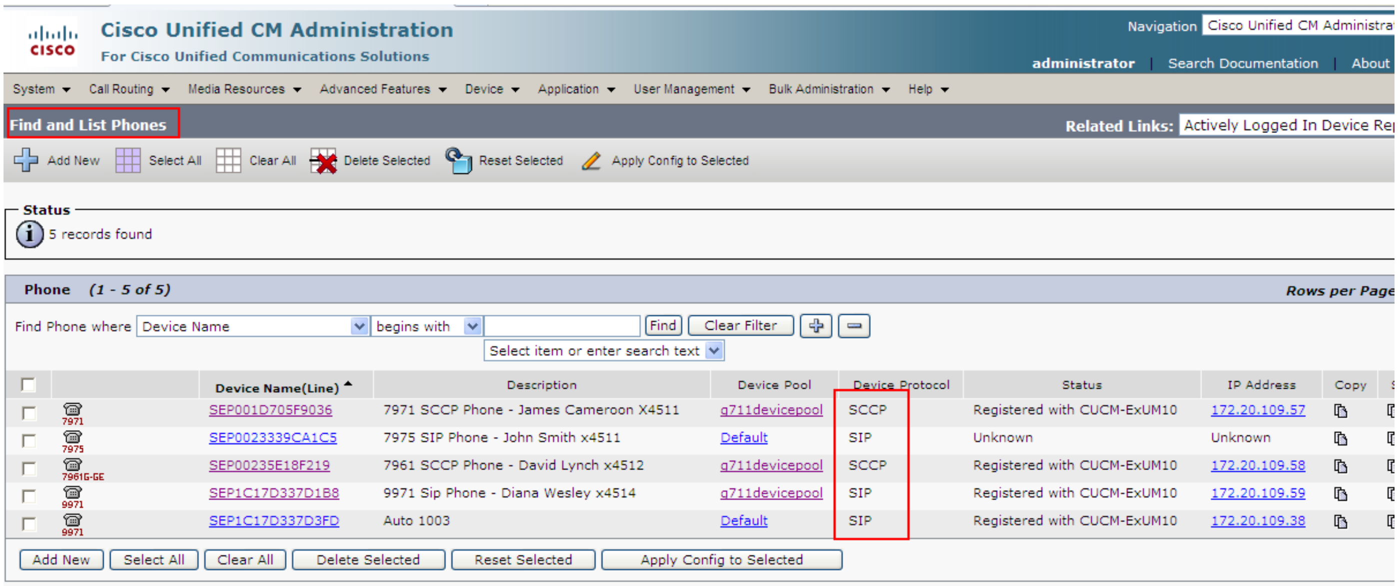
Ping Retry Timer (milliseconds)* 500

Ping Retry Count* 6

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

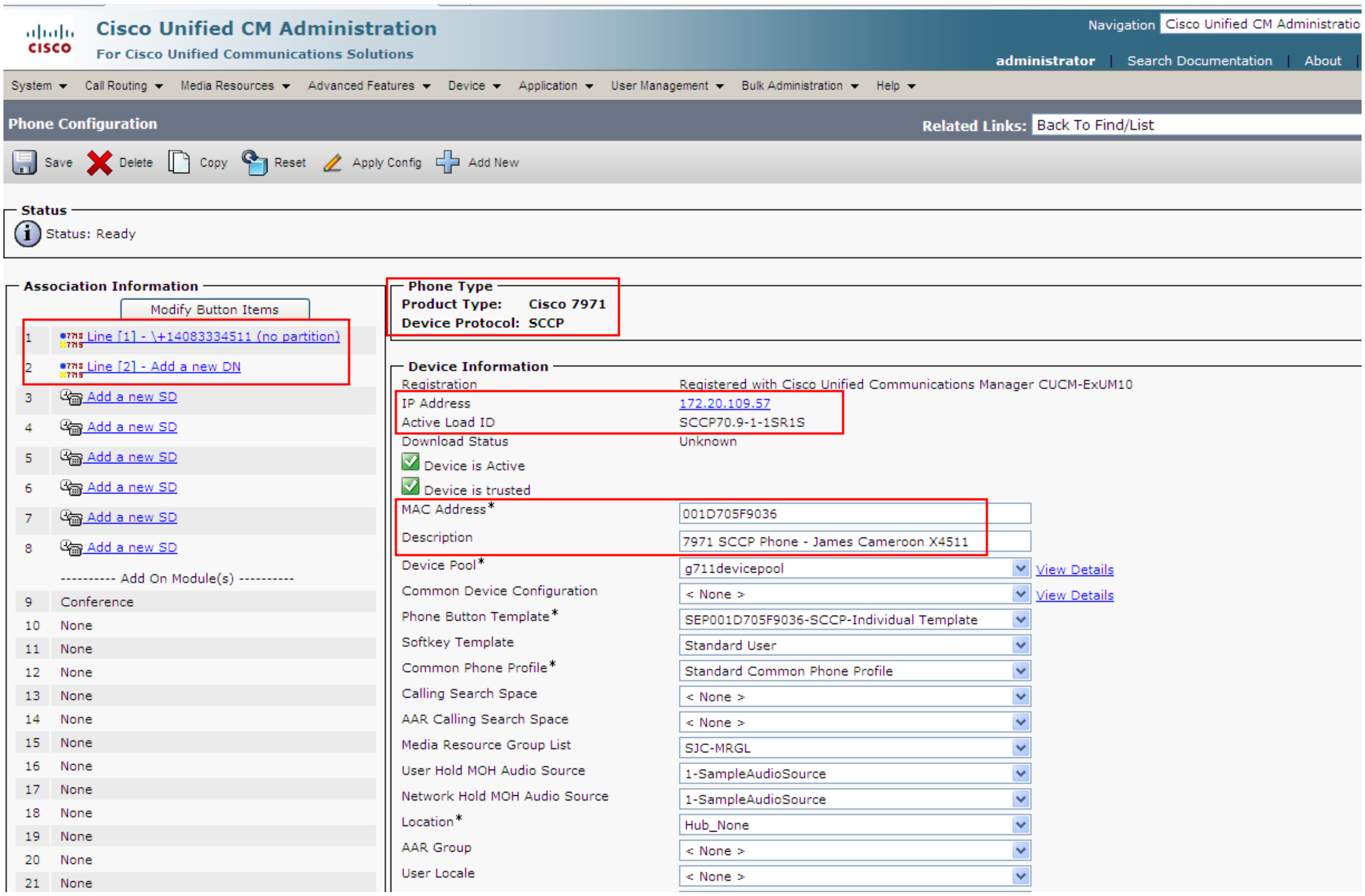
*- indicates required item.

SCCP/SIP Phone configurations on the Cisco Unified Call Manager



Device Name: SEP001D705F9036

Device protocol: SCCP



22	None
23	None
24	None
25	None
26	None
27	None
28	None
29	None
30	None
31	None
32	None
33	None
34	None
35	None
36	None
	---------- Unassigned Associated Items ----------
37	Add a new SD
38	Add a new SURL
39	Add a new BLF SD
40	Add a new BLF Directed Call Park
41	CallBack
42	Call Park
43	Call Pickup
44	Conference List
45	Do Not Disturb
46	End Call

Network Locale	< None >
Built In Bridge*	Default
Privacy*	Default
Device Mobility Mode*	Default — View Current Device Mobility Settings
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
Geolocation	< None >

- [x] Retry Video Call as Audio
- [ ] Ignore Presentation Indicators (internal calls only)
- [x] Allow Control of Device from CTI
- [x] Logged Into Hunt Group
- [ ] Remote Device
- [ ] Protected Device****
- [ ] Hot line Device*****
- [ ] Require off-premise location

47	Forward All
48	Group Call Pickup
49	Hold
50	Hunt Group Logout
51	Intercom [1] - Add a new Intercom
52	Malicious Call Identification
53	Meet Me Conference
54	Mobility
55	New Call
56	Other Pickup
57	Quality Reporting Tool
58	Redial
59	Remove Last Participant
60	Transfer
61	Video Mode
62	Queue Status
63	Privacy
64	None

Call Routing Information

Inbound Calls

Calling Party Transformation CSS: < None >

- [x] Use Device Pool Calling Party Transformation CSS

Outbound Calls

Calling Party Transformation CSS: < None >

- [x] Use Device Pool Calling Party Transformation CSS

Protocol Specific Information

Packet Capture Mode*	None
Packet Capture Duration	0
BLF Presence Group*	Standard Presence group
Device Security Profile*	Cisco 7971 - Standard SCCP Non-Secure Profile
SUBSCRIBE Calling Search Space	< None >

- [ ] Unattended Port
- [ ] Require DTMF Reception
- [ ] RFC2833 Disabled

Certification Authority Proxy Function (CAPF) Information

Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
Generate String	
Key Size (Bits)*	1024
Operation Completes By	2012 11 15 12 (YYYY:MM:DD:HH)

Certificate Operation Status: None

Note: Security Profile Contains Addition CAPF Settings.

Expansion Module Information

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

External Data Locations Information (Leave blank to use default)

Information	
Directory	
Messages	
Services	
Authentication Server	
Proxy Server	
Idle	
Idle Timer (seconds)	
Secure Authentication URL	
Secure Directory URL	
Secure Idle URL	
Secure Information URL	
Secure Messages URL	
Secure Services URL	

Extension Information

☐ Enable Extension Mobility

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

MLPP 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

☐ Disable Speakerphone

☐ Disable Speakerphone and Headset

	Param	Override Comm Settings
Forwarding Delay*	Disabled	
PC Port *	Enabled	
Settings Access*	Enabled	☐
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Video Capabilities*	Disabled	☐
Auto Line Select*	Disabled	
Web Access*	Disabled	☐
Days Display Not Active	Sunday Monday Tuesday	☐
Display On Time	07:30	☐

Display On Duration	10:30
Display Idle Timeout	01:00
Enable Power Save Plus	Sunday Monday Tuesday
Phone On Time	00:00
Phone Off Time	24:00
Phone Off Idle Timeout*	60
Enable Audible Alert	
EnergyWise Domain	
EnergyWise Endpoint Security Secret	
Allow EnergyWise Overrides	
Span to PC Port*	Disabled
Logging Display*	PC Controlled
Load Server	
Recording Tone*	Disabled
Recording Tone Local Volume*	100
Recording Tone Remote Volume*	50
Recording Tone Duration	
Display On When Incoming Call*	Disabled
RTCP*	Disabled
"more" Soft Key Timer	5
Auto Call Select*	Enabled
Log Server	
Advertise G.722 Codec*	Use System Default
Wideband Headset UI Control*	Enabled
Wideband 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
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	☐
Automatic Port Synchronization*	Disabled	☐
SSH Access*	Disabled	☐
LOGIN Access*	Enabled	☐
FIPS Mode*	Disabled	☐
80-bit SRTCP*	Disabled	☐

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

*- indicates required item.

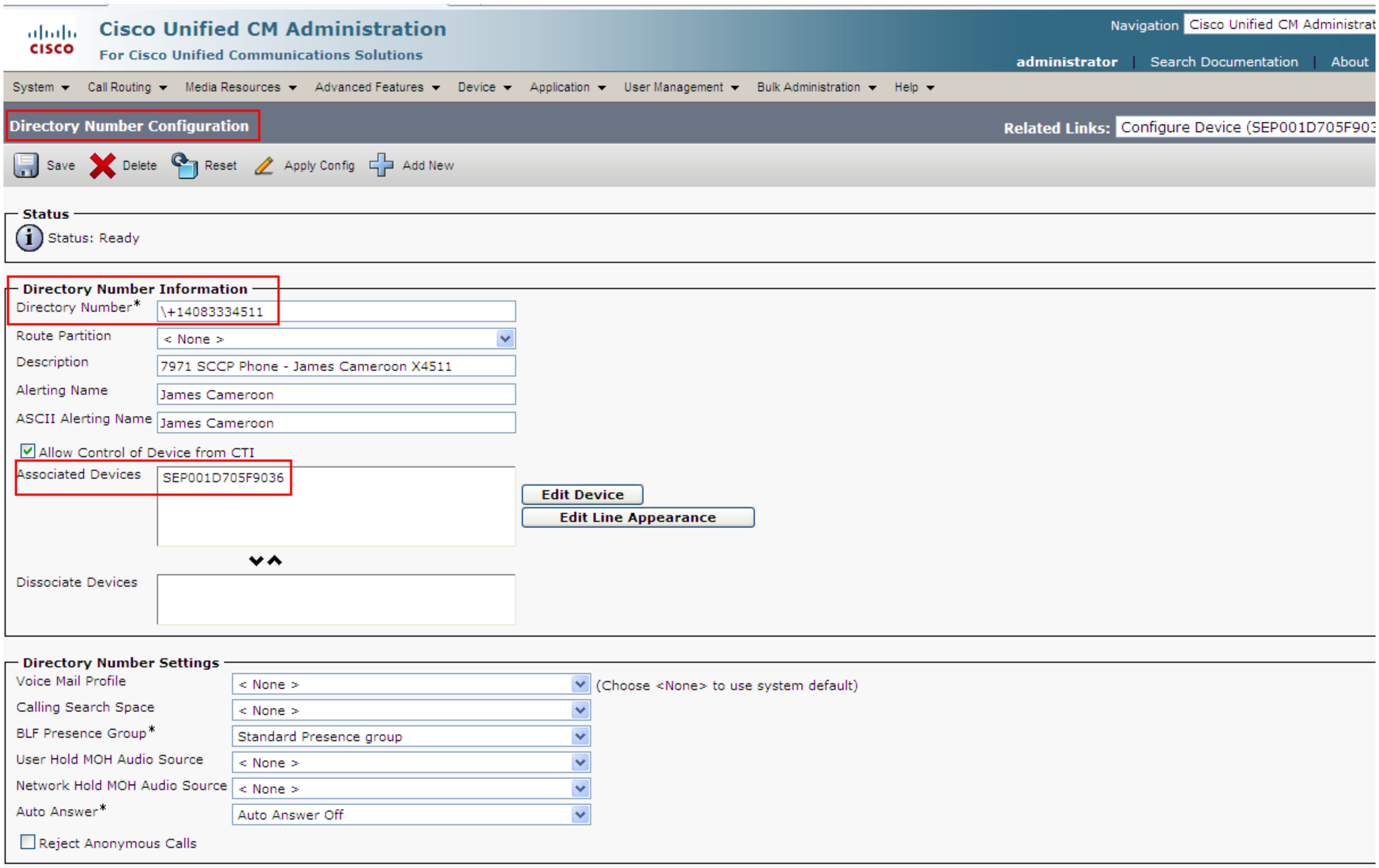
**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

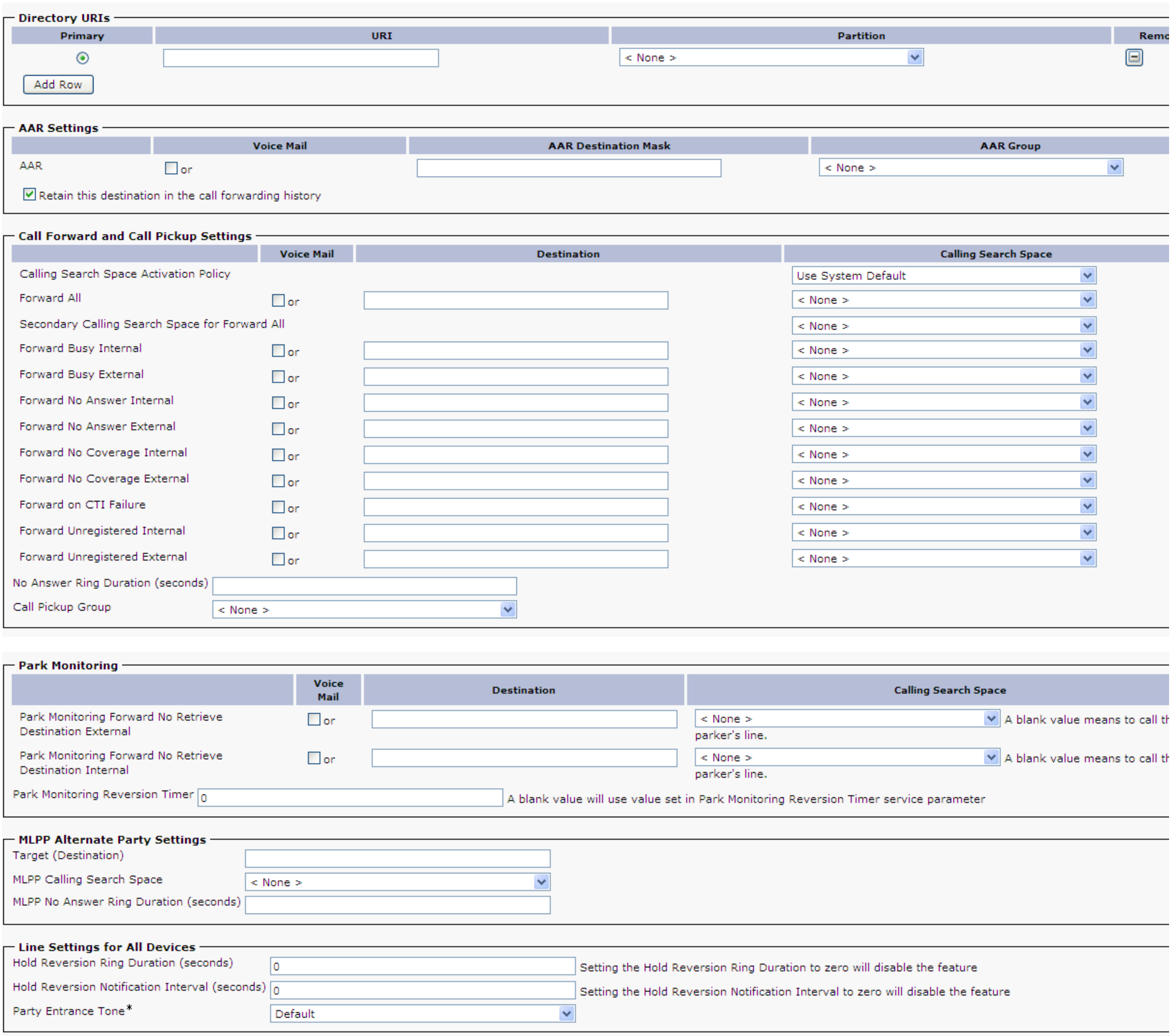
***Note: Security Profile Contains Addition CAPF Settings.

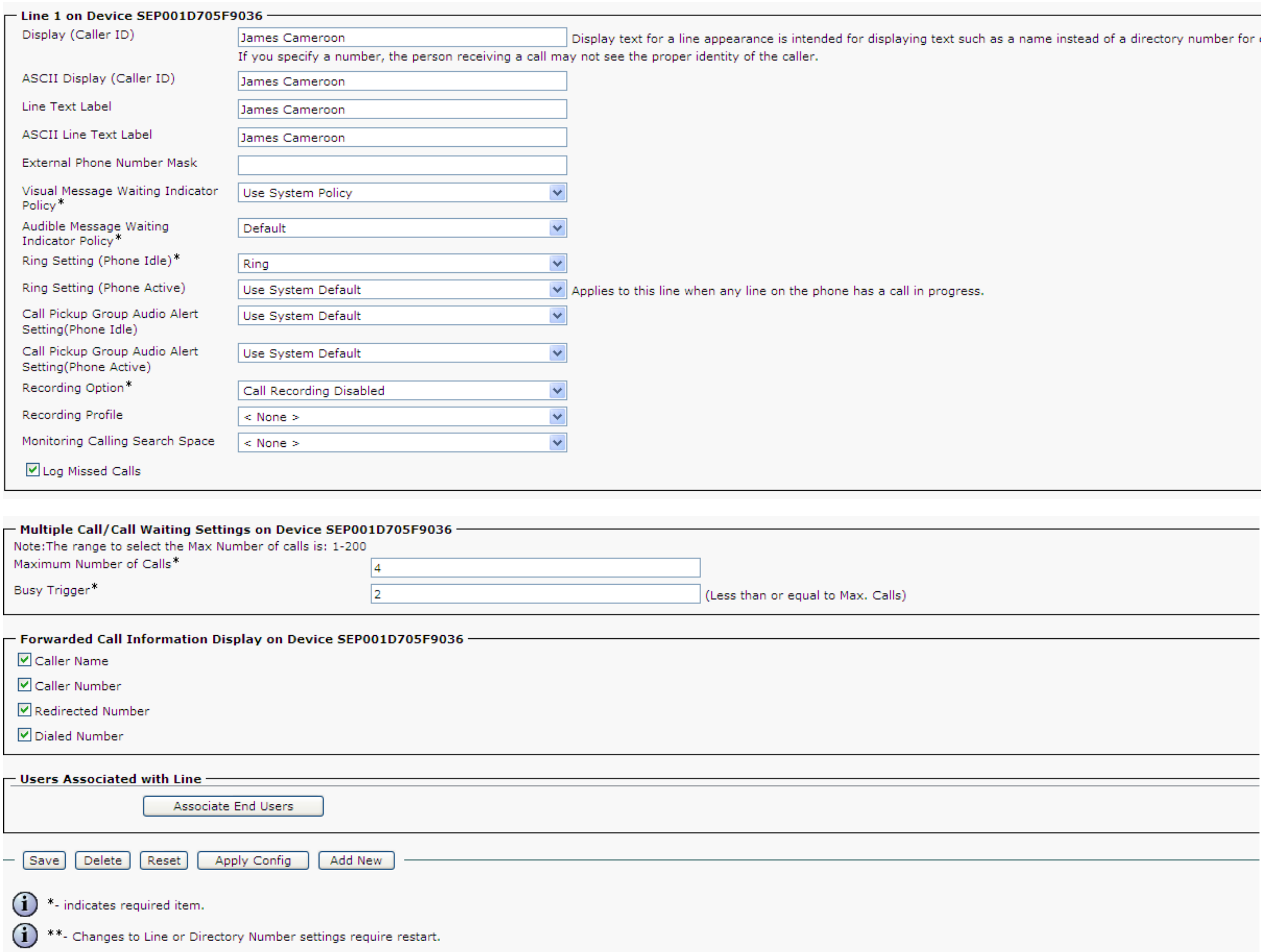
****Note: A Protected device means it is capable of playing Secure and Non-Secure Tones. When the checkbox is checked, the user will hear a Secure or Non-Secure Tone when the call is connected.

*****Note: A custom Softkey template without supplementary service Softkeys must be used for a Hot line Device.

Directory Number Information (Line [1]) of SCCP Phone with associated device SEP001D705F9036



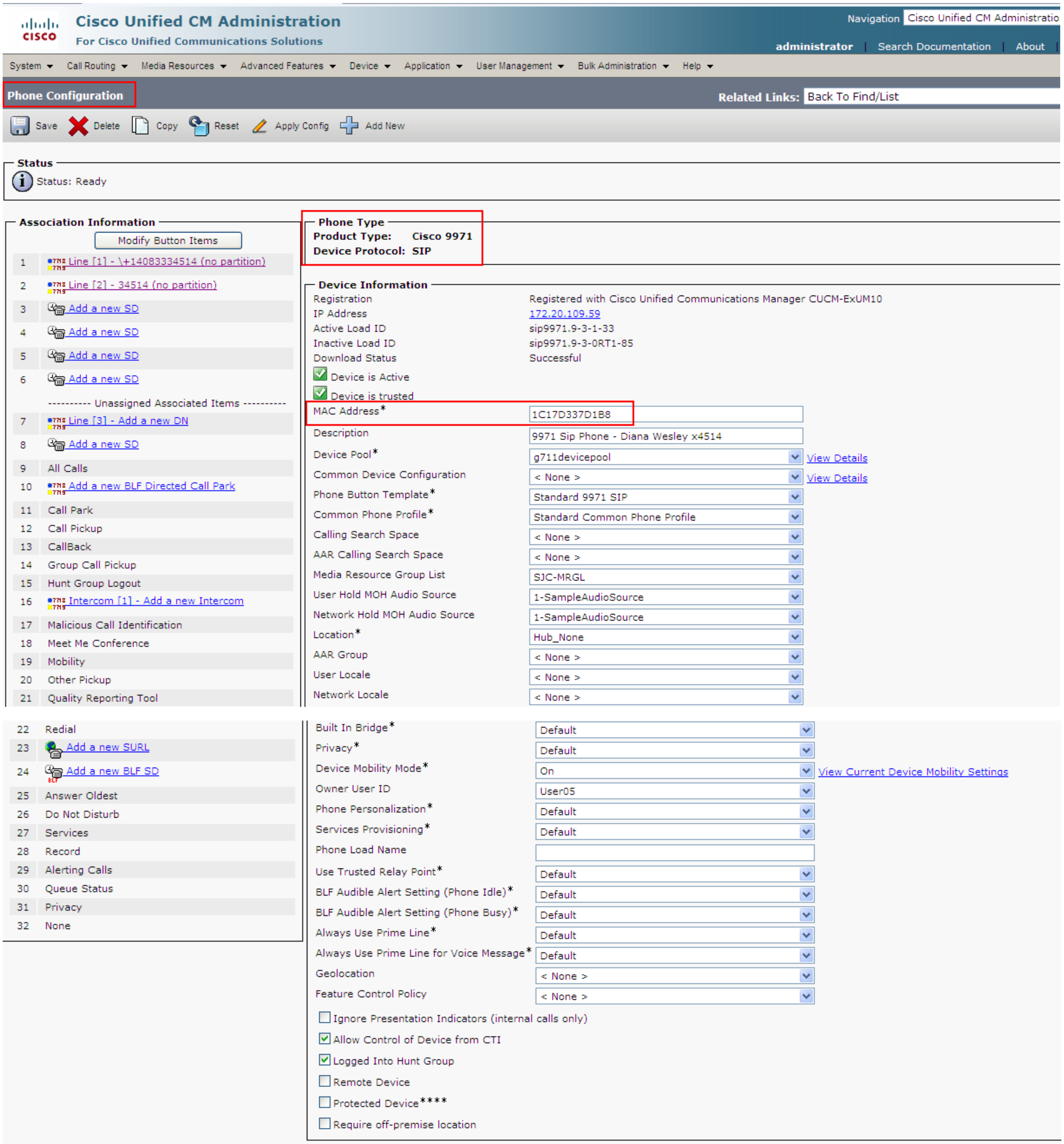

Directory URIs
Primary
URI
Partition
Remo
< None >
Add Row
AAR Settings
Voice Mail
AAR Destination Mask
AAR Group
AAR
or
< None >
Retain this destination in the call forwarding history
Call Forward and Call Pickup Settings
Voice Mail
Destination
Calling Search Space
Calling Search Space Activation Policy
Use System Default
Forward All
Secondary Calling Search Space for Forward All
Forward Busy Internal
Forward Busy External
Forward No Answer Internal
Forward No Answer External
Forward No Coverage Internal
Forward No Coverage External
Forward on CTI Failure
Forward Unregistered Internal
Forward Unregistered External
No Answer Ring Duration (seconds)
Call Pickup Group
Park Monitoring
Voice Mail
Destination
Calling Search Space
Park Monitoring Forward No Retrieve Destination External
A blank value means to call th parker's line.
Park Monitoring Forward No Retrieve Destination Internal
A blank value means to call th parker's line.
Park Monitoring Reversion Timer
0
A blank value will use value set in Park Monitoring Reversion Timer service parameter
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space
MLPP No Answer Ring Duration (seconds)
Line Settings for All Devices
Hold Reversion Ring Duration (seconds)
Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)
Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*
Default

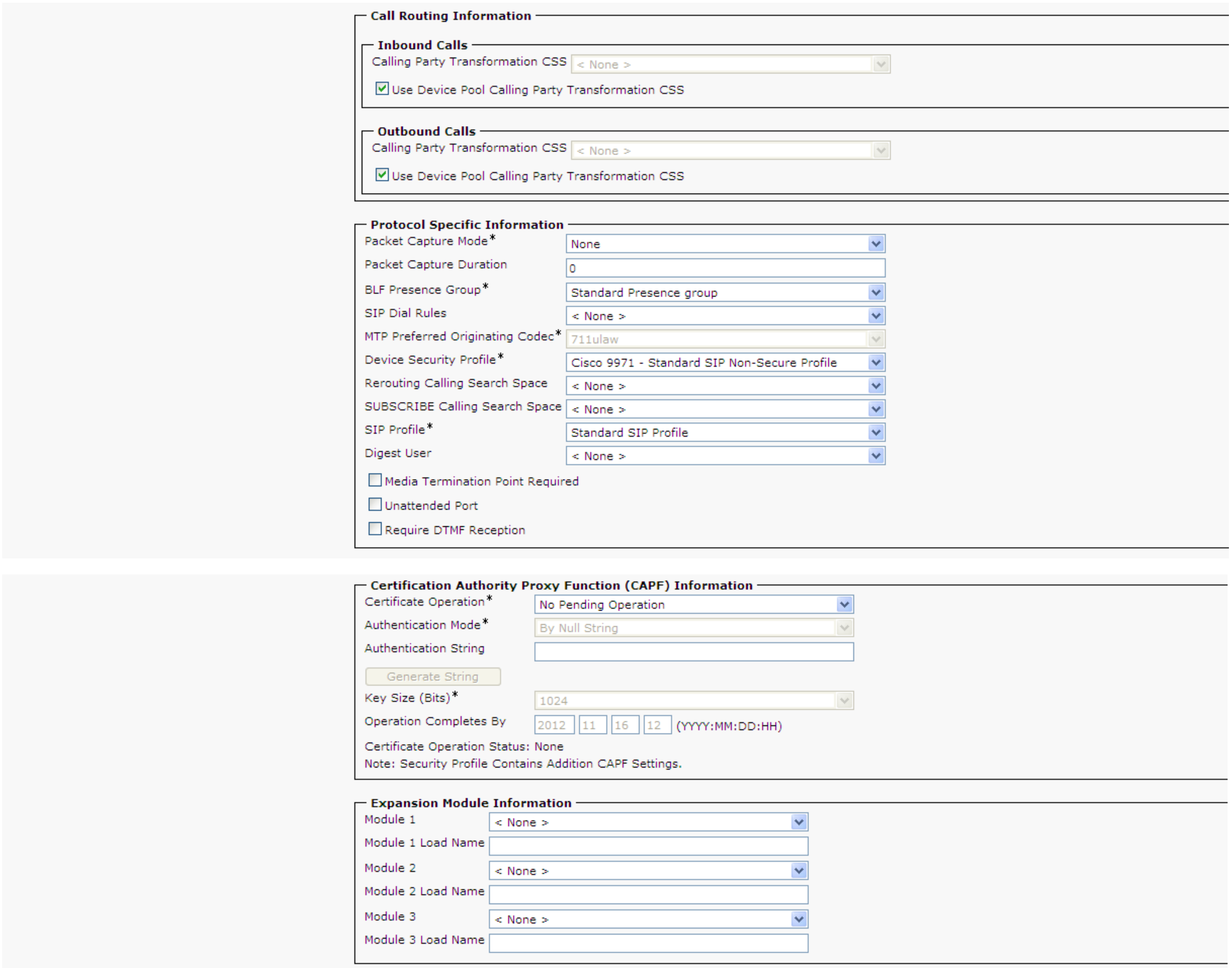

Line 1 on Device SEP001D705F9036
Display (Caller ID)
James Cameroon
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for
If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)
James Cameroon
Line Text Label
James Cameroon
ASCII Line Text Label
James Cameroon
External Phone Number Mask
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
Multiple Call/Call Waiting Settings on Device SEP001D705F9036
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 SEP001D705F9036
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.

CISCO

Device Name: SEP1C17D337D1B8

Device Protocol: SIP





Call Routing Information

Inbound Calls

Calling Party Transformation CSS: < None >

☑ Use Device Pool Calling Party Transformation CSS

Outbound Calls

Calling Party Transformation CSS: < None >

☑ Use Device Pool Calling Party Transformation CSS

Protocol Specific Information

Packet Capture Mode*	None
Packet Capture Duration	0
BLF 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
Digest User	< None >

☐ Media Termination Point Required

☐ Unattended Port

☐ Require DTMF Reception

Certification Authority Proxy Function (CAPF) Information

Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	

Generate String

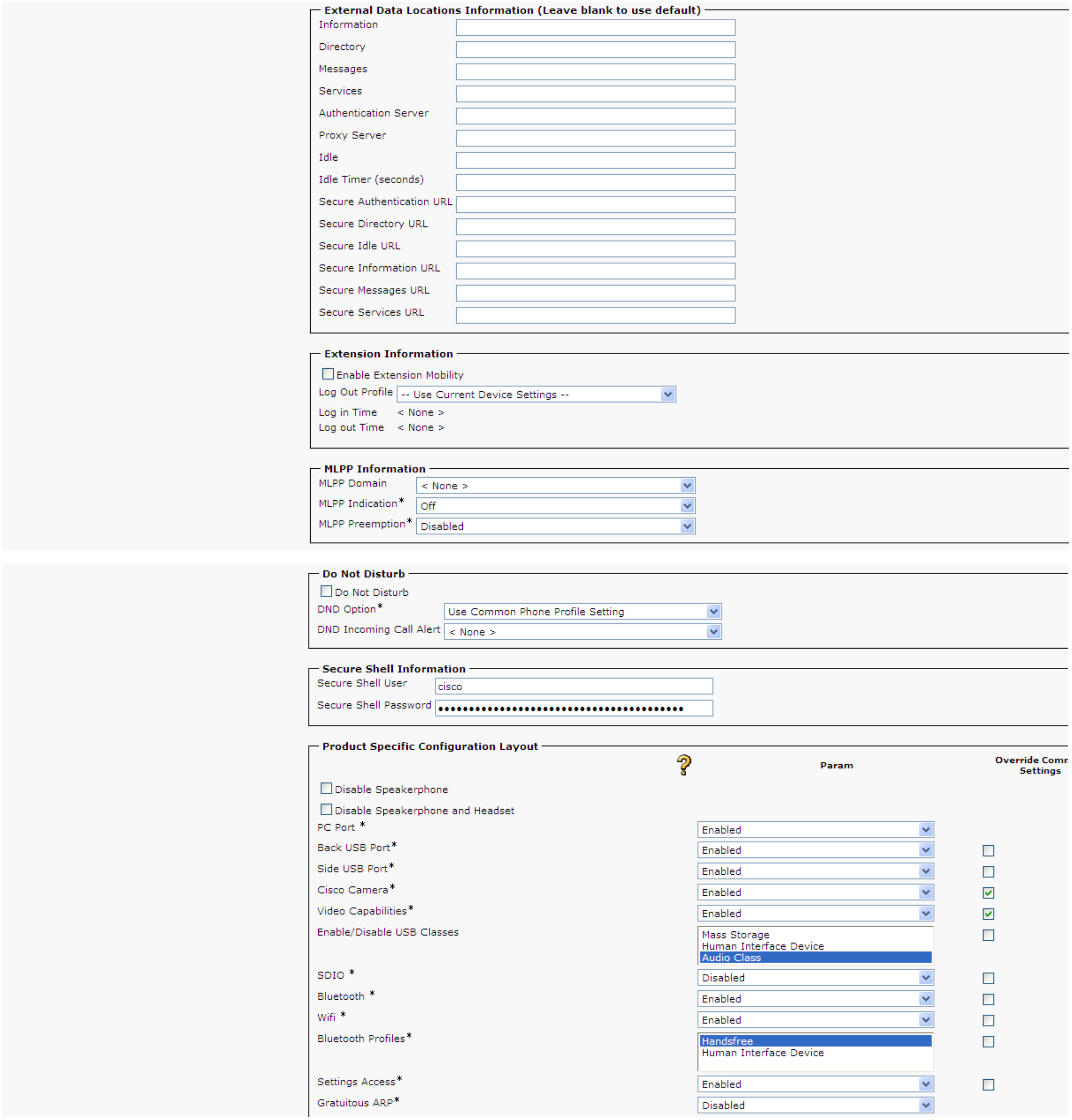
Key Size (Bits)*	1024
Operation Completes By	2012 11 16 12 (YYYY:MM:DD:HH)

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
Secure Messages URL
Secure Services URL
Extension Information
Enable Extension Mobility
Log Out Profile -- Use Current Device Settings --
Log in Time < None >
Log out Time < None >
MLPP Information
MLPP Domain < None >
MLPP Indication* Off
MLPP Preemption* Disabled
Do Not Disturb
Do Not Disturb
DND Option* Use Common Phone Profile Setting
DND Incoming Call Alert < None >
Secure Shell Information
Secure Shell User cisco
Secure Shell Password
Product Specific Configuration Layout
Param
Override Comr Settings
Disable Speakerphone
Disable Speakerphone and Headset
PC Port * Enabled
Back USB Port* Enabled
Side USB Port* Enabled
Cisco Camera* Enabled
Video Capabilities* Enabled
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	☐
Show All Calls on Primary Line*	Disabled	
Days Display Not Active	Sunday Monday Tuesday	☐
Display On Time	07:30	☐
Display On Duration	10:30	☐
Display Idle Timeout	01:00	☐
HTTPS Server*	http and https Enabled	☑
Enable Power Save Plus	Sunday Monday Tuesday	☐
Phone On Time	00:00	☐
Phone Off Time	24:00	☐
Phone Off Idle Timeout*	60	☐
☐ Enable Audible Alert		☐
EnergyWise Domain		☐
EnergyWise Endpoint Security Secret		☐
☐ Allow EnergyWise Overrides		☐
Span to PC Port*	Disabled	
Logging Display*	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*	Enabled	☑
Log Server		
Advertise G.722 and iSAC Codecs *	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	
SSH Access*	Disabled	☐
Incoming Call Toast Timer*	5	☐
Provide Dial Tone from Release Button*	Disabled	☐
Hide Video By Default*	Disabled	☐
Background Image		☐

Simplified New Call UI*	Disabled	☐
Enable VXC VPN for MAC		
VXC VPN Option*	Dual Tunnel	☐
VXC Challenge*	Challenge	☐
VXC-M Servers		☐
Revert to All Calls*	Disabled	☐
80-bit SRTCP*	Disabled	☐
RTCP for Video*	Enabled	☐
Record Call Log from Shared Line*	Disabled	☐

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

*- indicates required item.

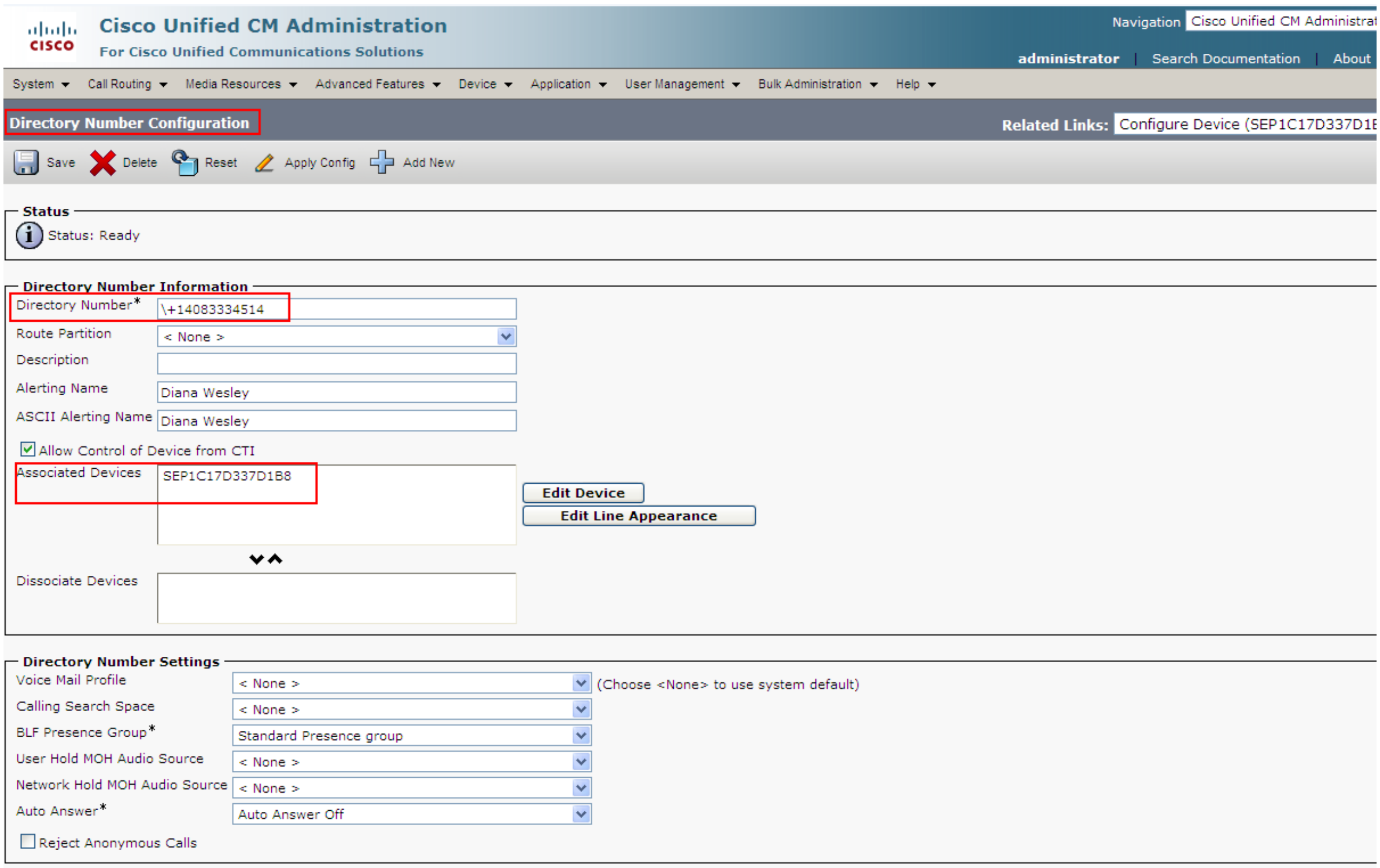
**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

***Note: Security Profile Contains Addition CAPF Settings.

****Note: A Protected device means it is capable of playing Secure and Non-Secure Tones. When the checkbox is checked, the user will hear a Secure or Non-Secure Tone when the call is connected.

*****Note: A custom Softkey template without supplementary service Softkeys must be used for a Hot line Device.

Directory Number Information (Line [1]) of SIP Phone with associated device SEP1C17D337D1B8



Directory URIs

Primary	URI	Partition	Rem
◉		< None >	

Add Row

AAR Settings

	Voice Mail	AAR Destination Mask	AAR Group
AAR	☐ or		< None >

☐ Retain this destination in the call forwarding history

Call Forward and Call Pickup Settings

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

No Answer Ring Duration (seconds)

Call Pickup Group: < None >

Park Monitoring

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

Park Monitoring Reversion Timer: 0 — 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 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 SEP1C17D337D1B8

Display (Caller ID): Diana Wesley — Display text for a line appearance is intended for displaying text such as a name instead of a directory number for c

If you specify a number, the person receiving a call may not see the proper identity of the caller.

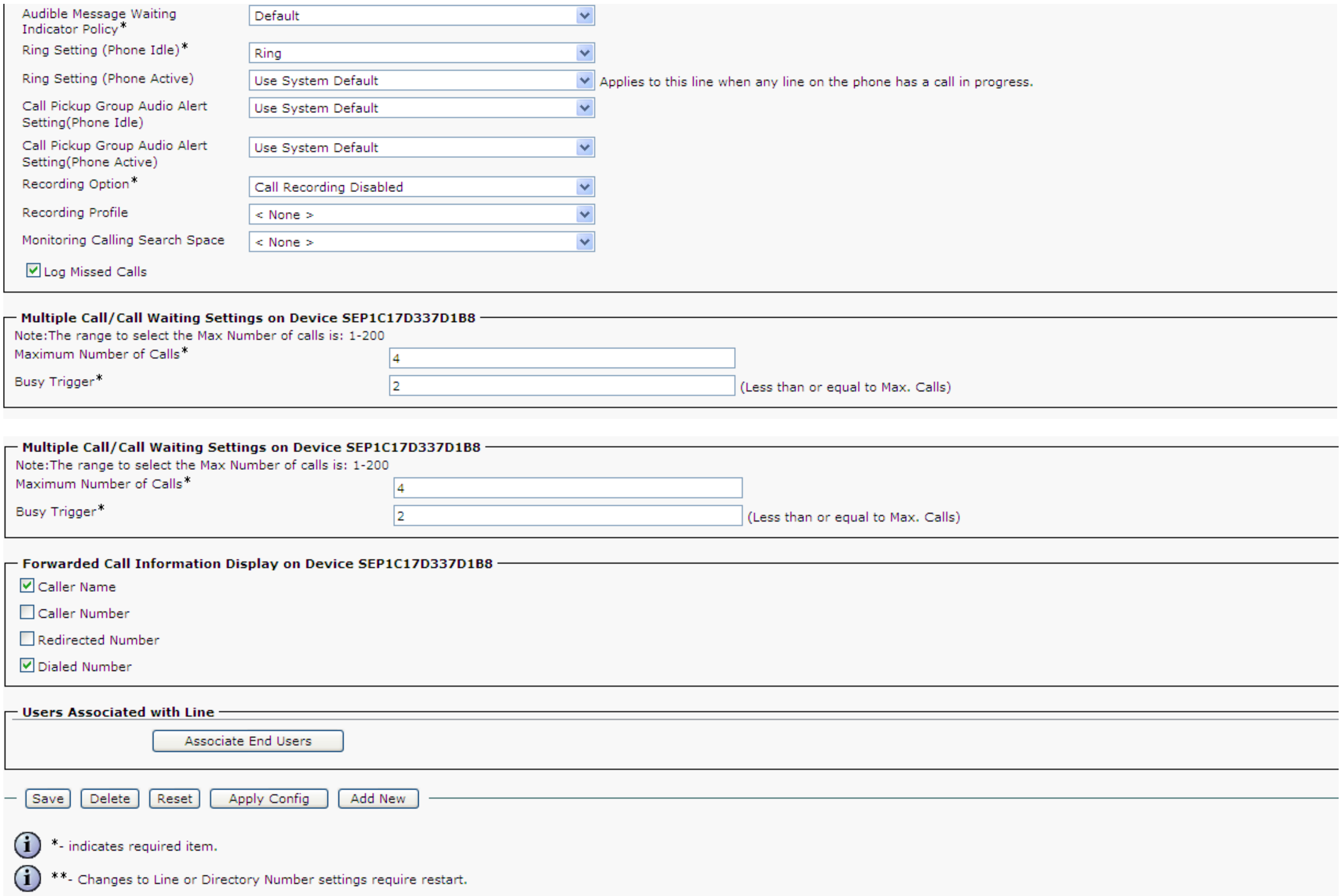
ASCII Display (Caller ID): Diana Wesley

Line Text Label: Diana Wesley

ASCII Line Text Label: Diana Wesley

External Phone Number Mask

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
Multiple Call/Call Waiting Settings on Device SEP1C17D337D1B8
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)
Multiple Call/Call Waiting Settings on Device SEP1C17D337D1B8
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 SEP1C17D337D1B8
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.

Cisco IOS Gateway Configurations

Cisco Unified Border Element Configuration on the ASR

```
CUBE-ASR1K_Vz_152>en
Password:
CUBE-ASR1K_Vz_152#
CUBE-ASR1K_Vz_152#
CUBE-ASR1K_Vz_152#
CUBE-ASR1K_Vz_152#
CUBE-ASR1K_Vz_152#sho ver
CUBE-ASR1K_Vz_152#sho version
Cisco IOS Software, IOS-XE Software (PPC_LINUX_IOSD-ADVENTERPRISE-M), Version 15.1(3)S2, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2011 by Cisco Systems, Inc.
Compiled Mon 12-Dec-11 15:09 by mcpre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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_152 uptime is 2 days, 16 hours, 8 minutes
Uptime for this control processor is 2 days, 16 hours, 10 minutes
System returned to ROM by reload
System image file is "bootflash:asr1000rp1-adventerprise.03.04.02.S.151-3.S2.bin"
Last reload reason: Reload Command


cisco ASR1002 (2RU) processor with 1708551K/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_152#sho run
CUBE-ASR1K_Vz_152#sho running-config
Building configuration...

Current configuration : 8552 bytes
!
! Last configuration change at 14:56:29 UTC Fri Oct 26 2012
!
version 15.1
```

```
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CUBE-ASR1K_Vz_152
!
boot-start-marker
boot system bootflash:asr1000rp1-adventerprise.03.04.02.S.151-3.S2.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging buffered 20000000
enable password cisco
!
no aaa new-model
!
ipc zone default
 association 1
  no shutdown
!
!
!
!
ip domain name ciscolab.globalipcom.com
ip name-server 172.31.124.36
!
!
!
ipv6 multicast rpf use-bgp
ipv6 multicast vrf Mgmt-intf rpf use-bgp
!
!
!
multilink bundle-name authenticated
!
!
!
voice service voip
 allow-connections sip to sip
 no supplementary-service sip refer
 redirect ip2ip
 sip
  header-passing
  asserted-id pai
  localhost dns:gw1.ciscolab.globalipcom.com
  no update-callerid
  early-offer forced
  history-info
  midcall-signaling passthru
  privacy-policy passthru
  g729 annexb-all
```

```
  sip-profiles 2
  no call service stop
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
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, CANCEL, ACK, PRACK, SUBSCRIBE, NOTIFY, INFO, REGISTER"
 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, CANCEL, 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 /81/ /1\1/
!
voice translation-rule 2
 rule 1 /^84089550468/ /14089550468/
!
!
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 ios_user
ip ftp password cisco
!
!
!
!
```

```
!
interface GigabitEthernet0/0/0
 ip address 172.20.110.152 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 no ip address
 standby delay minimum 30 reload 60
 standby version 2
 standby 6 priority 50
 standby 6 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 forward-protocol nd
!
no ip http 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
ip route 172.31.124.0 255.255.255.0 172.20.110.150
!
!
!
!
control-plane
!
!
!
dial-peer voice 9000 voip[1]
 description to Service Provider - SP facing
 destination-pattern 1T
 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
```

[1] Outbound dial peer towards Service Provider-SP facing. This dial peer is used for the outbound call from SME to Service Provider

```
!
dial-peer voice 9001 voip
 description Incoming to SME - SME facing
 destination-pattern 408933....
 session protocol sipv2
 session target ipv4:172.20.236.252
 session transport tcp
 dtmf-relay rtp-nte
 fax protocol pass-through g711ulaw
!
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 6170 voip
 description To Fax Machines on Siemens
 destination-pattern 408933617.
 session protocol sipv2
 session target ipv4:172.20.236.252
 session transport udp
 incoming called-number 408933617.
 codec g711ulaw
 fax protocol pass-through g711ulaw
!
dial-peer voice 3470 voip
 description To Fax Machines on CUCM
 destination-pattern 40893334..
 session protocol sipv2
 session target ipv4:172.20.236.252
 session transport udp
 incoming called-number 40893334..
 codec g711ulaw
 fax protocol pass-through g711ulaw
!
dial-peer voice 408 voip
 description To Service Provider 7-digit dialing - SP facing
 shutdown
 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 9003 voip
 description Fax calls using G.711u - SP facing
 translation-profile outgoing Outgoing-Fax-G711
 destination-pattern 81..........
 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[2]
 description to Service Provider - 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
!
dial-peer voice 9901 voip
 description Incoming to SME - SP facing
 session protocol sipv2
 session target sip-server
 session transport tcp
 incoming called-number 408933....
 dtmf-relay rtp-nte
 fax protocol pass-through g711ulaw
!
dial-peer voice 4408 voip
 description To Service Provider 7-digit dialing - SME facing
 shutdown
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number .......
 dtmf-relay rtp-nte
 fax protocol pass-through g711ulaw
!
dial-peer voice 9903 voip
 description Fax calls using G.711u - SME facing
 session protocol sipv2
 session target sip-server
 incoming called-number 81..........
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 codec g711ulaw
 fax protocol pass-through g711ulaw
!
dial-peer voice 9004 voip
 description To Service Provider 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
 fax protocol pass-through g711ulaw
!
dial-peer voice 9904 voip
 description To Service Provider International calls - SME facing
 session protocol sipv2
```

[2] Outbound dial peer towards Service Provider-SME facing. This dial peer is used for the outbound call from SME to Service Provider

```
 session target sip-server
 session transport udp
 incoming called-number 011T
 dtmf-relay rtp-nte
 fax protocol pass-through g711ulaw
!
dial-peer voice 237 voip
 description Outgoing to CM-Polaris
 shutdown
 destination-pattern 408273....
 session protocol sipv2
 session target ipv4:172.20.236.50
 session transport tcp
!
dial-peer voice 408273 voip
 description incoming call from VZ to SME
 destination-pattern 408273398[12]
 session protocol sipv2
 session target ipv4:172.20.85.110:5060
 session transport tcp
 voice-class codec 1
 no voice-class sip early-offer forced
 dtmf-relay rtp-nte
!
dial-peer voice 1111 voip[3]
 description Incoming-to-SME-172-20-109-254 SME-facing
 destination-pattern 408273398.
 session protocol sipv2
 session target ipv4:172.20.109.254
 session transport tcp
 dtmf-relay rtp-nte
 codec g711ulaw
!
dial-peer voice 1112 voip[4]
 description Incoming-to-SME-172-20-109-254 SP-facing
 session protocol sipv2
 session target sip-server
 session transport tcp
 incoming called-number 408273398.
 dtmf-relay rtp-nte
 codec g711ulaw
!
!
sip-ua
 set pstn-cause 1 sip-status 503
 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
```

[3] Incoming dial peer towards SME - SME facing. This dial peer is used for the inbound call to SME from Service Provider
[4] Incoming dial peer towards SME - SP facing. This dial peer is used for the inbound call to SME from Service provider

```
 password cisco
 login
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 password cisco
 login
!
end
```

CUBE-ASR1K_Vz_152#

Acronyms

Acronym	Definition
Lync	Lync 2010 Server
DTMF	Dual Tone Multi Frequency
SIP	Session Initiation Protocol
SDP	Session Description Protocol
B2BUA	Back to Back User Agent
LyncIT	Lync server 2010 Interoperability Test Tool
TCP	Transmission Control Protocol
TLS	Transport Layer Security
GW	Gateway
S/W	Software
ASR	Aggregation Services Routers
Cisco UCM-SME	Cisco Unified Communications Manager – Session Manager Edition
CUBE	Cisco Unified Border Element
SP	Service Provider
PSTN	Public Switched Telephone Network
FQDN	Fully Qualified Domain Name
SCCP	Skinny Call Control Protocol
MTP	Media Termination Point
UDP	User Datagram Protocol
RTCP	Real Time Control Protocol

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