

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

September 28, 2011 - Initial Version

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Introduction

- This application note describes the necessary steps and configurations for connectivity between Cisco Unified Communications Manager-Session Manager Edition (Cisco UCM-SME) 8.5(1) via Direct SIP to Microsoft Lync 2010 server using Microsoft Mediation Server, and via SIP connection to a Cisco Unified Communications Manager (Cisco UCM) 8.5(1), and SIP connection to simulated Service Provider (SP) using Cisco Unified Border Element (Cisco UBE) 1.4. The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability between the different leaf nodes (Microsoft Lync 2010, Cisco UCM and Cisco UBE/SP) via the Cisco UCM-SME.
- Features tested are basic call, 3-way (ad-hoc) conference, call transfer (attended and unattended), call forward (all, busy and no answer), hold/resume, DTMF inter-working and voice mail. This test setup also includes a connection to a simulated service provider using SIP trunks. Cisco UBE is used as a session border controller (SBC), providing demarcation, security, and inter-working services between the customer's private network and the service provider's SIP network. Voice mail was tested for Cisco UCM users with mailboxes registered to Unity Connection via a Direct SIP trunk from Cisco UCM-SME. Fax could not be tested (see limitations section).
- During testing, a Cisco ASR 1002 router was used as Cisco UBE, however other Cisco voice gateways, such as the 3900 series or AS5400 XM can be used and the decision to choose what Cisco gateway model to use is left to the customer. The customer should choose a Cisco IOS gateway model based on the capabilities and the capacity that will be required based on the planned network deployment. Here is a list of Cisco IOS products capable of running Cisco UBE.

[Cisco ASR 1000 Series Aggregation Services Router](#)

[Cisco 3900 Series Integrated Services Routers](#)

[Cisco 2900 Series Integrated Services Routers](#)

[Cisco 3800 Series Integrated Services Routers](#)

[Cisco 2800 Series Integrated Services Routers](#)

[Cisco AS5400XM Universal Gateway](#)

[Cisco AS5350XM Universal Gateway](#)

- If additional guidance on the Cisco UBE is needed, please refer to the Cisco UBE section on the Cisco Interoperability Portal (www.cisco.com/go/interoperability).
- This configuration was tested to simulated service provider. Results may vary based on service provider being used.



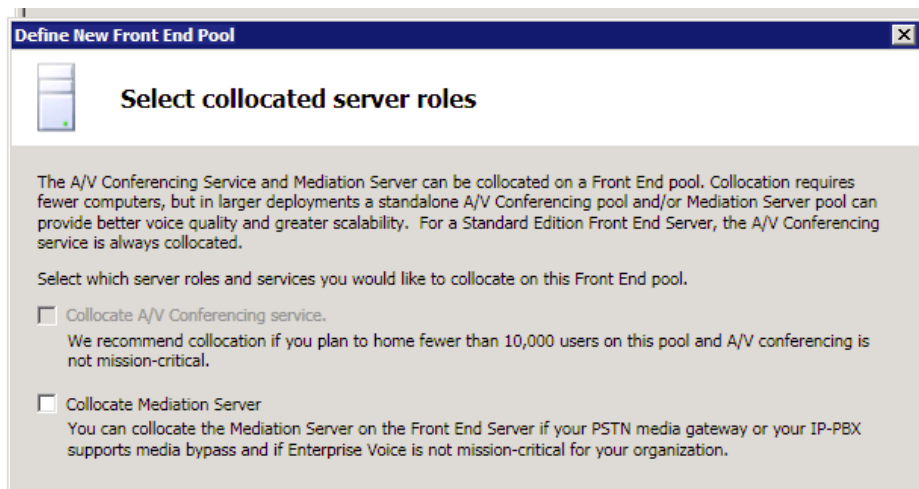
Capabilities

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

Limitations

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

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



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

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

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

- Microsoft Lync 2010 only supports TCP transport to carry SIP messages. If UDP is required CUBE can be used to perform the conversion.
- Microsoft Lync 2010 only supports G711 ulaw or alaw on the outside interface. If G729 is required CUBE or transcoder registered to SME can perform the transcoding function.
- Microsoft Lync 2010 does not support G711 Fax Passthrough.
- Fax T.38 to Service provider is not tested. Service Provider used for this test does not support Fax T.38 switchover.
- On Lync Mediation Server, RTCP is set to disabled, as RTCP is an end point specific configuration and not all Cisco phones support RTCP. Phones like 7970, 7962, and 9971 have this support and this can be enabled under Phone Configuration – Product Specific Configuration Layout.



Features Supported

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

Features Not Supported and Not Tested

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



System Components

Hardware Requirements

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

Software Requirements

The following software is required:

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

IMPORTANT:

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



System Configuration

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

Configuring the Microsoft LYNC 2010 Server Standard Edition

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

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

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

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

For Microsoft deployment and installation guides, go to:

<http://technet.microsoft.com/en-us/library/gg293124.aspx>

<http://technet.microsoft.com/en-us/library/gg398616.aspx>



Configuration Steps:

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

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

Domain Name System Configuration

Start > Administrative Tools > DNS



Forward Lookup Zone

Name	Type	Data	Timestamp
_msdcs			
_sites			
_tcp			
_udp			
DomainDnsZones			
ForestDnsZones			
MedServer34			
LYNC2010CLIENT2	Host (A)	172.20.109.170	7/27/2011 4:00:00
CUP-DUCATI	Host (A)	172.20.109.251	static
CM-TELUGU	Host (A)	172.20.109.254	static
UCCLIENT12	Host (A)	172.20.109.54	7/22/2011 3:00:00
Lab-PC-3	Host (A)	172.20.109.56	7/25/2011 3:00:00
(same as parent folder)	Host (A)	172.20.117.120	8/19/2011 8:00:00
lync2010ad-rtm	Host (A)	172.20.117.120	static
LYNC2010MEDPRM	Host (A)	172.20.117.130	8/19/2011 3:00:00
MedServer3	Host (A)	172.20.117.131	8/16/2011 5:00:00
MedServer4	Host (A)	172.20.117.132	4/11/2011 2:00:00
LYNC2010FE1-RTM	Host (A)	172.20.117.152	8/16/2011 3:00:00
LYNC2010Client1	Host (A)	172.20.117.203	5/29/2011 7:00:00
LYNC2010CLIENT3	Host (A)	172.20.201.169	8/21/2011 3:00:00
LYNC2010CLIENT4	Host (A)	172.20.201.171	8/16/2011 2:00:00
lyncpc1-PC	Host (A)	172.20.8.141	8/17/2011 3:00:00
lyncpc2-PC	Host (A)	172.20.8.142	8/17/2011 3:00:00
lyncpc3-PC	Host (A)	172.20.8.143	8/18/2011 11:00:00
CUBE	Host (A)	172.20.85.101	static
CUCM-EXUM10	Host (A)	172.20.85.110	static
CUCM1	Host (A)	172.20.85.110	static
CUCM2	Host (A)	172.20.85.110	static
(same as parent folder)	Start of Authority (SOA)	[562], lync2010ad-rtm.lync2010rtm.com., hostmaster.lync...	static
(same as parent folder)	Name Server (NS)	lync2010ad-rtm.lync2010rtm.com.	static

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



Reverse Lookup Zones Configuration

The screenshot shows the DNS Manager console with the following data:

Name	Type	Data	Timestamp
(same as parent folder)	Start of Authority (SOA)	[33], lync2010ad-rtm.lync2010rtm.com., hostmaster.lync2...	static
(same as parent folder)	Name Server (NS)	lync2010ad-rtm.lync2010rtm.com.	static
172.20.117.120	Pointer (PTR)	lync2010ad-rtm.lync2010rtm.com.	8/19/2011 5:00:00
172.20.117.203	Pointer (PTR)	lync2010client1.lync2010rtm.com.	5/29/2011 7:00:00
172.20.117.152	Pointer (PTR)	lync2010fe1-rtm.lync2010rtm.com.	static
172.20.117.130	Pointer (PTR)	lync2010medprtm.lync2010rtm.com.	static
172.20.117.131	Pointer (PTR)	medserver3.lync2010rtm.com.	static
172.20.117.132	Pointer (PTR)	medserver4.lync2010rtm.com.	static

PTR Records added for the Lync server and Active Directory.

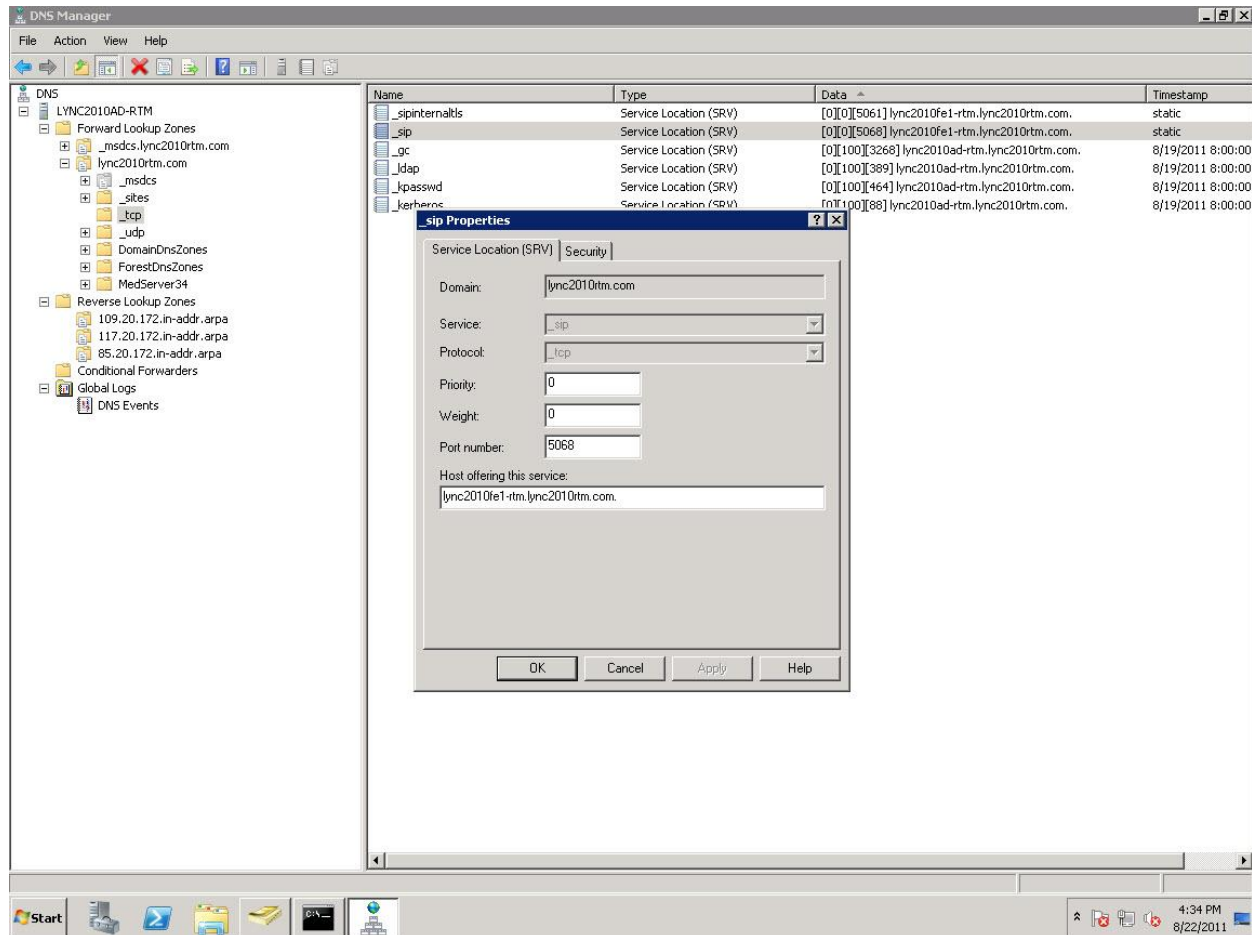


SRV Records Configuration

The screenshot shows the DNS Manager console with the following table of SRV records:

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
_lpsswld	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

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
 - _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

The screenshot displays the Lync Server 2010 Topology Builder application. The left pane shows a tree view of the topology, with the site **LYNC2010FE1-RTM.lync2010rtm.com** selected. The main pane shows the configuration details for this site, organized into several sections:

- 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**:
 - Internal web services

The right pane shows the **Actions** menu for the selected site, including options like **Edit Properties...**, **Topology**, **View**, **Delete**, and **Help**.



Lync Server 2010, Topology Builder

File Action View Help

Lync Server 2010

- LyncRTM
 - Standard Edition Front End Servers
 - LYNC2010FE1-RTM.lync2010rtm.com
 - Enterprise Edition Front End pools
 - Director pools
 - A/V Conferencing pools
 - SQL stores
 - File stores
 - Mediation pools
 - LYNC2010FE1-RTM.lync2010rtm.com
 - MedServer34.lync2010rtm.com
 - MedServer4.lync2010rtm.com
 - PSTN gateways
 - CUCM-ExUM10.lync2010rtm.com
 - 172.20.201.254
 - 172.20.85.110
 - CUCM1.lync2010rtm.com
 - CUCM2.lync2010rtm.com
 - 172.20.85.101
 - 172.20.8.41
 - 172.20.201.169
 - Monitoring Servers
 - Archiving Servers
 - Edge pools
 - Trusted application servers
 - Branch sites

Internal web services

Ports: HTTP: 80 HTTPS: 443

External web services

FQDN: LYNC2010FE1-RTM.lync2010rtm.com

Ports: HTTP: 8080 HTTPS: 4443

Conferencing

- Instant Messaging Conferencing service: Enabled
- Web Conferencing service: Enabled
- Application Sharing service: Enabled
- A/V Conferencing service: Enabled

Mediation Server

Collocated Mediation Server: Enabled

TLS listening port: 5067

TCP listening port: 5068

PSTN Gateways:

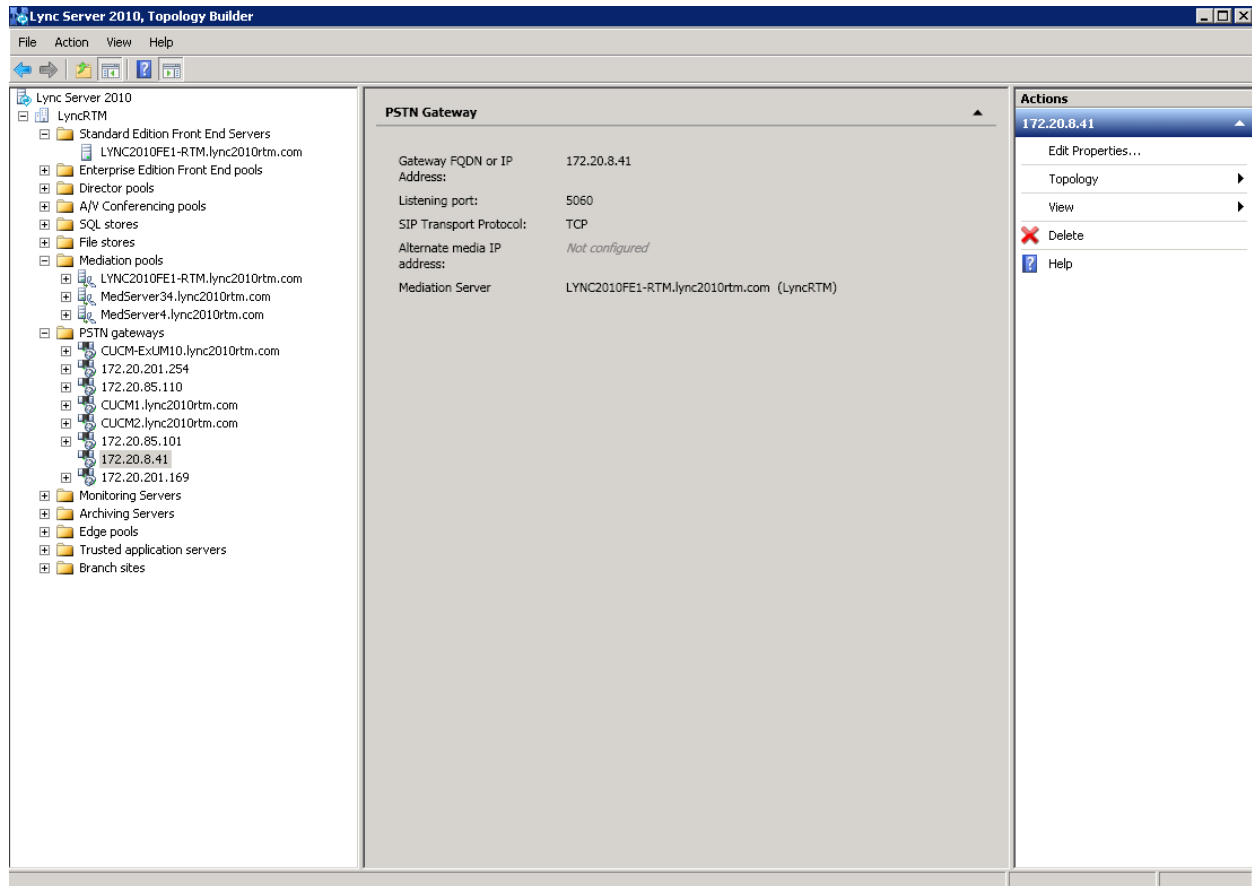
Default	Gateway	Site
	CUCM-ExUM10.lync2010rtm.com	LyncRTM
	172.20.201.254	LyncRTM
	172.20.85.110	LyncRTM
	172.20.85.101	LyncRTM
✓	172.20.8.41	LyncRTM
	172.20.201.169	LyncRTM

Actions

LYNC2010FE1-RTM.lync2010rtm...

- 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)



Listening port of the SME is left to default 5060.



Mediation Server Overview

The screenshot displays the Lync Server 2010 Topology Builder application. The left pane shows a tree view of the topology, with 'LyncRTM' selected under 'Lync Server 2010'. The main pane shows the configuration for the 'LyncRTM' site. The 'Site' section includes fields for Name, Description, City, State/Province, and Country/Region Code. The 'Call Admission Control Setting' section shows the Call Admission Control set to 'LYNC2010FE1-RTM.lync2010rtm.com'. The 'Site federation route assignment' section shows the Federation status as 'Disabled'. The right pane shows the 'Actions' menu for 'LyncRTM', with options for New, Edit Properties..., Topology, View, Delete, and Help.

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-ExtUM10.lync2010rtm.com
 - 172.20.201.254
 - 172.20.85.110
 - CUCM1.lync2010rtm.com
 - CUCM2.lync2010rtm.com
 - 172.20.85.101
 - 172.20.8.41
 - 172.20.201.169
 - Monitoring Servers
 - Archiving Servers
 - Edge pools
 - Trusted application servers
 - Branch sites

Site

Name: LyncRTM
Description: LyncRTM
City: San Jose
State/Province: CA
Country/Region Code: USA

Call Admission Control Setting

Call Admission Control: LYNC2010FE1-RTM.lync2010rtm.com

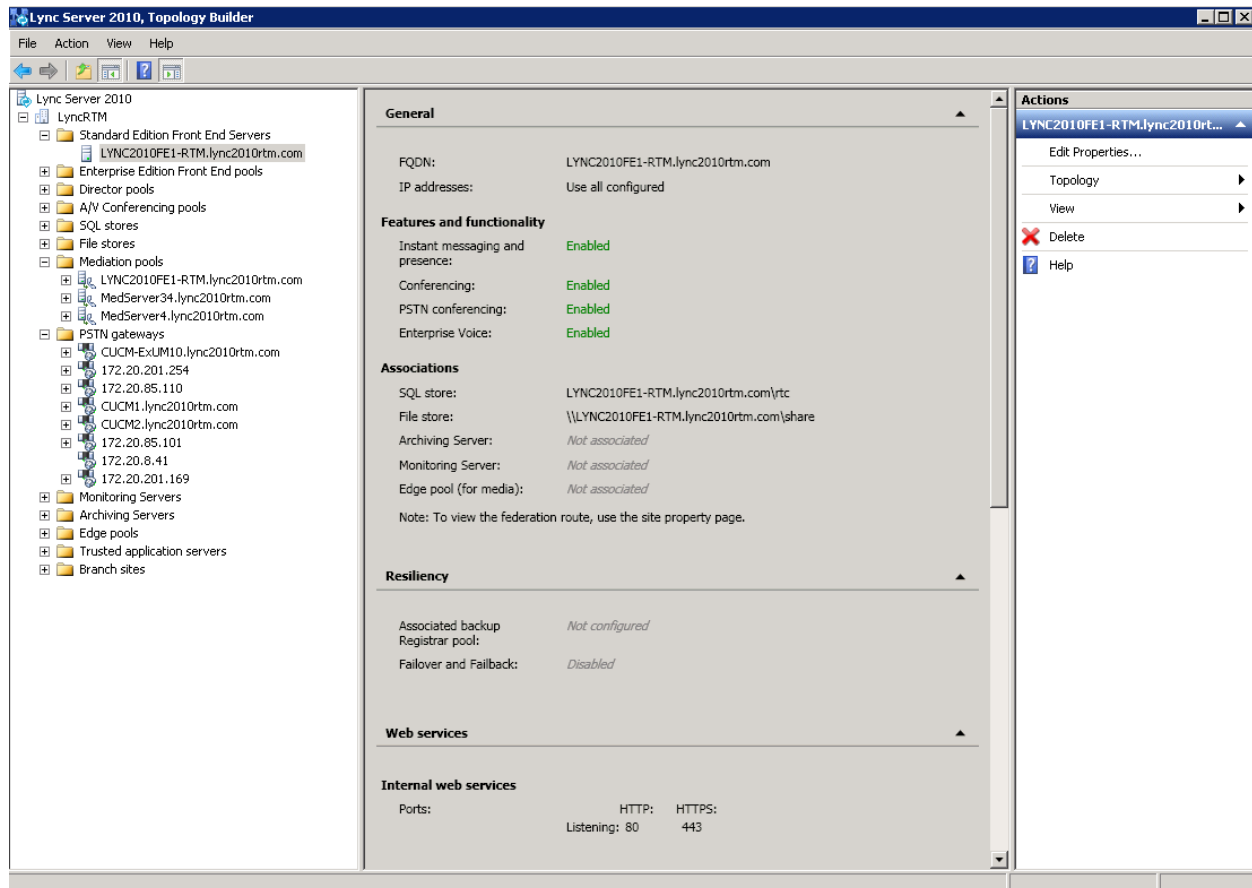
Site federation route assignment

Federation: Disabled

Actions

LyncRTM

- New
- Edit Properties...
- Topology
- View
- Delete
- Help



LYNC Front End server configuration with FQDN



Lync Server 2010, Topology Builder

File Action View Help

Lync Server 2010

- LyncRTM
 - Standard Edition Front End Servers
 - LYNC2010FE1-RTM.lync2010rtm.com
 - Enterprise Edition Front End pools
 - Director pools
 - A/V Conferencing pools
 - SQL stores
 - File stores
 - Mediation pools
 - LYNC2010FE1-RTM.lync2010rtm.com
 - MedServer34.lync2010rtm.com
 - MedServer4.lync2010rtm.com
 - PSTN gateways
 - CUCM-ExUM10.lync2010rtm.com
 - 172.20.201.254
 - 172.20.85.110
 - CUCM1.lync2010rtm.com
 - CUCM2.lync2010rtm.com
 - 172.20.85.101
 - 172.20.8.41
 - 172.20.201.169
 - Monitoring Servers
 - Archiving Servers
 - Edge pools
 - Trusted application servers
 - Branch sites

Mediation Server PSTN gateway

TLS listening port: 5067
TCP listening port: 5068

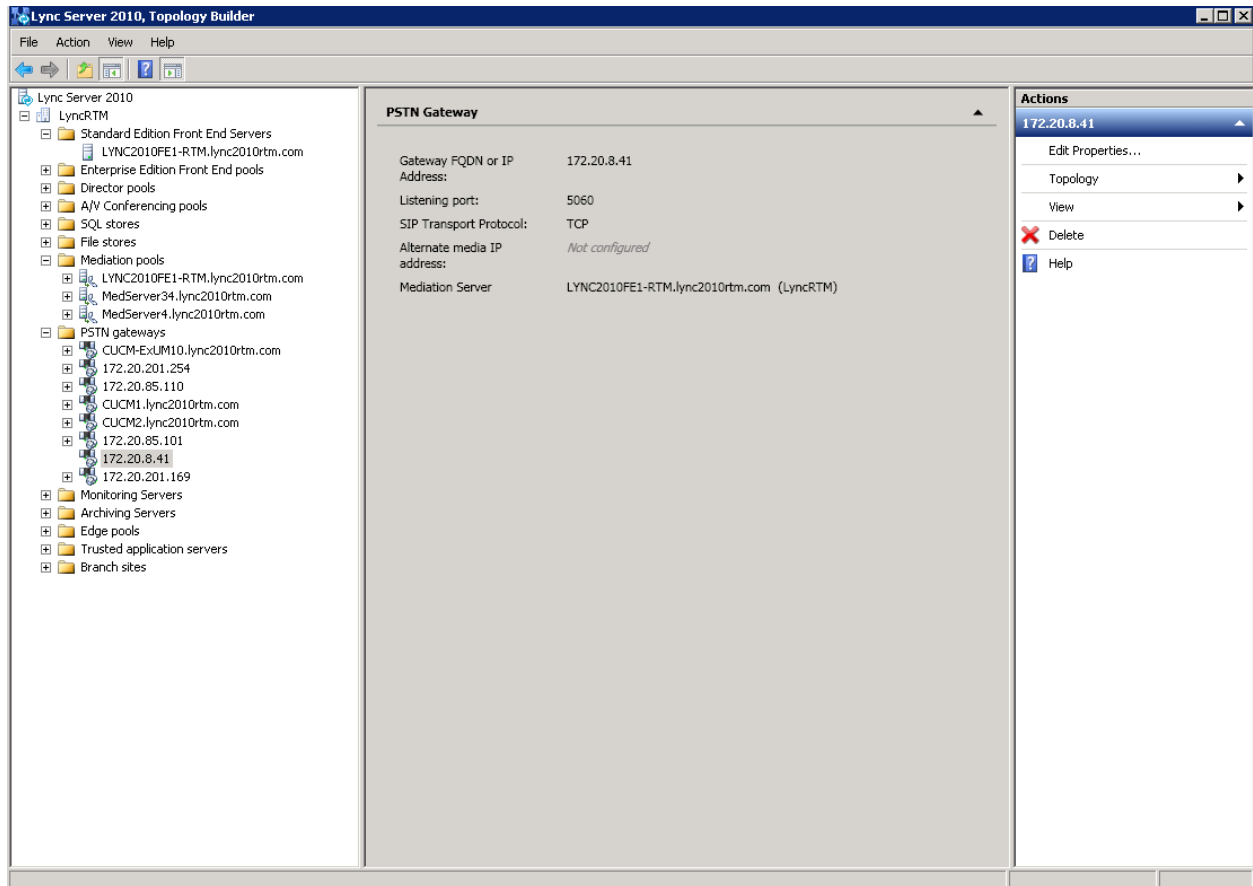
PSTN Gateways:

Default	Gateway	Site
	CUCM-ExUM10.lync2010rtm.com	LyncRTM
	172.20.201.254	LyncRTM
	172.20.85.110	LyncRTM
	172.20.85.101	LyncRTM
✓	172.20.8.41	LyncRTM
	172.20.201.169	LyncRTM

Actions

LYNC2010FE1-RTM.lync2010rtm...

- Edit Properties...
- Topology
- View
- Help

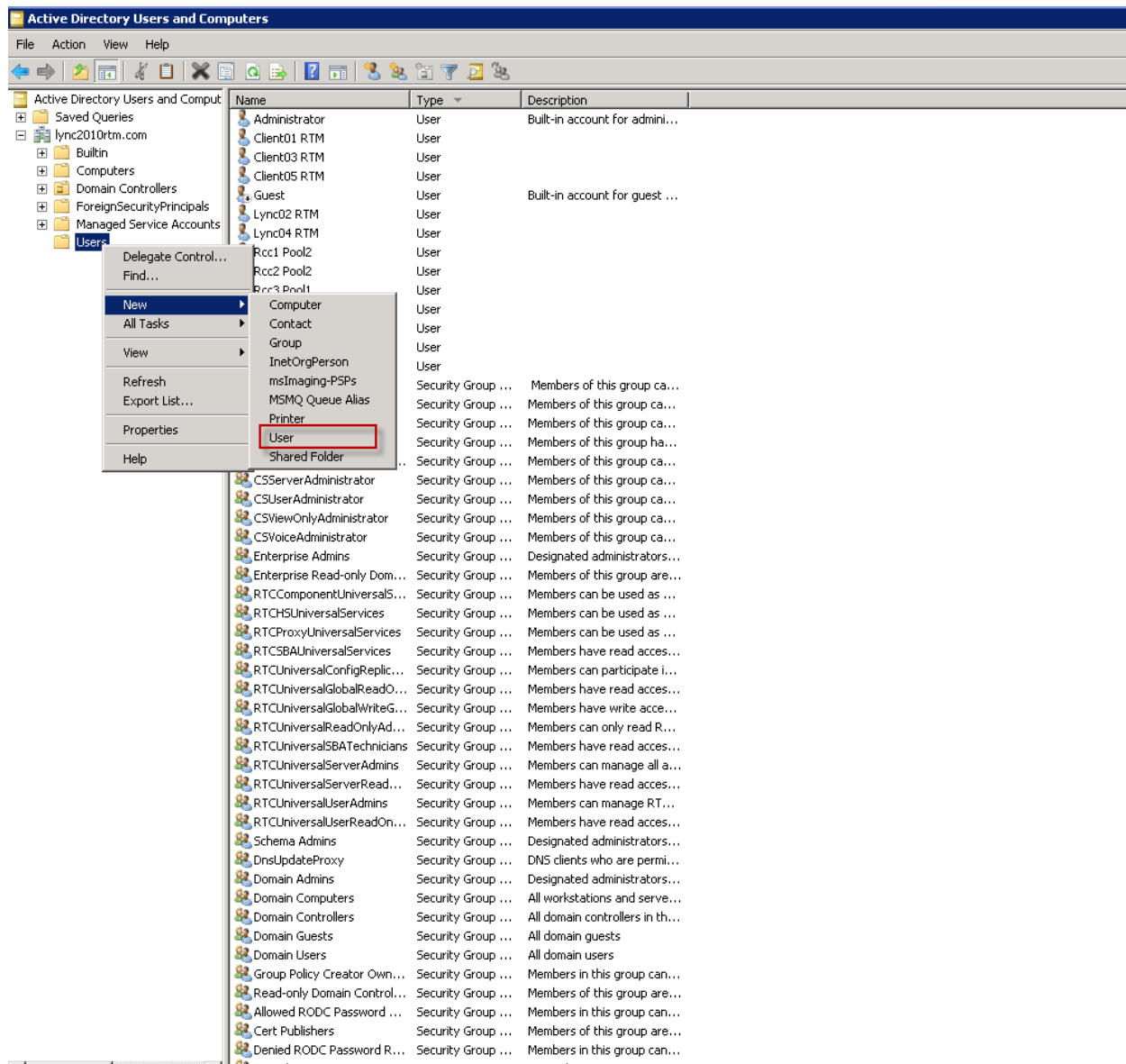


PSTN Gateway associated with the Mediation Pool



Active Directory User Configuration

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





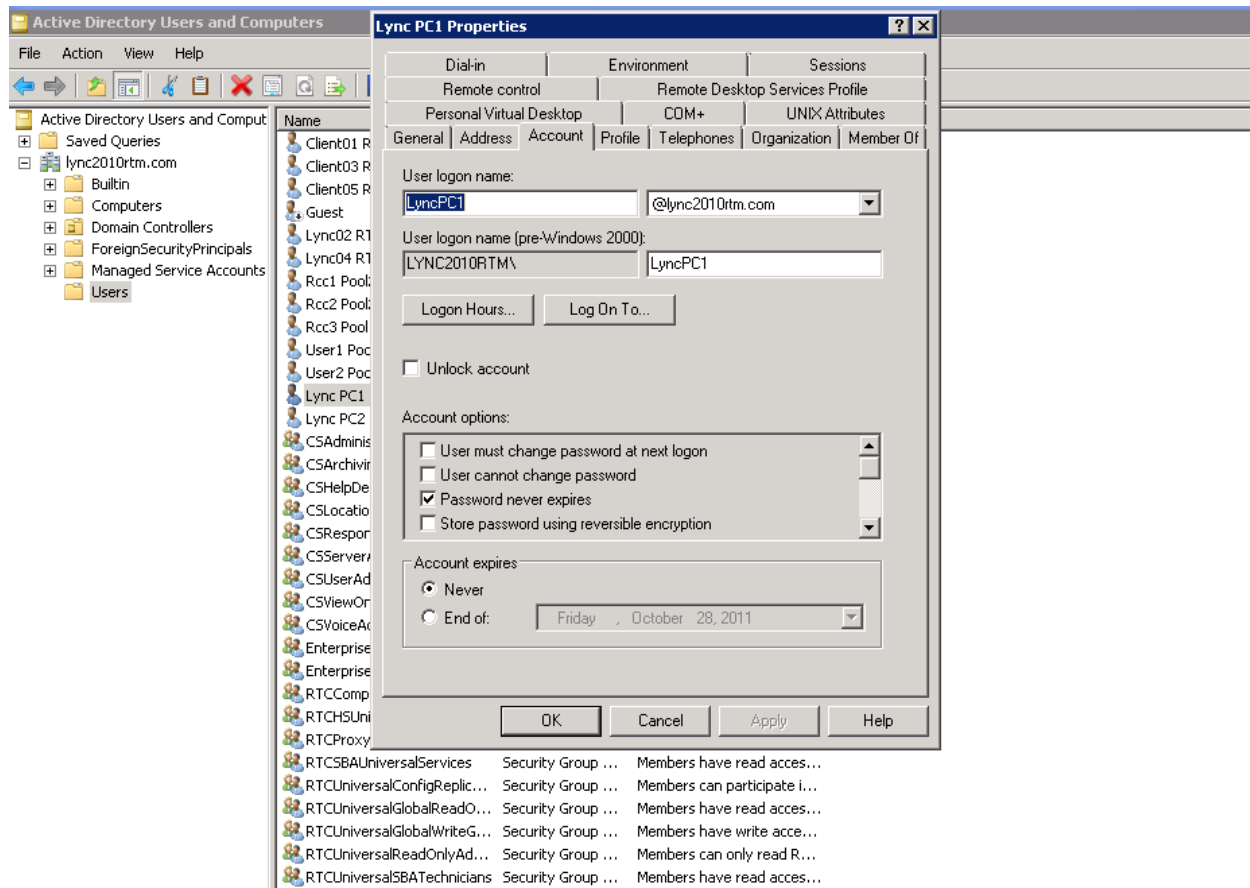
Active Directory Users and Computers

File Action View Help

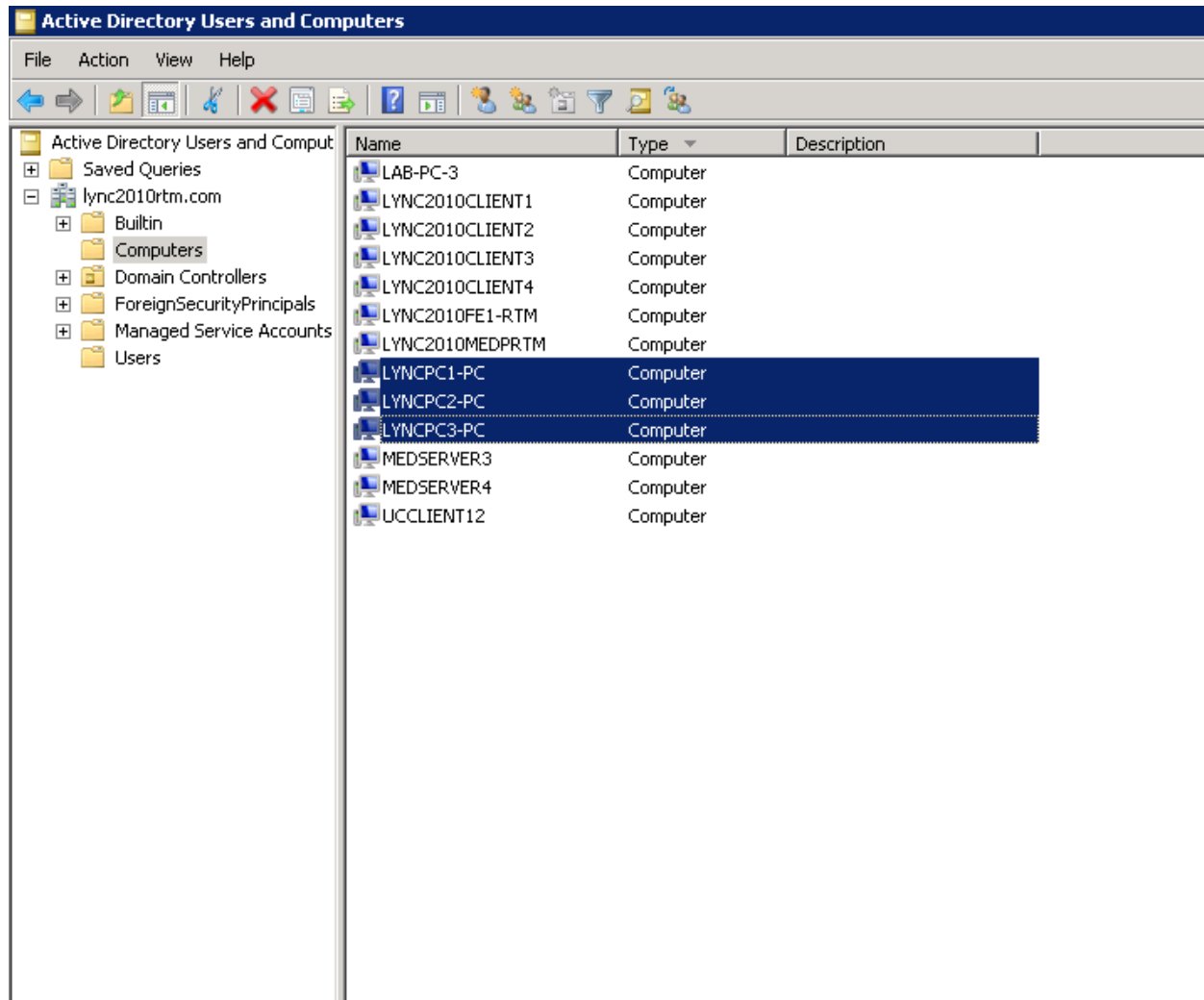
Active Directory Users and Computers

- Active Directory Users and Computers
 - Saved Queries
 - lync2010rtm.com
 - Builtin
 - Computers
 - Domain Controllers
 - ForeignSecurityPrincipals
 - Managed Service Accounts
 - Users**

Name	Type	Description
Client01 RTM	User	
Client03 RTM	User	
Client05 RTM	User	
Guest	User	Built-in account for guest ...
Lync02 RTM	User	
Lync04 RTM	User	
Rcc1 Pool2	User	
Rcc2 Pool2	User	
Rcc3 Pool1	User	
User1 Pool1	User	
User2 Pool1	User	
Lync PC1	User	
Lync PC2	User	
CSAdministrator	Security Group ...	Members of this group ca...
CSArchivingAdministrator	Security Group ...	Members of this group ca...
CSHelpDesk	Security Group ...	Members of this group ca...
CSLocationAdministrator	Security Group ...	Members of this group ha...
CSResponseGroupAdminis...	Security Group ...	Members of this group ca...
CSServerAdministrator	Security Group ...	Members of this group ca...
CSUserAdministrator	Security Group ...	Members of this group ca...
CSViewOnlyAdministrator	Security Group ...	Members of this group ca...
CSVoiceAdministrator	Security Group ...	Members of this group ca...
Enterprise Admins	Security Group ...	Designated administrators...
Enterprise Read-only Dom...	Security Group ...	Members of this group are...
RTCComponentUniversalS...	Security Group ...	Members can be used as ...
RTCHSUniversalServices	Security Group ...	Members can be used as ...



SME



Computers on the Active Directory running LYNC clients



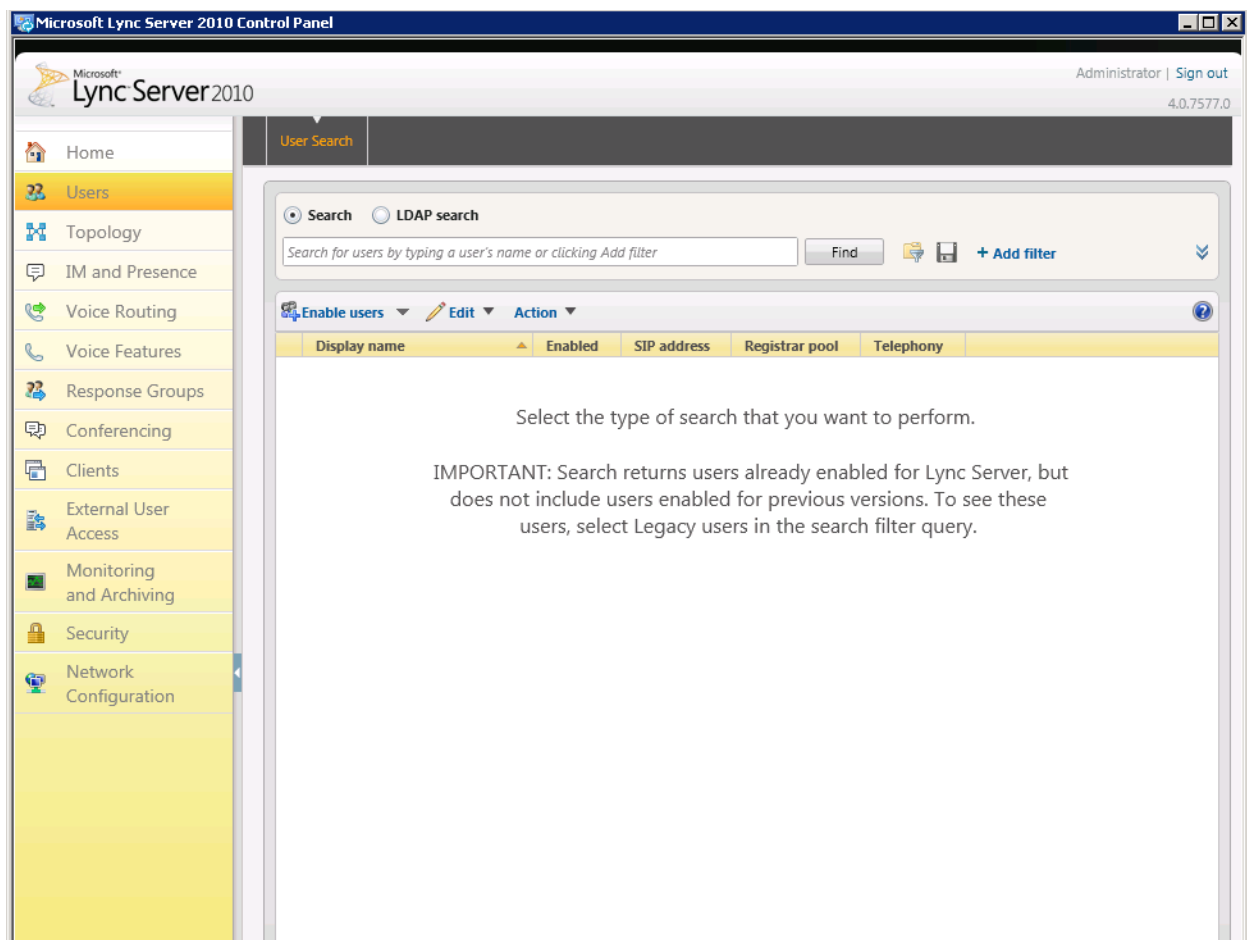
Lync Server 2010 Configuration

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 Administrator | [Sign out](#) 4.0.7577.0

User Search

Home Users Topology IM and Presence Voice Routing Voice Features Response Groups Conferencing Clients External User Access Monitoring and Archiving Security Network Configuration

Edit Lync Server User - Lync PC1

[Commit](#) [Cancel](#)

Display name:
Lync PC1

☒ **Enabled for Lync Server**

SIP address:
sip:LyncPC1 @ lync2010rtm.com

Registrar pool:
LYNC2010FE1-RTM.lync2010rtm.com

Telephony:
Enterprise Voice

Line URI:
tel:+14089333473

Dial plan policy:
SME_DialPlan [View...](#)

Voice policy:
SME_Policy [View...](#)

Conferencing policy:
<Automatic> [View...](#)

Client version policy:
<Automatic> [View...](#)

PIN policy:
<Automatic> [View...](#)

External access policy:
<Automatic> [View...](#)

Archiving policy:
<Automatic> [View...](#)

Location policy:



Lync Server 2010 4.0.7577.1

User Search

☒ Search ☐ LDAP search

Search for users by typing a user's name or clicking Add filter

Find

+ Add filter

Search results: 12

Enable users Edit Action

Display name	Enabled	SIP address	Registrar pool	Telephony
Client01 RTM	✓	sip:Client01@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Enterprise Voice
Client03 RTM	✓	sip:Client03@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Enterprise Voice
Client05 RTM	✓	sip:Client05@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Enterprise Voice
Lync PC1	✓	sip:LyncPC1@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Enterprise Voice
Lync PC2	✓	sip:LyncPC2@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Enterprise Voice
Lync02 RTM	✓	sip:Lync02@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Enterprise Voice
Lync04 RTM	✓	sip:Lync04@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Enterprise Voice
Rcc1 Pool2	✓	sip:Rcc1P2@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Remote call control
Rcc2 Pool2	✓	sip:Rcc2P2@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Remote call control
Rcc3 Pool1	✓	sip:Rcc3p1@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Remote call control
User1 Pool1	✓	sip:User1P1@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Remote call control
User2 Pool1	✓	sip:User2P1@lync2010rtm.com	LYNC2010FE1-RTM,lync2010rtm.com	Remote call control

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

Microsoft Lync Server 2010 Administrator | Sign out 4.0.

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 - SME_DialPlan

OK Cancel

Scope: User

Name: SME_DialPlan

Simple name: SME_DialPlan

Description: Use to Dial to SME

Dial-in conferencing region:

External access prefix:

Associated Normalization Rules

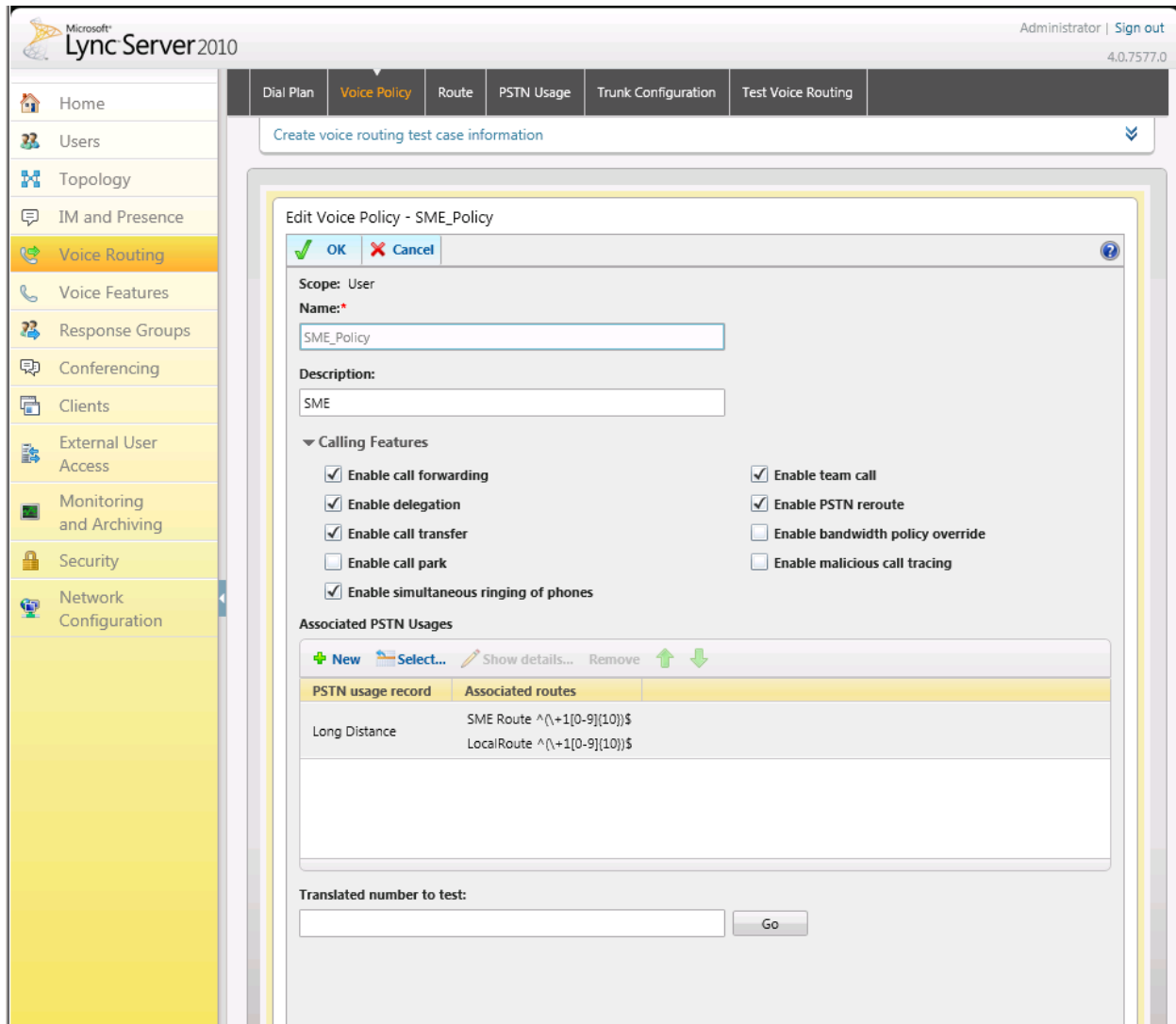
New Copy Paste Select... Show details... Remove

Normalization rule	State	Pattern to match	Translation pattern
4-digit-SME	Committed	^(6\d{3})\$	+1408933\$1
NANP	Committed	^(1\d{10})\$	+\$1

Dialed number to test:

Go

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



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 Policy - SME_Policy

OK Cancel

Scope: User

Name: SME_Policy

Description: SME

Calling Features

- ☒ Enable call forwarding
- ☒ Enable delegation
- ☒ Enable call transfer
- ☐ Enable call park
- ☒ Enable simultaneous ringing of phones
- ☒ Enable team call
- ☒ Enable PSTN reroute
- ☐ Enable bandwidth policy override
- ☐ Enable malicious call tracing

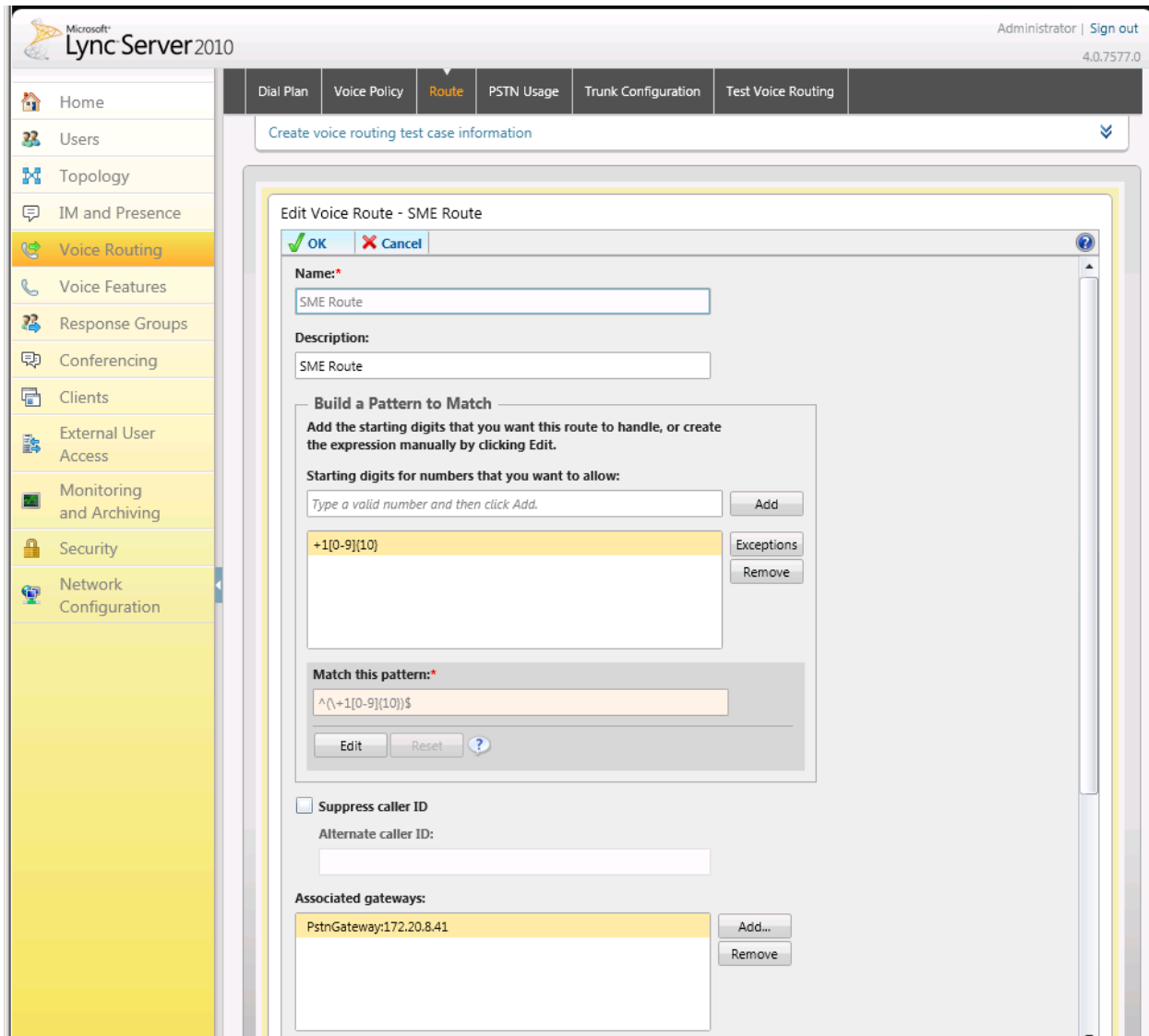
Associated PSTN Usages

New Select... Show details... Remove

PSTN usage record	Associated routes
Long Distance	SME Route ^(\+1[0-9]{10})\$ LocalRoute ^(\+1[0-9]{10})\$

Translated 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.



The screenshot shows the Microsoft Lync Server 2010 Administrator console. The left-hand navigation pane is expanded to 'Voice Routing'. The top navigation bar includes 'Dial Plan', 'Voice Policy', 'Route' (selected), 'PSTN Usage', 'Trunk Configuration', and 'Test Voice Routing'. A link 'Create voice routing test case information' is visible. The main content area displays the 'Edit Voice Route - SME Route' configuration window. This window has 'OK' and 'Cancel' buttons at the top. The 'Name' field is set to 'SME Route'. The 'Description' field is also 'SME Route'. Below this is a section titled 'Build a Pattern to Match' with instructions to add starting digits or create an expression manually. The 'Starting digits for numbers that you want to allow:' section shows a list with '+1[0-9]{10}' and buttons for 'Add', 'Exceptions', and 'Remove'. The 'Match this pattern:' section shows the regular expression '^(\+1[0-9]{10})\$' with 'Edit' and 'Reset' buttons. There is a checkbox for 'Suppress caller ID' and an 'Alternate caller ID' field. The 'Associated gateways:' section lists 'PstnGateway:172.20.8.41' with 'Add...' and 'Remove' buttons.

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

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



Microsoft Lync Server 2010 Administrator | Sign out 4.0.7577.0

Dial Plan Voice Policy Route PSTN Usage **Trunk Configuration** Test Voice Routing

Create voice routing test case information

Edit Trunk Configuration - Global

OK Cancel

Scope: Global

Name: Global

Maximum early dialogs supported: 20

Encryption support level: Not supported

☐ Enable media bypass

☒ Centralized media processing

☐ Enable refer support

Associated Translation Rules

New Copy Paste Select... Show details... Remove

Translation rule	State	Pattern to match	Translation pattern
Pass_ALL	Committed	^(\\+?\\d*)\$	\$1



Lync 2010 Server Management Shell Commands

Lync Server/Client Address Book Updating

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

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

Update-CsAddressBook

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

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

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

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

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

Lync 2010 Server draining mode

```
Stop-CsWindowsService -Graceful rtcmedsrv
```

Some Lync2010 useful commands

```
Get-CsTrunkConfiguration
```

```
Set-CsTrunkConfiguration
```

```
Get-CsMediaConfiguration
```

```
Set-CsMediaConfiguration
```

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

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



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

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

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

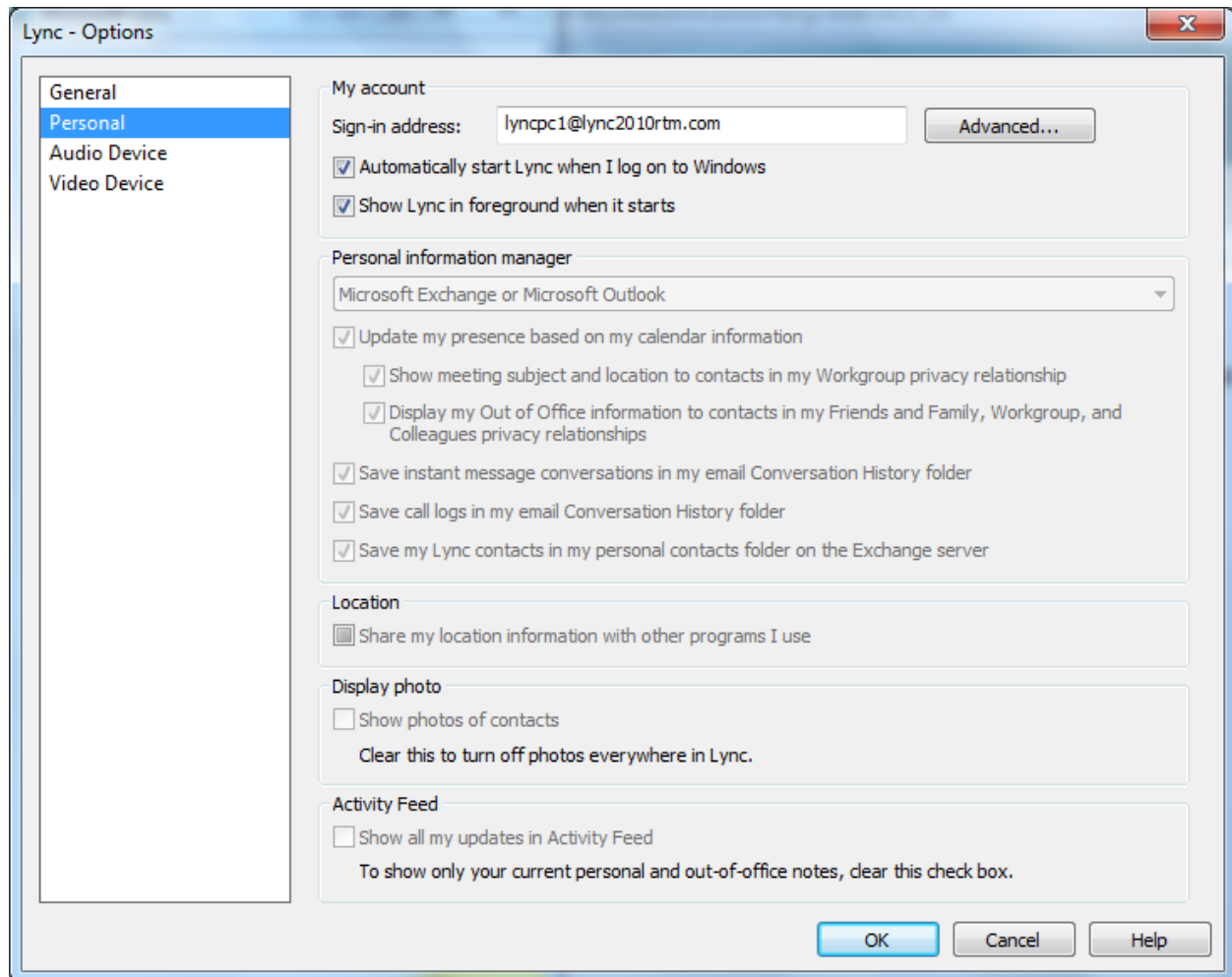
Lync2010 - enable music on hold:

set-csclientpolicy -EnableClientMusicOnHold \$TRUE

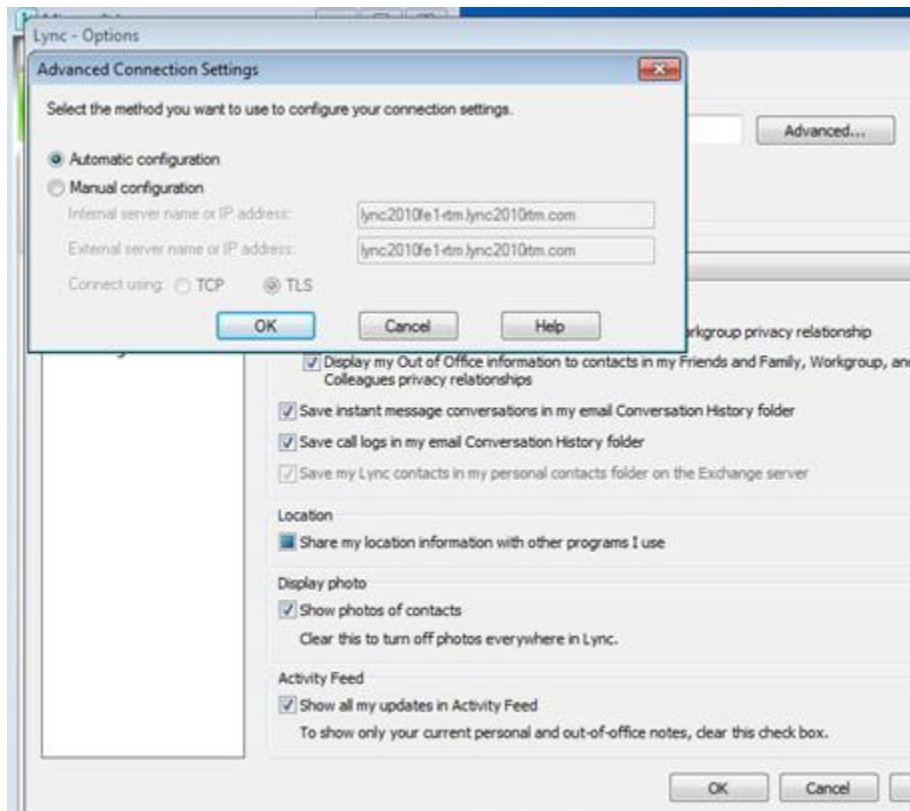
Microsoft LYNC 2010 Enterprise Voice Client Configuration

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





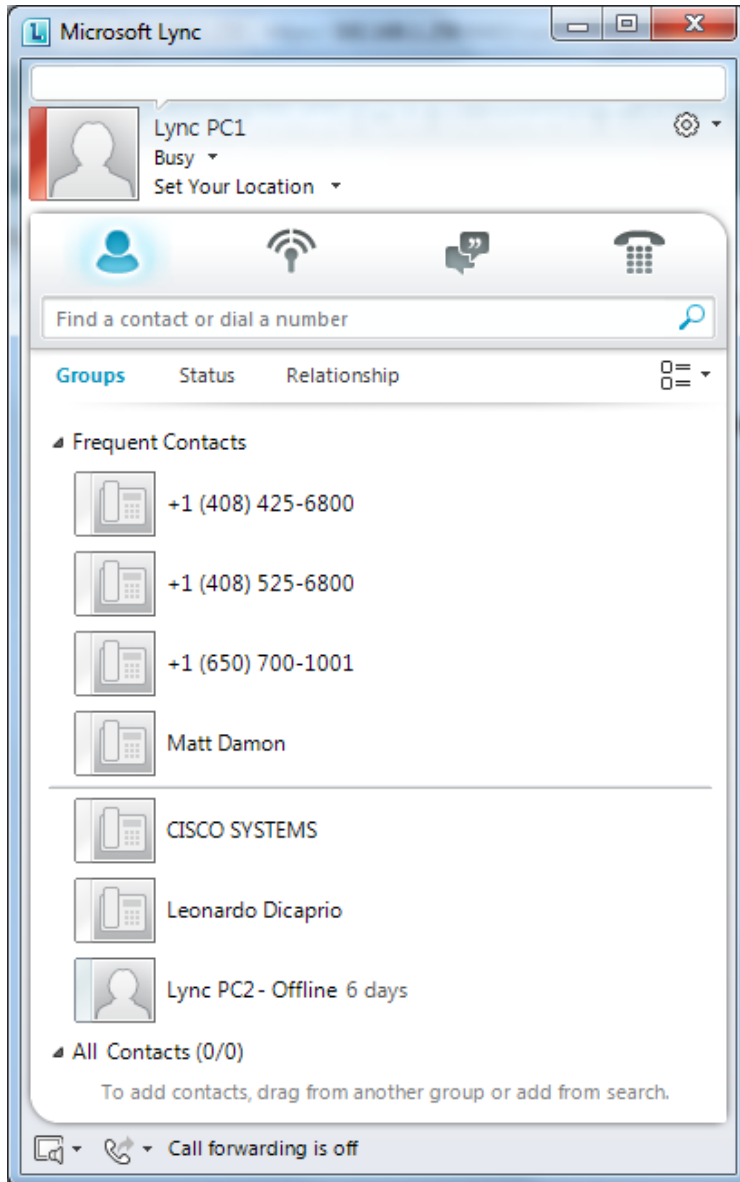
Click Advanced button to select the Advanced Connection Settings.



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

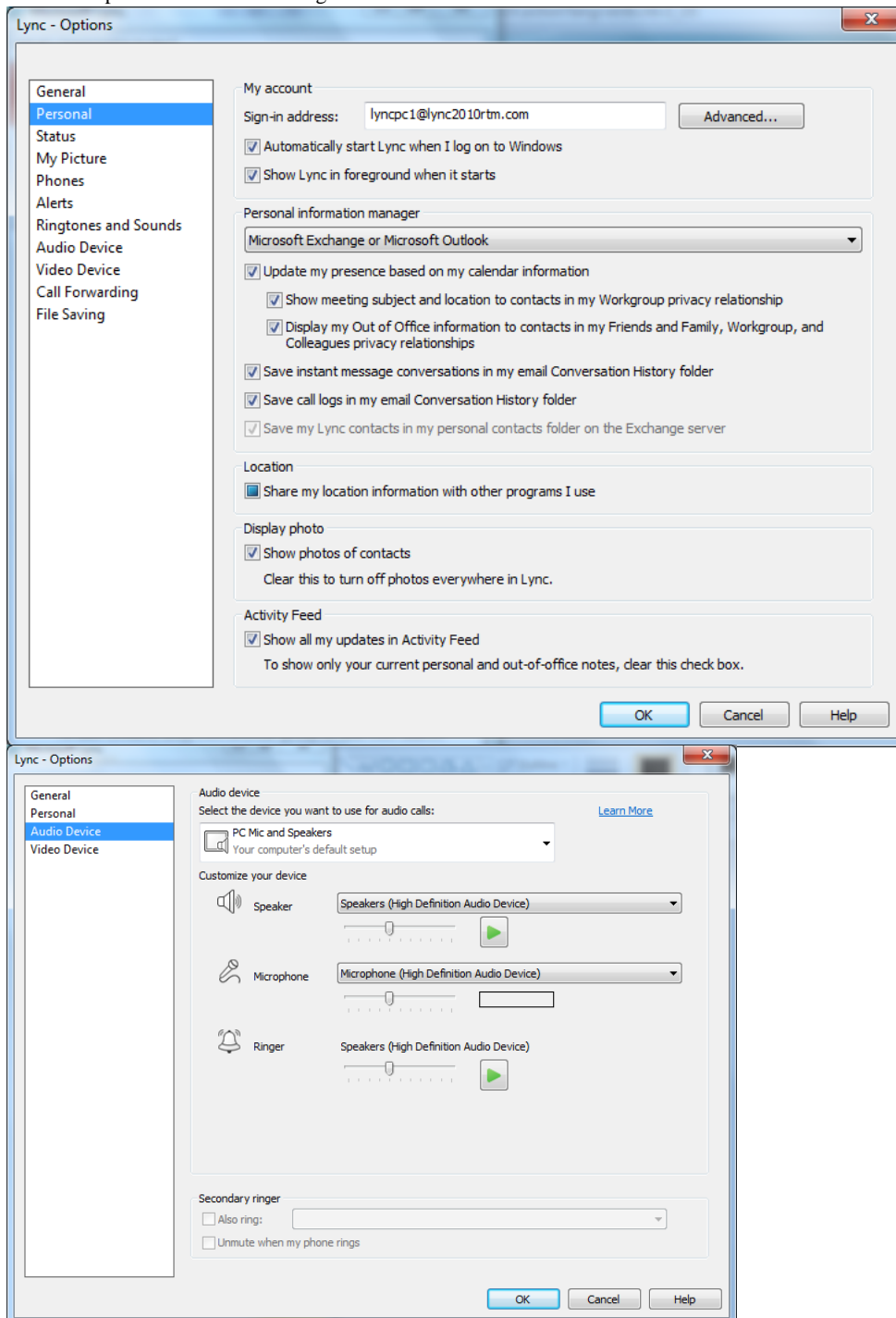


After signing add contacts



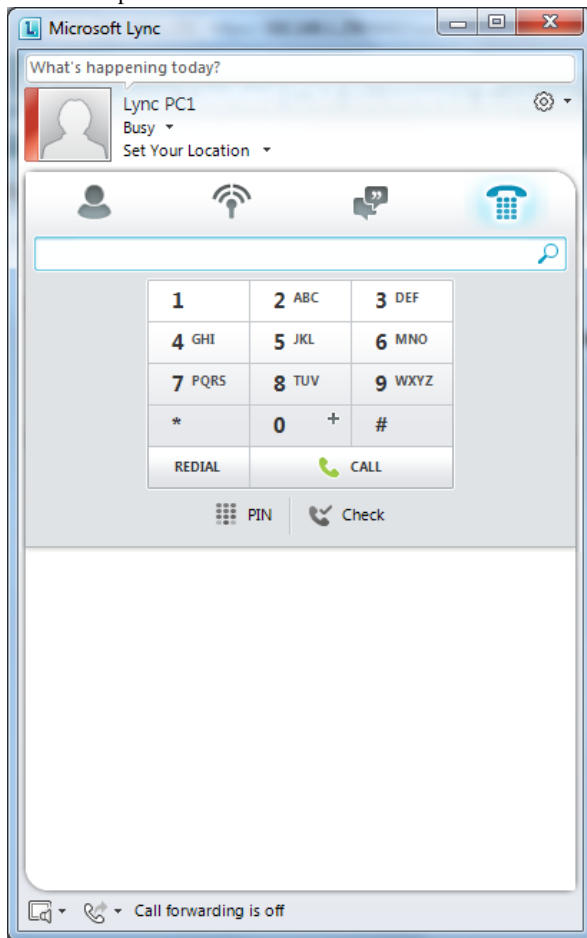


Personal Options and Audio Settings



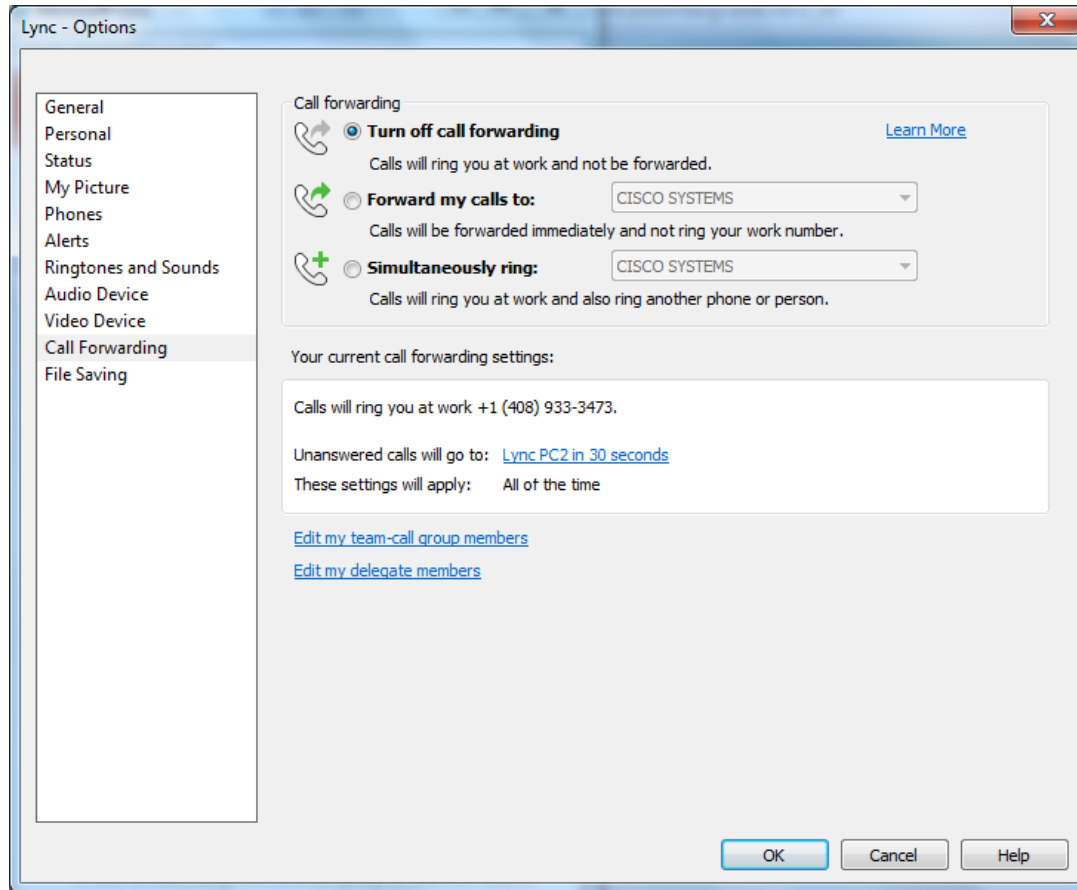


LYNC Dialpad





Call Forwarding Configurations



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



Cisco Unified Communications Manager- Session Manager Edition Configurations

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

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

Cisco Unified Communications Manager- Session Manager Edition – Software release

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM
admin | Search Documentation

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Cisco Unified CM Administration
System version: 8.5.1.12900-7

Licensing Warnings:
System is operating on Demo licenses. Please upload relevant license files.
Please visit the License Report Page for more details.

VMware Installation: 2 vCPU Intel(R) Xeon(R) CPU E5504 @ 2.00GHz, disk 1: 200Gbytes, 5120Mbytes RAM

Last Successful Login: Sep 20, 2011 12:44:11 PM

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This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

For Cisco Technical Support please visit our [Technical Support](#) web site.



Cisco Unified Communications Manager-Session Manager Edition-Region Configurations

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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admin | Search

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Region Configuration Related Li

Save Delete Reset Apply Config Add New

Region Information

Name*

Region Relationships

Region	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)
Default	64 kbps (G.722, G.711)	384
G711-CUCM	64 kbps (G.722, G.711)	384
g729	8 kbps (G.729)	384

NOTE: Regions(s) not displayed Use System Default Use System Default

Modify Relationship to other Regions

Regions	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)
Default G711-CUCM g729	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps

*- indicates required item.

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



Cisco Unified Communications Manager – Session Manager Edition – Device Pools

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | Search Documentation | About | Log Out

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

Find and List Device Pools

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

Status
3 records found

Device Pool (1 - 3 of 3) Rows per Page 50

Find Device Pool where: Device Pool Name begins with Find Clear Filter

<input type="checkbox"/>	Name ^	Cisco Unified CM Group	Region	Date/Time Group	Copy
<input type="checkbox"/>	Default	Default	Default	CMLocal	
<input type="checkbox"/>	G729-DevicePool	Default	g729	CMLocal	
<input type="checkbox"/>	Internal-DP	Default	G711-CUCM	CMLocal	

Add New | Select All | Clear All | Delete Selected



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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Device Pool Configuration

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Device Pool Information
Device Pool: Internal-DP (7 members**)

Device Pool Settings
Device Pool Name* Internal-DP
Cisco Unified Communications Manager Group* Default ▾
Calling Search Space for Auto-registration < None > ▾
Adjunct CSS < None > ▾
Reverted Call Focus Priority Default ▾
Local Route Group < None > ▾
Intercompany Media Services Enrolled Group < None > ▾

Roaming Sensitive Settings
Date/Time Group* CMLocal ▾
Region* G711-CUCM ▾
Media Resource Group List Internal-MRGL ▾
Location Hub_None ▾
Network Locale < None > ▾
SRST Reference* Disable ▾
Connection Monitor Duration***
Single Button Barge* Default ▾
Join Across Lines* Default ▾
Physical Location < None > ▾
Device Mobility Group < None > ▾

Device Mobility Related Information****
Device Mobility Calling Search Space < None > ▾
AAR Calling Search Space < None > ▾

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



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

Device Pool ConfigurationRelated Link

Save Delete Copy Reset Apply Config Add New

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 in the field is empty in which case there is no prefix assigned.

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text"/>	< None >
International Number	<input type="text" value="Default"/>	<input type="text"/>	< None >
Unknown Number	<input type="text" value="Default"/>	<input type="text"/>	< None >
Subscriber Number	<input type="text" value="Default"/>	<input type="text"/>	< 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 in the field is empty in which case there is no prefix assigned.

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	< None >
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	< None >
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	< None >
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	< None >

Connected Party Settings
Connected Party Transformation CSS < None >



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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Nav

admin

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

Device Pool Configuration

Save Delete Copy Reset Apply Config Add New

Device Pool Settings
Device Pool Name* G729-DevicePool
Cisco Unified Communications Manager Group* Default ▾
Calling Search Space for Auto-registration Internal-CSS ▾
Adjunct CSS < None > ▾
Reverted Call Focus Priority Default ▾
Local Route Group < None > ▾
Intercompany Media Services Enrolled Group < None > ▾

Roaming Sensitive Settings
Date/Time Group* CMLocal ▾
Region* g729 ▾
Media Resource Group List < None > ▾
Location Hub_None ▾
Network Locale < None > ▾
SRST Reference* Disable ▾
Connection Monitor Duration***
Single Button Barge* Default ▾
Join Across Lines* Default ▾
Physical Location < None > ▾
Device Mobility Group < None > ▾

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



Device Pool Configuration

Related Li

Save

Delete

Copy

Reset

Apply Config

Add New

Geolocation Filter

< None >

Call Routing Information

Incoming Calling Party Settings

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

Clear Prefix Settings

Default Prefix Settings

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

Incoming Called Party Settings

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

Clear Prefix Settings

Default Prefix Settings

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

Connected Party Settings

Connected Party Transformation CSS

< None >

Save

Delete

Copy

Reset


Apply Config

Add New

*. indicates required item



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







**Cisco Unified CM Administration**
For Cisco Unified Communications Solutions


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admin | [Search Documentation](#)



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Find and List Media Termination Points

 Add New  Select All  Clear All  Delete Selected  Reset Selected  Apply Config to Selected

Status
 3 records found

Media Termination Point (1 - 3 of 3) Row:

Find Media Termination Point where begins with  

<input type="checkbox"/>		Name ^	Description	Device Pool	Status	IP Address	
<input type="checkbox"/>		G729-Soft		Internal-DP	Registered with 172.20.8.41	172.20.8.58	
<input type="checkbox"/>		IOS-MTP-SJC		Internal-DP	Registered with 172.20.8.41	172.20.8.58	
<input type="checkbox"/>		MTP_2	MTP_SME851	Internal-DP	Registered with 172.20.8.41	172.20.8.41	N



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Nav

admin

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

Media Termination Point Configuration

Save Reset Apply Config

Status
Status: Ready

Media Termination Point Information
Registration Registered with Cisco Unified Communications Manager 172.20.8.41
IP Address 172.20.8.41
IPv6 Address 0000:0000:0000:0000:0000:0000:0000:0000
Media Termination Point Type* Cisco Media Termination Point Software
Host Server* 172.20.8.41
Media Termination Point Name* MTP_2
Description MTP_SME851
Device Pool* Internal-DP
☒ Trusted Relay Point

Save Reset Apply Config

i *- indicates required item.



Cisco Unified Communications Manager – Session manager Edition – Transcoder Configuration

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Find and List Transcoders

+ Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected

Status
1 records found

Transcoder (1 - 1 of 1) Rows per Page

Find Transcoder where Name ▾ begins with ▾ Find Clear Filter

<input type="checkbox"/>	Name ^	Description	Device Pool	Status	IP Address
<input type="checkbox"/>	XCODE-SJC	XCODE-SJC	Default	Registered with 172.20.8.41	172.20.8.58

Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation | Cisco Unified CM Adminis

admin

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

Transcoder Configuration

Save Delete Copy Reset Apply Config Add New

Transcoder Information

Transcoder: XCODE-SJC (XCODE-SJC)

Registration Registered with Cisco Unified Communications Manager 172.20.8.41

IP Address 172.20.8.58

IPv6 Address 0000:0000:0000:0000:0000:0000:0000:0000

IOS Transcoder Info

Transcoder Type* Cisco IOS Enhanced Media Termination Point

Description XCODE-SJC

Device Name* XCODE-SJC

Device Pool* Default [View Details](#)

Common Device Configuration < None > [View Details](#)

Special Load Information Leave blank to use default

☒ Trusted Relay Point

Save Delete Copy Reset Apply Config Add New

*- indicates required item.



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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation
admin

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

Media Resource Group Configuration

Save Delete Copy Add New

Status
Status: Ready

Media Resource Group Status
Media Resource Group: SME-MRG (used by 5 devices)

Media Resource Group Information
Name* SME-MRG
Description

Devices for this Group
Available Media Resources**
G729-Soft
CFB_2 (CFB)
Selected Media Resources*
ANN_2 (ANN)
IOS-MTP-SJC (MTP)
MOH_2 (MOH)
MTP_2 (MTP)
XCODE-SJC
☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Save Delete Copy Add New

i *- indicates required item.
i **Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)



MRG for LYNC MRGL

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Media Resource Group Configuration

Save Delete Copy Add New

Status
Status: Ready

Media Resource Group Status
Media Resource Group: LYNC-MRG (used by 1 devices)

Media Resource Group Information
Name* LYNC-MRG
Description

Devices for this Group
Available Media Resources**
ANN_2
CFB_2
G729-Soft
IOS-MTP-SJC
MOH_2
Selected Media Resources*
MTP_2 (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)



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

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". A secondary navigation bar lists various system components like System, Call Routing, Media Resources, etc. The main content area is titled "Find and List Media Resource Group Lists". It features a toolbar with buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below this, a status bar indicates "2 records found". The main table, titled "Media Resource Group List (1 - 2 of 2)", has a search bar and a table with two columns: a checkbox column and a "Name" column. The table lists two entries: "Internal-MRGL" and "LYNC-MRGL", both with checkboxes in the first column and document icons in the second. At the bottom of the table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

	Name ^	
<input type="checkbox"/>	Internal-MRGL	
<input type="checkbox"/>	LYNC-MRGL	

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Media Resource Group List Configuration Re

Save Delete Copy Add New

Status

Status: Ready

Media Resource Group List Status

Media Resource Group List: Internal-MRGL (used by 5 devices)

Media Resource Group List Information

Name*

Media Resource Groups for this List

Available Media Resource Groups

LYNC-MRG

Selected Media Resource Groups

SME-MRG

Save Delete Copy Add New

*- indicates required item.



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

Cisco Unified CM Administration
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Media Resource Group List Configuration

Save Delete Copy Add New

— Status —
 Status: Ready

— Media Resource Group List Status —
Media Resource Group List: LYNC-MRGL (used by 1 devices)

— Media Resource Group List Information —
Name*

— Media Resource Groups for this List —
Available Media Resource Groups

SME-MRG

▼ ▲

Selected Media Resource Groups

LYNC-MRG


▼ ▲

Save Delete Copy Add New

*- indicates required item.



Cisco Unified Communications Manager –Session Manager Edition – Route Patterns





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For Cisco Unified Communications Solutions


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Find and List Route Patterns

 Add New  Select All  Clear All  Delete Selected

Status
 14 records found

Route Patterns (1 - 14 of 14) Row

Find Route Patterns where begins with  

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



Route Pattern to the Microsoft LYNC 2010 server

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Route Pattern Configuration

Save Delete Copy Add New

Status: Ready

- Pattern Definition -

Route Pattern*

40893334XX

Route Partition

LYNC-PT ▾

Description

Numbering Plan

-- Not Selected -- ▾

Route Filter

< None > ▾

MLPP Precedence*

Default ▾

Resource Priority Namespace Network Domain

< None > ▾

Route Class*

Default ▾

Gateway/Route List*

CUCM-SIP-Trunk ▾ [\(Edit\)](#)

Route Option

☒ Route this pattern

☐ Block this pattern

No Error ▾

Call Classification*

OffNet ▾

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level*

0

☐ Require Client Matter Code

- Calling Party Transformations -

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

+1

Calling Line ID Presentation*

Default ▾

Calling Name Presentation*

Default ▾

Calling Party Number Type*

Cisco CallManager ▾

Calling Party Numbering Plan*

Cisco CallManager ▾



Route Pattern Configuration

Save Delete Copy Add New

☐ Require Client Matter Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Calling Name Presentation*
Calling Party Number Type*
Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*
Connected Name Presentation*

Called Party Transformations

Discard Digits
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol
Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

Save Delete Copy Add New

*- indicates required item.



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

Route Pattern Configuration

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition
Route Pattern* 4089336XXX
Route Partition CUCM-PT
Description
Numbering Plan -- Not Selected --
Route Filter < None >
MLPP Precedence* Default
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* CUCM-SIP-Trunk [\(Edit\)](#)
Route Option
☒ Route this pattern
☐ Block this pattern No Error
Call Classification* OffNet
☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level* 0
☐ Require Client Matter Code

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



Route Pattern Configuration

Save Delete Copy Add New

☐ Require Forced Authorization Code

Authorization Level*

☐ 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*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value=" < Not Exist >"/>	<input type="text"/>



Route Pattern to route calls to the LYNC server

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Route Pattern Configuration Relat

Save Delete Copy Add New

— Status —
 Status: Ready

— Pattern Definition —
Route Pattern*
Route Partition
Description
Numbering Plan
Route Filter
MLPP Precedence*
Resource Priority Namespace Network Domain
Route Class*
Gateway/Route List* [\(Edit\)](#)
Route Option
☒ Route this pattern
☐ Block this pattern
Call Classification*
☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level*
☐ 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*
Calling Name Presentation*
Calling Party Number Type*
Calling Party Numbering Plan*



☐ 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*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

· Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

· Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

· ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value=" < Not Exist >"/>	<input type="text"/>

*- indicates required item.



Route Pattern to Service Provider

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Route Pattern Configuration

Rela

Save

Delete

Copy

Add New

Status

Status: Ready

Pattern Definition

Route Pattern*

9.@

Route Partition

CUBE-PSTN-PT

Description

Numbering Plan*

NANP

Route Filter

< None >

MLPP Precedence*

Default

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Gateway/Route List*

PSTN-Access-RL

Route Option

Route this pattern

Block this pattern

No Error

Call Classification*

OffNet

Allow Device Override

Provide Outside Dial Tone

Allow Overlap Sending

Urgent Priority

Require Forced Authorization Code

Authorization Level*

0

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Default

Calling Name Presentation*

Default

Calling Party Number Type*

Cisco CallManager

Calling Party Numbering Plan*

Cisco CallManager



Route Pattern Configuration

Save

Delete

Copy

Add New

☐ Require Client Matter Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Default

Calling Name Presentation*

Default

Calling Party Number Type*

Cisco CallManager

Calling Party Numbering Plan*

Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*

Default

Connected Name Presentation*

Default

Called Party Transformations

Discard Digits

PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Cisco CallManager

Called Party Numbering Plan*

Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol

-- Not Selected --

Carrier Identification Code

Network Service

-- Not Selected --

Service Parameter Name

< Not Exist >

Service Parameter Value

Save

Delete

Copy

Add New

* - indicates required item.



International Route Pattern

Cisco Unified CM Administration
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Route Pattern Configuration

Save Delete Copy Add New

Status
Status: Ready

Pattern Definition
Route Pattern* 9.011!
Route Partition CUBE-PSTN-PT ▾
Description
Numbering Plan -- Not Selected -- ▾
Route Filter < None > ▾
MLPP Precedence* Default ▾
Resource Priority Namespace Network Domain < None > ▾
Route Class* Default ▾
Gateway/Route List* PSTN-Access-RL ▾ (Edit)
Route Option
☒ Route this pattern
☐ Block this pattern No Error ▾
Call Classification* OffNet ▾
☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level* 0
☐ Require Client Matter Code

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



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Route Pattern Configuration

Save Delete Copy Add New

☐ Require forced Authorization Code

Authorization Level*

☐ 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*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>



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

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

Route List Configuration Relat

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

Route List Information

Registration

Registered with Cisco Unified Communications Manager 172.20.8.41

IP Address

172.20.8.41

☒ Device is trusted

Name*

PSTN-Access-RL

Description

Cisco Unified Communications Manager Group*

Default ▾

☒ Enable this Route List (change effective on Save; no reset required)

☒ Run On All Active Unified CM Nodes

Route List Member Information

Selected Groups**

PSTN-Access

▼

▲

Add Route Group

Removed Groups***

▼

▲

Route List Details

[PSTN-Access](#)

Save Delete Copy Reset Apply Config Add New

*- indicates required item.

**Ordered by highest priority

***Will be removed from Route List when you click Save

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SME Route Group for PSTN Trunk Access

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Route Group Configuration Related

Save Delete Add New

Status: Ready

Route Group Information

Route Group Name*

Distribution Algorithm*

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains

Available Devices**

CUBE-Trunk

Port(s)

Current Route Group Members

Selected Devices (ordered by priority)*

CUBE-Trunk (All Ports)


Removed Devices***

Route Group Members

CUBE-Trunk



Cisco UCM-SME SIP Trunks:






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

Find and List Trunks

 Add New  Select All  Clear All  Delete Selected  Reset Selected

Status
 8 records found

Trunks (1 - 8 of 8) Rows per Page 50

Find Trunks where Device Name ▾ begins with ▾  

<input type="checkbox"/>	Name ▲	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Security Profile
<input type="checkbox"/>	 CUBE-Trunk		Internal-CSS	Internal-DP			PSTN-Access	1	SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 CUCM-SIP-Trunk		PSTN-Access	Internal-DP	91.4089336XXX	CUCM-PT			SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 CUCM-SIP-Trunk		PSTN-Access	Internal-DP	4089336XXX	CUCM-PT			SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 CUCM-SIP-Trunk		PSTN-Access	Internal-DP	44XX	CUCM-PT			SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 CUCM-SIP-Trunk		PSTN-Access	Internal-DP	51XX	CUCM-PT			SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 LYNC-Trunk		PSTN-Access	Internal-DP	91.40893334XX	LYNC-PT			SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 LYNC-Trunk		PSTN-Access	Internal-DP	347X	LYNC-PT			SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 LYNC-Trunk		PSTN-Access	Internal-DP	40893334XX	LYNC-PT			SIP Trunk	Non Secure SIP Trunk Profile



SME SIP trunk to Service Provider via ASR CUBE

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Trunk Configuration

Related

Save Delete Reset Add New

— Device Information —

Product: SIP Trunk

Device Protocol: SIP

Trunk Service Type: None(Default)

Device Name*: CUBE-Trunk

Description:

Device Pool*: Internal-DP

Common Device Configuration: < None >

Call Classification*: Use System Default

Media Resource Group List: Internal-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



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Add New

☒ Asserted-Identity

Asserted-Type*

PP1

SIP Privacy*

Default

Inbound Calls

Significant Digits*

10

Connected Line ID Presentation*

Allowed

Connected Name Presentation*

Allowed

Calling Search Space

Internal-CSS

AAR Calling Search Space

< None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the 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
Incoming Number	<div>Default</div>	<div>0</div>	<div>< None ></div>	<div><input checked="" type="checkbox"/></div>

Connected Party Settings

Connected Party Transformation CSS

< None >

☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS

< None >

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

< None >

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Originator

Calling Line ID Presentation*

Allowed



Trunk Configuration

Save Delete Reset Add New

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection* Originator

Calling Line ID Presentation* Allowed

Calling Name Presentation* Allowed

Caller ID DN

Caller Name

☒ Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	172.20.110.151		5060

MTP Preferred Originating Codec* 711ulaw

Presence Group* Standard Presence group

SIP Trunk Security Profile* Non Secure SIP Trunk Profile

Rerouting Calling Search Space < None >

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

SUBSCRIBE Calling Search Space < None >

SIP Profile* SME-Profile22

DTMF Signaling Method* RFC 2833

Normalization Script

Normalization Script < None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		



Cisco UCM-SME SIP trunk to Cisco UCM

Cisco Unified CM Administration
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Trunk Configuration Related

Save Delete Reset Add New

Status
Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	CUCM-SIP-Trunk
Description	
Device Pool*	Internal-DP
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	Internal-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
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> 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



Trunk Configuration

Related I

Save

Delete

Reset

Add New

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type*

PAI

SIP Privacy*

None

Inbound Calls

Significant Digits*

All

Connected Line ID Presentation*

Allowed

Connected Name Presentation*

Allowed

Calling Search Space

PSTN-Access

AAR Calling Search Space

< None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
Incoming Number	Default	0	< None >

Connected Party Settings

Connected Party Transformation CSS

< None >

☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS

< None >

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

< None >

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*



Trunk ConfigurationRelated

Save

Delete

Reset

Add New

Calling Line ID Presentation*

Allowed

Calling Name Presentation*

Allowed

Caller ID DN

Caller Name

☒ Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination

☐ Destination Address is an SRV

Destination Address

Destination Address IPv6

Destination Port

1*

172.20.8.42

5060

MTP Preferred Originating Codec*

G729/G729a

Presence Group*

Standard Presence group

SIP Trunk Security Profile*

Non Secure SIP Trunk Profile

Rerouting Calling Search Space

< None >

Out-Of-Dialog Refer Calling Search Space

< None >

SUBSCRIBE Calling Search Space

< None >

SIP Profile*

SME SIP Profile

DTMF Signaling Method*

RFC 2833

Normalization Script

Normalization Script

< None >

☐ Enable Trace

Parameter Name

Parameter Value

1

Geolocation Configuration

Geolocation

< None >

Geolocation Filter

< None >

☐ Send Geolocation Information



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

Trunk Configuration		Related
<div>Save Delete Reset Add New</div>		
Device Information		
Product:	SIP Trunk	
Device Protocol:	SIP	
Trunk Service Type	None(Default)	
Device Name*	<input type="text" value="LYNC-Trunk"/>	
Description	<input type="text"/>	
Device Pool*	<input type="text" value="Internal-DP"/>	
Common Device Configuration	<input type="text" value=" < None >"/>	
Call Classification*	<input type="text" value="Use System Default"/>	
Media Resource Group List	<input type="text" value="LYNC-MRGL"/>	
Location*	<input type="text" value="Hub_None"/>	
AAR Group	<input type="text" value=" < None >"/>	
Tunneled Protocol*	<input type="text" value="None"/>	
QSIG Variant*	<input type="text" value="No Changes"/>	
ASN.1 ROSE OID Encoding*	<input type="text" value="No Changes"/>	
Packet Capture Mode*	<input type="text" value="None"/>	
Packet Capture Duration	<input type="text" value="0"/>	
<input checked="" type="checkbox"/> Media Termination Point Required		
<input checked="" type="checkbox"/> Retry Video Call as Audio		
<input type="checkbox"/> Path Replacement Support		
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name		
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU		
<input type="checkbox"/> Unattended Port		
<input type="checkbox"/> 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*	<input type="text" value="When using both sRTP and TLS"/>	
Route Class Signaling Enabled*	<input type="text" value="Default"/>	
Use Trusted Relay Point*	<input type="text" value="Default"/>	
<input checked="" type="checkbox"/> PSTN Access		
<input type="checkbox"/> Run On All Active Unified CM Nodes		



☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type*

SIP Privacy*

Inbound Calls

Significant Digits*

Connected Line ID Presentation*

Connected Name Presentation*

Calling Search Space

AAR Calling Search Space

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Number Type	Prefix	Strip Digits	Calling Search Space	
Incoming Number	<input type="text" value=""/>	<input type="text" value="2"/>	<input type="text" value=" < None >"/>	<input type="checkbox"/>

Connected Party Settings

Connected Party Transformation CSS

☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*



Trunk Configuration

Related L

Save

Delete

Reset

Add New

Calling Party Transformation CSS

< None >

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Originator

Calling Line ID Presentation*

Allowed

Calling Name Presentation*

Allowed

Caller ID DN

Caller Name

☒ Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination

☐ Destination Address is an SRV

Destination Address

Destination Address IPv6

Destination Port

1*

172.20.117.152

5068

MTP Preferred Originating Codec*

711ulaw

Presence Group*

Standard Presence group

SIP Trunk Security Profile*

Non Secure SIP Trunk Profile

Rerouting Calling Search Space

< None >

Out-Of-Dialog Refer Calling Search Space

< None >

SUBSCRIBE Calling Search Space

< None >

SIP Profile*

SME-Profile22

DTMF Signaling Method*

RFC 2833

Normalization Script

< None >

☐ Enable Trace

Parameter Name

Parameter Value


1

Configuration Summary



Configuring the Cisco Unified Communications Manager

Cisco UCM Software release

**Cisco Unified CM Administration**
For Cisco Unified Communications Solutions

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Cisco Unified CM Administration

System version: 8.5.1.12900-7

Licensing Warnings:
System is operating on Demo licenses. Please upload relevant license files.
Please visit the License Report Page for more details.

VMware Installation: 2 vCPU Intel(R) Xeon(R) CPU 5140 @ 2.33GHz, disk 1: 100Gbytes, 4096Mbytes RAM

Last Successful Logon: Sep 19, 2011 1:47:38 PM

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A summary of U.S. laws governing Cisco cryptographic products may be found at our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

For Cisco Technical Support please visit our [Technical Support](#) web site.



Cisco UCM Regions Configurations





**Cisco Unified CM Administration**
For Cisco Unified Communications Solutions


Navigation Cisco Unified CM Administration

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Find and List Regions

 Add New  Select All  Clear All  Delete Selected

Status
 3 records found

Regions (1 - 3 of 3) Rows per Page 50

Find Regions where Name begins with  

	Name ^
<input type="checkbox"/>	Default
<input type="checkbox"/>	G711-Region
<input type="checkbox"/>	G729-Region



Cisco UCM Regions Configurations – G711 Region

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Region Configuration Related L

Save Delete Reset Apply Config Add New

Region Information

Name*

Region Relationships

Region	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)
Default	64 kbps (G.722, G.711)	384
G711-Region	64 kbps (G.722, G.711)	384
G729-Region	8 kbps (G.729)	384

NOTE: Regions(s) not displayed Use System Default Use System Default

Modify Relationship to other Regions

Regions	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)
<div>Default G711-Region G729-Region</div>	<div>Keep Current Setting ▾</div>	<div><input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps</div>

Save Delete Reset Apply Config Add New

*- indicates required item.



Cisco UCM Region Configurations – G729 Region

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Region Configuration

Related Links

Save

Delete

Reset

Apply Config

Add New

Region Information

Name*

Region Relationships

Region	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)
Default	8 kbps (G.729)	384
G711-Region	8 kbps (G.729)	384
G729-Region	8 kbps (G.729)	384

NOTE: Region(s) not displayed

Use System Default

Use System Default

Use System Default

Modify Relationship to other Regions

Regions	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)
<div>Default G711-Region G729-Region</div>	<div>Keep Current Setting ▾</div>	<div><div><input checked="" type="radio"/> Keep Current Setting</div><div><input type="radio"/> Use System Default</div><div><input type="radio"/> None</div><div><input type="text" value=""/> kbps</div></div>

Save

Delete

Reset

Apply Config

Add New

i

*. indicates required item.



Cisco UCM Device Pool

System

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Find and List Device Pools

Add New

Select All

Clear All

Delete Selected

Status

3 records found

Device Pool

(1 - 3 of 3)

Rows per Page 50

Find Device Pool where

Device Pool Name

begins with

Find

Clear Filter

	Name ^	Cisco Unified CM Group	Region	Date/Time Group	Cc
<input type="checkbox"/>	Default	Default	Default	CMLocal	
<input type="checkbox"/>	SJC-DP	Default	G711-Region	CMLocal	
<input type="checkbox"/>	SJC-DP-G729	Default	G729-Region	CMLocal	

Add New

Select All

Clear All

Delete Selected



Cisco UCM Device Pool for G711

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Device Pool Configuration

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Device Pool Information
Device Pool: SJC-DP (9 members**)

Device Pool Settings
Device Pool Name* SJC-DP
Cisco Unified Communications Manager Group* Default ▾
Calling Search Space for Auto-registration PSTN-Access ▾
Adjunct CSS < None > ▾
Reverted Call Focus Priority Default ▾
Local Route Group < None > ▾
Intercompany Media Services Enrolled Group < None > ▾

Roaming Sensitive Settings
Date/Time Group* CMLocal ▾
Region* G711-Region ▾
Media Resource Group List SJC-MRGL ▾
Location Hub_None ▾
Network Locale < None > ▾
SRST Reference* Use Default Gateway ▾
Connection Monitor Duration***
Single Button Barge* Barge ▾
Join Across Lines* On ▾
Physical Location < None > ▾
Device Mobility Group < None > ▾

Device Mobility Related Information****
Device Mobility Calling Search Space < None > ▾
AAR Calling Search Space < None > ▾



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Device Pool Configuration Related Links

Save Delete Copy Reset Apply Config Add New

Connection Profile Selection

Single Button Barge*

Join Across Lines*

Physical Location

Device Mobility Group

— Device Mobility Related Information**** —

Device Mobility Calling Search Space

AAR Calling Search Space

AAR Group

Calling Party Transformation CSS

Called Party Transformation CSS

— Geolocation Configuration —

Geolocation

Geolocation Filter

— 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. If the field is empty in which case there is no prefix assigned.

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="< None >"/>
International Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="< None >"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="< None >"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="< 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. If the field is empty in which case there is no prefix assigned.



Cisco UCM Device Pool for G729

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Device Pool Configuration

Save Delete Copy Reset Apply Config Add New

Device Pool Settings

Device Pool Name*	SJC-DP-G729
Cisco Unified Communications Manager Group*	Default ▾
Calling Search Space for Auto-registration	PSTN-Access ▾
Adjunct CSS	< None > ▾
Reverted Call Focus Priority	Default ▾
Local Route Group	< None > ▾
Intercompany Media Services Enrolled Group	< None > ▾

Roaming Sensitive Settings

Date/Time Group*	CMLocal ▾
Region*	G729-Region ▾
Media Resource Group List	SJC-MRGL ▾
Location	Hub_None ▾
Network Locale	< None > ▾
SRST Reference*	Use Default Gateway ▾
Connection Monitor Duration***	
Single Button Barge*	Barge ▾
Join Across Lines*	On ▾
Physical Location	< None > ▾
Device Mobility Group	< None > ▾

Device Mobility Related Information****

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

Geolocation Configuration



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Device Pool Configuration

SaveDeleteCopyResetApply ConfigAdd New

Connection Monitor Duration

Single Button Barge*

Join Across Lines*

Physical Location

Device Mobility Group

Device Mobility Related Information****

Device Mobility Calling Search Space

AAR Calling Search Space

AAR Group

Calling Party Transformation CSS

Called Party Transformation CSS

Geolocation Configuration

Geolocation

Geolocation Filter

Call Routing Information

Incoming Calling Party Settings

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

Clear Prefix Settings

Default Prefix Settings


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

Incoming Called Party Settings

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



Cisco UCM Media Termination Point Configuration




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Media Termination Point Configuration

 Save  Reset  Apply Config


- Status -

 Status: Ready

- Media Termination Point Information -

Registration	Registered with Cisco Unified Communications Manager 172.20.8.42
IP Address	172.20.8.42
IPv6 Address	0000:0000:0000:0000:0000:0000:0000:0000
Media Termination Point Type*	Cisco Media Termination Point Software
Host Server*	172.20.8.42
Media Termination Point Name*	<input type="text" value="MTP_2"/>
Description	<input type="text" value="MTP_CUCM851"/>
Device Pool*	<input type="text" value="SJC-DP"/>
<input checked="" type="checkbox"/> Trusted Relay Point	

-

 *- indicates required item.



Cisco UCM Conference Bridge Configurations

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Find and List Conference Bridges

Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected

Status
 2 records found


Conference Bridges (1 - 2 of 2) Rows per Page

Find Conference Bridges where Name ▾ begins with ▾

<input type="checkbox"/>	Conference Bridge Name ^	Description	Device Pool	Status	IP Address
<input type="checkbox"/>	CFB_2	CFB_CUCM851	SJC-DP	Registered with 172.20.8.42	172.20.8.42
<input type="checkbox"/>	CONF-VGW3	CONF-VGW3	SJC-DP	Registered with 172.20.8.42	172.20.8.58






Software Conference Bridge


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Conference Bridge ConfigurationRelated Links: [Back To Find/List](#)

 Save  Reset  Apply Config


Status


 Status: Ready

Conference Bridge Information

Conference Bridge : CFB_2 (CFB_CUCM851)
Registration Registered with Cisco Unified Communications Manager 172.20.8.42
IP Address 172.20.8.42

Software Conference Bridge Info

Conference Bridge Type* Cisco Conference Bridge Software
Host Server 172.20.8.42
 Device is not trusted
Conference Bridge Name*
Description
Device Pool*
Common Device Configuration
Location*
Use Trusted Relay Point*

 *- indicates required item.



Hardware Conference Bridge

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Call Routing ▾

Media Resources ▾

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Help ▾

Conference Bridge Configuration

Related Links: [Back To Find/List](#)

Save

Delete

Copy

Reset

Apply Config

Add New

Status

Status: Ready

Conference Bridge Information

Conference Bridge : CONF-VGW3 (CONF-VGW3)

Registration Registered with Cisco Unified Communications Manager 172.20.8.42

IP Address [172.20.8.58](#)

IOS Conference Bridge Info

Conference Bridge Type* Cisco IOS Conference Bridge

Device is not trusted

Conference Bridge Name*

Description

Device Pool*

Common Device Configuration


Location*

Use Trusted Relay Point*

*- indicates required item.



Cisco UCM Media Resource Group





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Media Resource Group Configuration

 Save  Delete  Copy  Add New

- Status

 Status: Ready

- Media Resource Group Status

Media Resource Group: SJC-MRG (used by 9 devices)

- Media Resource Group Information

Name*

Description





- Devices for this Group


Available Media Resources**


Selected Media Resources*

ANN_2 (ANN)
CFB_2 (CFB)
CONF-VGW3 (CFB)
MOH_2 (MOH)
MTP_2 (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)







Cisco UCM Media Resource Group List


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Media Resource Group List Configuration

 Save  Delete  Copy  Add New

— Status

 Status: Ready

— Media Resource Group List Status

Media Resource Group List: SJC-MRGL (used by 9 devices)

— Media Resource Group List Information





Name*


— Media Resource Groups for this List

Available Media Resource Groups

Selected Media Resource Groups


SJC-MRG

 Save  Delete  Copy  Add New

 *- indicates required item.



Cisco UCM Route Pattern Configuration





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Find and List Route Patterns

 Add New  Select All  Clear All  Delete Selected

Status

 8 records found


Route Patterns (1 - 8 of 8) Rows per P

Find Route Patterns where  

<input type="checkbox"/>	Pattern ^	Description	Partition	Route Filter	Associated Device
<input type="checkbox"/>	347X		SJC-Phones-PT		Fax-GW
<input type="checkbox"/>	7.@	Fax LD	SJC-LD-CUBE-PT		SME
<input type="checkbox"/>	7011!	FaxInternational	SJC-LD-CUBE-PT		SME
<input type="checkbox"/>	7011!#	Fax LD	SJC-LD-CUBE-PT		SME
<input type="checkbox"/>	9.@		CF-PT		SME
<input type="checkbox"/>	9.@		SJC-LD-CUBE-PT		SME
<input type="checkbox"/>	9011!		SJC-LD-CUBE-PT		SME
<input type="checkbox"/>	9011!#		SJC-LD-CUBE-PT		SME







PSTN Route Pattern

**Cisco Unified CM Administration**
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Route Pattern Configuration

 Save  Delete  Copy  Add New

Status

 Status: Ready

Pattern Definition

Route Pattern*	9.@
Route Partition	SJC-LD-CUBE-PT ▾
Description	
Numbering Plan*	NANP ▾
Route Filter	< None > ▾
MLPP Precedence*	Default ▾
Resource Priority Namespace Network Domain	< None > ▾
Route Class*	Default ▾
Gateway/Route List*	SME ▾ (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error ▾
Call Classification*	OffNet ▾

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level*

☐ 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 ▾



Route Pattern Configuration

Save Delete Copy Add New

☐ Require Client Matter Code

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Calling Name Presentation*
Calling Party Number Type*
Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*
Connected Name Presentation*

Called Party Transformations

Discard Digits
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element


Network Service Protocol
Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

Save Delete Copy Add New



International Route Pattern





**Cisco Unified CM Administration**
For Cisco Unified Communications Solutions

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Route Pattern Configuration

 Save  Delete  Copy  Add New

Status

 Status: Ready

Pattern Definition

Route Pattern*	<input type="text" value="9011!#"/>
Route Partition	<input type="text" value="SJC-LD-CUBE-PT"/>
Description	<input type="text"/>
Numbering Plan	<input type="text" value="-- Not Selected --"/>
Route Filter	<input type="text" value="< None >"/>
MLPP Precedence*	<input type="text" value="Default"/>
Resource Priority Namespace Network Domain	<input type="text" value="< None >"/>
Route Class*	<input type="text" value="Default"/>
Gateway/Route List*	<input type="text" value="SME"/> (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="No Error"/>
Call Classification*	<input type="text" value="OffNet"/>
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	<input type="text" value="0"/>
<input type="checkbox"/> Require Client Matter Code	

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	<input type="text"/>
Prefix Digits (Outgoing Calls)	<input type="text"/>
Calling Line ID Presentation*	<input type="text" value="Default"/>
Calling Name Presentation*	<input type="text" value="Default"/>
Calling Party Number Type*	<input type="text" value="Cisco CallManager"/>
Calling Party Numbering Plan*	<input type="text" value="Cisco CallManager"/>

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Route Pattern Configuration Related Links

Save Delete Copy Add New

☐ Require Client Matter Code

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

*- indicates required item.



Fax Route Pattern

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

Route Pattern Configuration Related L

Save Delete Copy Add New

Pattern Definition

Route Pattern*	7.0
Route Partition	SJC-LD-CUBE-PT ▾
Description	Fax LD
Numbering Plan*	NANP ▾
Route Filter	< None > ▾
MLPP Precedence*	Default ▾
Resource Priority Namespace Network Domain	< None > ▾
Route Class*	Default ▾
Gateway/Route List*	SME ▾ (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error ▾
Call Classification*	OffNet ▾
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0
<input type="checkbox"/> 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 ▾
---------------------------------	-----------



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Route Pattern Configuration

Save Delete Copy Add New

☐ Require Client Matter Code

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

*- indicates required item.



Cisco UCM SIP trunks configurations

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Find and List Trunks

Add New

Select All

Clear All

Delete Selected

Reset Selected

Status
 8 records found

Trunks (1 - 8 of 8)

Find Trunks where begins with

<input type="checkbox"/>	Name ^	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type
<input type="checkbox"/>	Fax-GW	PSTN-Access	SJC-DP	347X	SJC-Phones-PT				SIP Trunk
<input type="checkbox"/>	SME	Inbound-CSS	SJC-DP	7011	SJC-LD-CUBE-PT				SIP Trunk
<input type="checkbox"/>	SME	Inbound-CSS	SJC-DP	9.@	CF-PT				SIP Trunk
<input type="checkbox"/>	SME	Inbound-CSS	SJC-DP	7.@	SJC-LD-CUBE-PT				SIP Trunk
<input type="checkbox"/>	SME	Inbound-CSS	SJC-DP	9011	SJC-LD-CUBE-PT				SIP Trunk
<input type="checkbox"/>	SME	Inbound-CSS	SJC-DP	9011#	SJC-LD-CUBE-PT				SIP Trunk
<input type="checkbox"/>	SME	Inbound-CSS	SJC-DP	9.@	SJC-LD-CUBE-PT				SIP Trunk
<input type="checkbox"/>	SME	Inbound-CSS	SJC-DP	7011#	SJC-LD-CUBE-PT				SIP Trunk



SIP Trunk Configuration for the Session Manager Edition

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Trunk Configuration Related Links:

Save Delete Reset Add New

Device Information

Product:

Device Protocol:

Trunk Service Type

Device Name*

Description

Device Pool*

Common Device Configuration

Call Classification*

Media Resource Group List

Location*

AAR Group

Tunneled Protocol*

QSIG Variant*

ASN.1 ROSE OID Encoding*

Packet Capture Mode*

Packet Capture Duration

☐ 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*

Route Class Signaling Enabled*

Use Trusted Relay Point*

☒ PSTN Access

☐ Run On All Active Unified CM Nodes

SIP Trunk

SIP

None(Default)

SME

SJC-DP

< None >

Use System Default

SJC-MRGL

Hub_None

< None >

None

No Changes

No Changes

None

0

When using both sRTP and TLS

Default

Default

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Trunk Configuration

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Save

Delete

Reset

Add New

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type*

ppI

SIP Privacy*

Default

Inbound Calls

Significant Digits*

4

Connected Line ID Presentation*

Allowed

Connected Name Presentation*

Allowed

Calling Search Space

Inbound-CSS

AAR Calling Search Space

< None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
Incoming Number	<div>Default</div>	<div>0</div>	<div>< None ></div>

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



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Related

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Add New

Outbound Calls

Called Party Transformation CSS

< None >

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

< None >

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Last Redirect Number

Calling Line ID Presentation*

Default

Calling Name Presentation*

Default

Caller ID DN

Caller Name

☒ Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination

☐ Destination Address is an SRV

1*

Destination Address

172.20.8.41

Destination Address IPv6

Destination Port

5060

MTP Preferred Originating Codec*

711ulaw

Presence Group*

Standard Presence group

SIP Trunk Security Profile*

Non Secure SIP Trunk Profile

Rerouting Calling Search Space

< None >

Out-Of-Dialog Refer Calling Search Space

< None >

SUBSCRIBE Calling Search Space

< None >

SIP Profile*

SME-Profile1

DTMF Signaling Method*

No Preference

Normalization Script

Normalization Script

< None >

☐ Enable Trace

Parameter Name

Parameter Value



SIP Trunk Configuration for the Fax Gateway

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Trunk Configuration Related

Save Delete Reset Add New

Device Information

Product:
Device Protocol:
Trunk Service Type
Device Name*
Description
Device Pool*
Common Device Configuration
Call Classification*
Media Resource Group List
Location*
AAR Group
Tunneled Protocol*
QSIG Variant*
ASN.1 ROSE OID Encoding*
Packet Capture Mode*
Packet Capture Duration
☐ 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 info
Consider Traffic on This Trunk Secure*
Route Class Signaling Enabled*
Use Trusted Relay Point*
☒ PSTN Access
☐ Run On All Active Unified CM Nodes

SIP Trunk
SIP
None(Default)
Fax-GW

SJC-DP
< None >
Use System Default
SJC-MRGL
Hub_None
< None >
None
No Changes
No Changes
None
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 info
Consider Traffic on This Trunk Secure*
Route Class Signaling Enabled*
Use Trusted Relay Point*
☒ PSTN Access
☐ Run On All Active Unified CM Nodes



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Trunk Configuration Related Links: Ba

Save Delete Reset Add New

☒ Remote-Party-Id
☒ Asserted-Identity
Asserted-Type* Default ▾
SIP Privacy* Default ▾

Inbound Calls
Significant Digits* All ▾
Connected Line ID Presentation* Default ▾
Connected Name Presentation* Default ▾
Calling Search Space PSTN-Access ▾
AAR Calling Search Space < None > ▾
Prefix DN

☐ Redirecting Diversion Header Delivery - Inbound

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

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use
Incoming Number	Default	0	< None > ▾	<input checked="" type="checkbox"/>

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 Party Transformation*



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Trunk Configuration

Related

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SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	
1 *	172.20.109.201		5060	<div>+</div> <div>-</div>

MTP Preferred Originating Codec*

711ulaw

Presence Group*

Standard Presence group

SIP Trunk Security Profile*

Non Secure SIP Trunk Profile

Rerouting Calling Search Space

< None >

Out-Of-Dialog Refer Calling Search Space

< None >

SUBSCRIBE Calling Search Space

< None >

SIP Profile*

Standard SIP Profile

DTMF Signaling Method*

No Preference

Normalization Script

Normalization Script

< None >

☐ Enable Trace

	Parameter Name	Parameter Value	
1			<div>+</div> <div>-</div>

Geolocation Configuration

Geolocation

< None >

Geolocation Filter

< None >

☐ Send Geolocation Information

Save | Delete | Reset | Add New

*

indicates required item.

**

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



SCCP Phone configurations on the Cisco UCM

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Find and List Phones Related Links: [Actively Logged](#)

Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected

Status
 3 records found

Phone (1 - 3 of 3)

Find Phone where

<input type="checkbox"/>		Device Name(Line) ^	Description	Device Pool	Device Protocol	Status	IP Address
<input type="checkbox"/>	7961G-GE	SEP00235E18EDDC	Matt Damon 6162	SJC-DP	SCCP	Registered with 172.20.8.42	172.20.8.19
<input type="checkbox"/>	7961	SEP0026CB3BBA09	Mark Whalberg 6164	SJC-DP	SCCP	Unknown	Unknown
<input type="checkbox"/>	7961	SEP0026CB3BBA44	Leonardo DiCaprio 6163	SJC-DP	SCCP	Registered with 172.20.8.42	172.20.8.223

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Phone Configuration
Related Links: [Back To Find/List](#)

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

Status
 Status: Ready

Association Information


Modify Button Items

1	Line [1] - 6162 in SJC-Phones-PT
2	Line [2] - Add a new DN
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
----- Unassigned Associated Items -----	
7	Add a new SD
8	Add a new SURL
9	Add a new BLF SD
10	Add a new BLF Directed Call Park
11	CallBack
12	Call Park
13	Call Pickup
14	Conference List
15	Conference
16	Do Not Disturb
17	End Call
18	Forward All
19	Group Call Pickup
20	Hold
21	Hunt Group Logout
22	Intercom [1] - Add a new Intercom
23	Malicious Call Identification

Phone Type
Product Type: Cisco 7961G-GE
Device Protocol: SCCP

Device Information

Registration	Registered with Cisco Unified Communications Manager 172.20.8.42
IP Address	172.20.8.19
Active Load ID	Unknown
<input checked="" type="checkbox"/> Device is Active	
<input checked="" type="checkbox"/> Device is trusted	
MAC Address*	<input type="text" value="00235E18EDDC"/>
Description	<input type="text" value="Matt Damon 6162"/>
Device Pool*	<input type="text" value="SJC-DP"/> View Details
Common Device Configuration	<input type="text" value="< None >"/> View Details
Phone Button Template*	<input type="text" value="Standard 7961G-GE SCCP"/>
Softkey Template	<input type="text" value="< None >"/>
Common Phone Profile*	<input type="text" value="Standard Common Phone Profile"/>
Calling Search Space	<input type="text" value="PSTN-Access"/>
AAR Calling Search Space	<input type="text" value="< None >"/>
Media Resource Group List	<input type="text" value="SJC-MRGL"/>
User Hold MOH Audio Source	<input type="text" value="< None >"/>
Network Hold MOH Audio Source	<input type="text" value="< None >"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="< None >"/>
User Locale	<input type="text" value="< None >"/>
Network Locale	<input type="text" value="< None >"/>
Built In Bridge*	<input type="text" value="Default"/>
Privacy*	<input type="text" value="Default"/>



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Malicious Call Identification

24

Meet Me Conference

25

Mobility

26

New Call

27

Other Pickup

28

Quality Reporting Tool

29

Redial

30

Remove Last Participant

31

Transfer

32

Video Mode

33

Privacy

34

None

Device Mobility Mode*

Default

[View Current Device Mobili](#)

Owner User ID

< None >

Phone Personalization*

Default

Services Provisioning*

Default

Phone Load Name

Single Button Barge

Default

Join Across Lines

Default

Use Trusted Relay Point*

Default

BLF Audible Alert Setting (Phone Idle)*

Default

BLF Audible Alert Setting (Phone Busy)*

Default

Always Use Prime Line*

Default

Always Use Prime Line for Voice Message*

Default

Calling Party Transformation CSS

< None >

Geolocation

< None >

☒ Use Device Pool Calling Party Transformation CSS

☒ Retry Video Call as Audio

☐ Ignore Presentation Indicators (internal calls only)

☒ Allow Control of Device from CTI

☒ Logged Into Hunt Group

☐ Remote Device

☐ Protected Device****

☐ Hot line Device*****

Protocol Specific Information

Packet Capture Mode*

None

Packet Capture Duration

0

Presence Group*

Standard Presence group

Device Security Profile*

Cisco 7961G-GE - Standard SCCP Non-Secure Prot

SUBSCRIBE Calling Search Space

< None >



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Phone Configuration

Related Links: [Back To Find/...](#)

Save Delete Copy Reset Apply Config Add New

Packet Capture Mode

None

Packet Capture Duration

0

Presence Group*

Standard Presence group

Device Security Profile*

Cisco 7961G-GE - Standard SCCP Non-Secure Prol

SUBSCRIBE Calling Search Space

< None >

☐ 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

201110212 (YYYY:MM:DD:HH)

Certificate Operation Status: None

Note: Security Profile Contains Addition CAPF Settings.

Expansion Module Information

Module 1

< None >

Module 1 Load Name

Module 2

< None >

Module 2 Load Name

External Data Locations Information (Leave blank to use default)

Information

Directory

Messages

Services

Authentication Server



Phone Configuration

Related Links: [Back To Find/List](#)

Save

Delete

Copy

Reset

Apply Config

Add New

MLPP Information

MLPP Domain

< None >

MLPP Indication*

Default

MLPP Preemption*

Default

Do Not Disturb

☐ Do Not Disturb

DND Option*

Use Common Phone Profile Setting

DND Incoming Call Alert

< None >

Secure Shell Information

Secure Shell User

Secure Shell Password

Product Specific Configuration Layout

?

Param

Over

☐ Disable Speakerphone

☐ Disable Speakerphone and Headset

Forwarding Delay*

Disabled

PC Port *

Enabled

Settings Access*

Enabled

☐

Gratuitous ARP*

Disabled

PC Voice VLAN Access*

Enabled

Video Capabilities*

Disabled

☐

Auto Line Select*

Disabled

Web Access*

Disabled

☐

Span to PC Port*

Disabled

Logging Display*

PC Controlled

Load Server

☐

Recording Tone*

Disabled



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Save Delete Copy Reset Apply Config Add New

Recording Tone Remote Volume	50	
Recording Tone Duration		
RTCP*	Disabled	<input type="checkbox"/>
"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	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): PC Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled	<input type="checkbox"/>
LLDP Asset ID		
LLDP Power Priority*	Unknown	
Display Refresh Rate*	Normal	
IPv6 Load Server		<input type="checkbox"/>
IPv6 Log Server		
802.1x Authentication*	User Controlled	<input type="checkbox"/>
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	<input type="checkbox"/>
PC Port Remote Configuration*	Disabled	<input type="checkbox"/>
Automatic Port Configuration*		



DN Configuration for SCCP Phone

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Directory Number Configuration

Related Links: [Configure](#)

Save

Delete

Reset

Apply Config

Add New

Directory Number Information

Directory Number*

6162

Route Partition

SJC-Phones-PT

Description

Matt Damon

Alerting Name

Matt Damon

ASCII Alerting Name

Matt Damon

☒ Allow Control of Device from CTI

Associated Devices

SEP00235E18EDDC

Edit Device

Edit Line Appearance

Dissociate Devices

Directory Number Settings

Voice Mail Profile

< None >

(Choose <None> to use system default)

Calling Search Space

PSTN-Access

Presence Group*

Standard Presence group

User Hold MOH Audio Source

< None >

Network Hold MOH Audio Source

< None >

Auto Answer*

Auto Answer Off

AAR Settings

AAR

☐ or

< None >

☒ Retain this destination in the call forwarding history

Call Forward and Call Pickup Settings

Calling Search Space Activation Policy

Use System Default



☒ 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	<input type="checkbox"/> or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input checked="" type="checkbox"/> or		CF-CSS
Forward Busy External	<input checked="" type="checkbox"/> or		CF-CSS
Forward No Answer Internal	<input checked="" type="checkbox"/> or		CF-CSS
Forward No Answer External	<input checked="" type="checkbox"/> or		CF-CSS
Forward No Coverage Internal	<input checked="" type="checkbox"/> or		CF-CSS
Forward No Coverage External	<input checked="" type="checkbox"/> or		CF-CSS
Forward on CTI Failure	<input type="checkbox"/> or		CF-CSS
Forward Unregistered Internal	<input checked="" type="checkbox"/> or		CF-CSS
Forward Unregistered External	<input checked="" type="checkbox"/> or		CF-CSS
to Answer Ring Duration (seconds)			
Call Pickup Group			< None >

Park Monitoring

	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or		< None > A blank value means to parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or		< None > A blank value means to parker's line.
Park Monitoring Reversion Timer		0	A blank value will use value set in Park Monitoring Reversion Timer service parameter



Directory Number Configuration

Related Links: [Configure Device](#)

Save

Delete

Reset

Apply Config

Add New

MLPP Alternate Party Settings

Target (Destination)

MLPP Calling Search Space

< None >

MLPP No Answer Ring Duration (seconds)

Line Settings for All Devices

Hold Reversion Ring Duration (seconds)

0

Setting the Hold Reversion Ring Duration to zero will disable the feature

Hold Reversion Notification Interval (seconds)

0

Setting the Hold Reversion Notification Interval to zero will disable the feature

Party Entrance Tone*

Default

Line 1 on Device SEP00235E18EDDC

Display (Internal Caller ID)

Matt Damon

Display text for a line appearance is intended for displaying text such as a name instead of a dir calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.

ASCII Display (Internal Caller ID)

Matt Damon

Line Text Label

Matt Damon 6162

ASCII Line Text Label

Matt Damon 6162

External Phone Number Mask

408933XXXX

Visual Message Waiting Indicator Policy*

Use System Policy

Audible Message Waiting Indicator Policy*

Default

Ring Setting (Phone Idle)*

Ring

Ring Setting (Phone Active)

Use System Default

Applies to this line when any line on the phone has a call in progress.

Call Pickup Group Audio Alert Setting(Phone Idle)

Use System Default

Call Pickup Group Audio Alert Setting(Phone Active)

Use System Default

Recording Option*

Call Recording Disabled

Recording Profile

< None >

Monitoring Calling Search Space

< None >

☒ Log Missed Calls



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Directory Number Configuration

Related Links: Configure

Save

Delete

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Apply Config

Add New

Audible Message Waiting Indicator Policy*

Ring Setting (Phone Idle)*

Ring Setting (Phone Active)

Call Pickup Group Audio Alert Setting(Phone Idle)

Call Pickup Group Audio Alert Setting(Phone Active)

Recording Option*

Recording Profile

Monitoring Calling Search Space

☒ Log Missed Calls

Default

Ring

Use System Default

Use System Default

Use System Default

Call Recording Disabled

< None >

< None >

Applies to this line when any line on the phone has a call in progress.

Multiple Call/Call Waiting Settings on Device SEP00235E18EDDC

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

Maximum Number of Calls*

Busy Trigger*

3

2

(Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP00235E18EDDC

☒ Caller Name

☒ Caller Number

☒ Redirected Number

☒ Dialed Number

Users Associated with Line

Associate End Users

Save

Delete

Reset

Apply Config

Add New

i

*. indicates required item.

i

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



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Save Delete Copy Reset Apply Config Add New

24	Answer Oldest
25	Do Not Disturb
26	Services
27	Record
28	Privacy
29	None

Device Mobility Mode* Default [View Current Device Mobility Settings](#)

Owner User ID < None >

Phone Personalization* Default

Services Provisioning* Default

Phone Load Name

Use Trusted Relay Point* Default

BLF Audible Alert Setting (Phone Idle)* Default

BLF Audible Alert Setting (Phone Busy)* Default

Always Use Prime Line* Default

Always Use Prime Line for Voice Message* Default

Calling Party Transformation CSS < None >

Geolocation < None >

Feature Control Policy < None >

☒ Use Device Pool Calling Party Transformation CSS

☐ Ignore Presentation Indicators (internal calls only)

☒ Allow Control of Device from CTI

☒ Logged Into Hunt Group

☐ Remote Device

☐ Protected Device****

Protocol Specific Information

Packet Capture Mode* None

Packet Capture Duration 0

Presence Group* Standard Presence group

SIP Dial Rules < None >

MTP Preferred Originating Codec* 711ulaw

Device Security Profile* Cisco 9971 - Standard SIP Non-Secure Profile

Rerouting Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* Standard SIP Profile

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

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

Phone Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Reset Apply Config Add New

Certificate Operation* No Pending Operation

Authentication Mode* By Null String

Authentication String

Generate String

Key Size (Bits)* 1024

Operation Completes By 2011 10 12 12 (YYYY:MM:DD:HH)

Certificate Operation Status: None

Note: Security Profile Contains Addition CAPF Settings.

Expansion Module Information

Module 1 < None >

Module 1 Load Name

Module 2 < None >

Module 2 Load Name

Module 3 < None >

Module 3 Load Name

External Data Locations Information (Leave blank to use default)

Information

Directory

Messages

Services

Authentication Server

Proxy Server

Idle

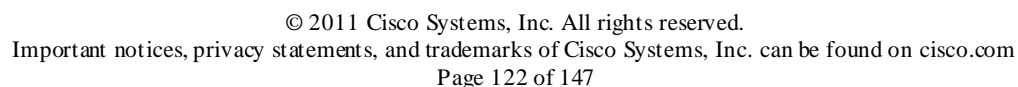
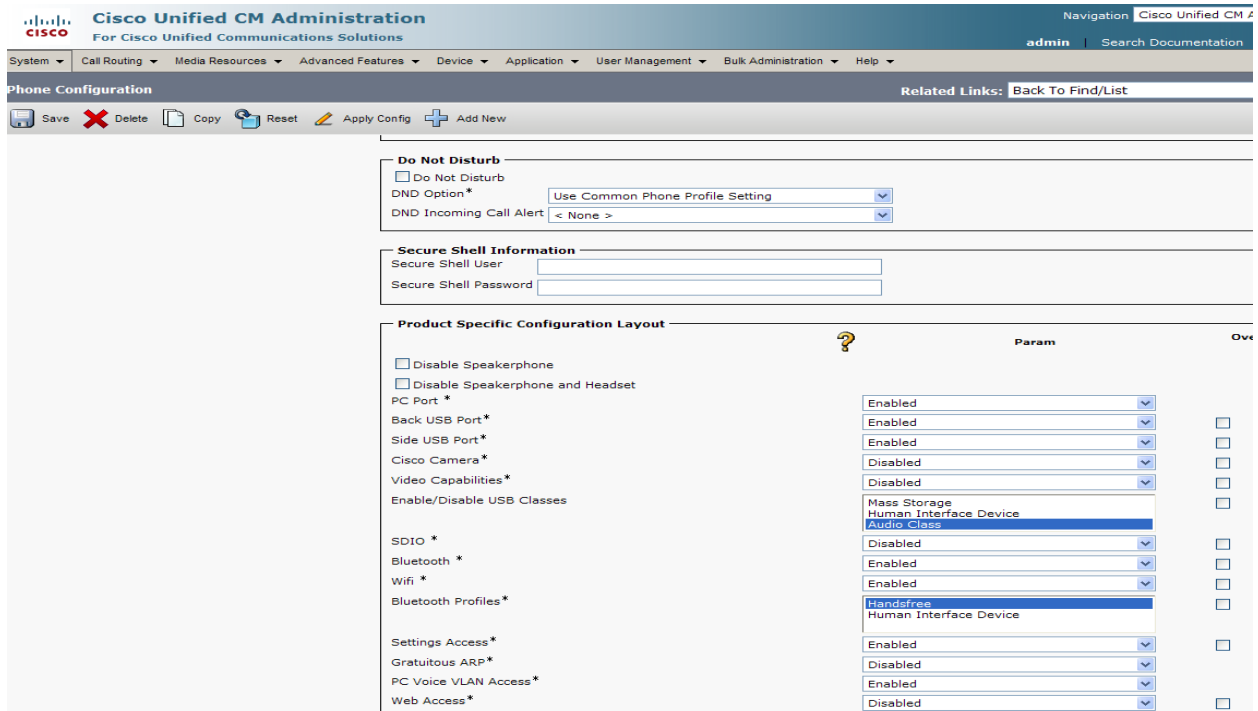
Idle Timer (seconds)

Secure Authentication URL

Secure Directory URL

Secure Idle URL

Secure Information URL





DN Configuration for the SIP Phone

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Directory Number ConfigurationRelated Links: [Configure De](#)

Save Delete Reset Apply Config Add New

- Status -
 Status: Ready

- Directory Number Information -
Directory Number*
Route Partition
Description
Alerting Name
ASCII Alerting Name
☒ Allow Control of Device from CTI
Associated Devices

▼ ^
Dissociate Devices

- Directory Number Settings -
Voice Mail Profile (Choose <None> to use system default)
Calling Search Space
Presence Group*
User Hold MOH Audio Source
Network Hold MOH Audio Source
Auto Answer*

- AAR Settings -

	Voice Mail	AAR Destination Mask	AAR Gro
AAR <input type="checkbox"/>	or <input type="text"/>	<input type="text" value="< None >"/>	
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history			



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Directory Number Configuration

Related Links: [Configure Device \(SEPE8BA7006ACA1\)](#)

Save | Delete | Reset | Apply Config | Add New

Configure and Call Forward Settings

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	<input type="checkbox"/> or		CF-CSS
Secondary Calling Search Space for Forward All			CF-CSS
Forward Busy Internal	<input type="checkbox"/> or		CF-CSS
Forward Busy External	<input type="checkbox"/> or		CF-CSS
Forward No Answer Internal	<input type="checkbox"/> or		CF-CSS
Forward No Answer External	<input type="checkbox"/> or		CF-CSS
Forward No Coverage Internal	<input type="checkbox"/> or		CF-CSS
Forward No Coverage External	<input type="checkbox"/> or		CF-CSS
Forward on CTI Failure	<input type="checkbox"/> or		CF-CSS
Forward Unregistered Internal	<input type="checkbox"/> or		CF-CSS
Forward Unregistered External	<input type="checkbox"/> or		CF-CSS
No Answer Ring Duration (seconds)		25	
Call Pickup Group		< None >	

Park Monitoring

	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or		< None > A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or		< None > A blank value means to call the parker's line.
Park Monitoring Reversion Timer			A blank value will use value set in Park Monitoring Reversion Timer service parameter

MLPP Alternate Party Settings

Target (Destination)	
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	



Line 1 on Device SEPE8BA7006ACA1

Display (Internal Caller ID)	<input type="text" value="Joe Doe"/>	Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)	<input type="text" value="Joe Doe"/>	
Line Text Label	<input type="text" value="Joe Doe"/>	
ASCII Line Text Label	<input type="text" value="Joe Doe"/>	
External Phone Number Mask	<input type="text" value="4089336165"/>	
Visual Message Waiting Indicator Policy*	<input type="button" value="Use System Policy"/>	
Audible Message Waiting Indicator Policy*	<input type="button" value="Default"/>	
Ring Setting (Phone Idle)*	<input type="button" value="Use System Default"/>	
Ring Setting (Phone Active)	<input type="button" value="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)	<input type="button" value="Use System Default"/>	
Call Pickup Group Audio Alert Setting(Phone Active)	<input type="button" value="Use System Default"/>	
Recording Option*	<input type="button" value="Call Recording Disabled"/>	
Recording Profile	<input type="button" value=" < None >"/>	
Monitoring Calling Search Space	<input type="button" value=" < None >"/>	
<input checked="" type="checkbox"/> Log Missed Calls		

Multiple Call/Call Waiting Settings on Device SEPE8BA7006ACA1

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

Maximum Number of Calls*	<input type="text" value="4"/>	
Busy Trigger*	<input type="text" value="2"/>	(Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEPE8BA7006ACA1

- ☒ Caller Name
 - ☒ Caller Number
 - ☒ Redirected Number
 - ☒ Dialed Number
-



Cisco IOS Gateway Configurations

Cisco Unified Border Element Configuration on the ASR

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

Cisco IOS-XE software, Copyright (c) 2005-2011 by cisco Systems, Inc.
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ROM: IOS-XE ROMMON

CUBE-ASR1K_Vz_151 uptime is 5 weeks, 2 days, 19 hours, 48 minutes
Uptime for this control processor is 5 weeks, 2 days, 19 hours, 50 minutes
System returned to ROM by reload
System image file is "bootflash:asr1000rp1-adventerprisek9.03.03.01.S.151-2.S1.bin"
Last reload reason: Reload Command
This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
<http://www.cisco.com/wwl/export/crypto/tool/stqrg.html>

If you require further assistance please contact us by sending email to export@cisco.com.



cisco ASR1002 (2RU) processor with 1703423K/6147K bytes of memory.
4 Gigabit Ethernet interfaces
32768K bytes of non-volatile configuration memory.
4194304K bytes of physical memory.
7798783K bytes of eUSB flash at bootflash:.

Configuration register is 0x2102

CUBE-ASR1K_Vz_151#show run
Building configuration...

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



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




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



```
!  
control-plane  
!  
!  
!  
dial-peer voice 9000 voip1  
description to SP facing  
destination-pattern .....  
session protocol sipv2  
session target sip-server  
session transport udp  
voice-class sip asserted-id pai  
dtmf-relay rtp-nte  
fax protocol pass-through g711ulaw  
!  
dial-peer voice 9001 voip2  
description Incoming to SME - SME facing  
destination-pattern 408933....  
session protocol sipv2  
session target ipv4:172.20.8.41  
session transport tcp  
dtmf-relay rtp-nte  
codec g711ulaw  
fax protocol pass-through g711ulaw  
no vad  
!  
dial-peer voice 577 voip  
description Outbound to Fax Machines  
destination-pattern 408577....  
session protocol sipv2  
session target sip-server  
session transport udp  
incoming called-number 408577....  
codec g711ulaw  
fax protocol pass-through g711ulaw  
!  
dial-peer voice 3470 voip3  
description To Fax Machines towards SME  
shutdown  
destination-pattern 40893334..  
session protocol sipv2  
session target ipv4:172.20.8.41  
session transport udp  
incoming called-number 40893334..  
codec g711ulaw  
fax protocol pass-through g711ulaw  
!
```

¹ Outbound dialpeer towards service provider

² Outbound dialpeer towards SME. CUBE sends all the 10 digits to SME for centralized dialplan resolution

³ Outbound dialpeer towards fax machine gateway behind the SME (refer topology diagram for details)



```
dial-peer voice 408 voip4
description To SP 7-digit dialing - SP facing
destination-pattern .....
session protocol sipv2
session target sip-server
session transport udp
voice-class sip asserted-id pai
dtmf-relay rtp-nte
codec g711ulaw
fax protocol pass-through g711ulaw
!
dial-peer voice 9003 voip5
description Fax calls using G.711u - SP facing
translation-profile outgoing Outgoing-Fax-G711
destination-pattern 71.....
session protocol sipv2
session target sip-server
voice-class sip early-offer forced
dtmf-relay rtp-nte
codec g711ulaw
fax protocol pass-through g711ulaw
!
dial-peer voice 9900 voip6
description to SP - SME facing
session protocol sipv2
session transport udp
incoming called-number .....
voice-class sip asserted-id pai
dtmf-relay rtp-nte
codec g711ulaw
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 9901 voip7
description Incoming to SME - SP facing
session protocol sipv2
session transport tcp
incoming called-number 408933....
dtmf-relay rtp-nte
codec g711ulaw
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 4408 voip
description To SP 7-digit dialing - SME facing
session protocol sipv2
session transport udp
incoming called-number .....
```

⁴ Dialpeer for 7 digit dialing towards the SP

⁵ Dialpeer for Fax calls using G.711u - SP facing

⁶ Incoming dialpeer SME facing

⁷ Incoming dialpeer SP facing



```
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 9004 voip8
description To SP - International calls - SP facing
destination-pattern 011T
session protocol sipv2
session target sip-server
session transport udp
voice-class sip early-offer forced
dtmf-relay rtp-nte
codec g711ulaw
fax protocol pass-through g711ulaw
!
dial-peer voice 9904 voip9
description To SP International calls - SME facing
session protocol sipv2
session target sip-server
session transport udp
incoming called-number 011T
dtmf-relay rtp-nte
codec g711ulaw
fax protocol pass-through g711ulaw
!
dial-peer voice 1800 voip10
destination-pattern 1800.....
session protocol sipv2
session target sip-server
session transport udp
voice-class sip asserted-id pai
dtmf-relay rtp-nte
codec g711ulaw
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 7 voip11
session protocol sipv2
session target sip-server
session transport udp
incoming called-number 71.....
voice-class sip asserted-id pai
dtmf-relay rtp-nte
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
!
sip-ua
set pstn-cause 1 sip-status 503
```

⁸ International dialing dialpeer towards the SP

⁹ Incoming dialpeer for international calls from the SME

¹⁰ Toll free dialing dialpeer

¹¹ Incoming fax dialpeer SME facing



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



IOS MTP and Hardware CFB Configuration

MSFT-GW2#show ver

Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.1(3)T1, RELEASE SOFTWARE (fc2)

Technical Support: <http://www.cisco.com/techsupport>

Copyright (c) 1986-2011 by Cisco Systems, Inc.

Compiled Sun 27-Mar-11 07:36 by prod_rel_team

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

MSFT-GW2 uptime is 3 weeks, 6 days, 1 hour, 49 minutes

System returned to ROM by reload at 18:30:00 UTC Tue Aug 30 2011

System image file is "flash0:c2900-universalk9-mz.SPA.151-3.T1.bin"

Last reload type: Normal Reload

Last reload reason: Reload Command

Cisco CISCO2921/K9 (revision 1.0) with 999392K/49152K bytes of memory.

Processor board ID FTX1448AH5X

3 Gigabit Ethernet interfaces

2 Channelized E1/PRI ports

2 Channelized (E1 or T1)/PRI ports

DRAM configuration is 64 bits wide with parity enabled.

255K bytes of non-volatile configuration memory.

4099032K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

Device#	PID	SN
*0	CISCO2921/K9	FTX1448AH5X

Technology Package License Information for Module:'c2900'

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

Configuration register is 0x2102

MSFT-GW2#show run



Building configuration...

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

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



```
!  
crypto pki certificate chain TP-self-signed-3443052460  
certificate self-signed 01  
3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 04050030  
31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274  
69666963 6174652D 33343433 30353234 3630301E 170D3131 30383330 31383332  
32345A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649  
4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D33 34343330  
35323436 3030819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281  
8100B84F A83B31D6 4CCB9507 199B0675 21042C87 3AC072A1 A9B87146 B33A81B9  
71449C3A 8640E523 F2E51F5C 53DA8844 25CD60F4 173A27B2 9FE5D4D0 FC6B2086  
CB52FBE7 5B75DBBA 6B0A522C 3FD88D1A 5ABE950D A0E334B8 5DC5C7E4 874CD72B  
51A60F19 F935309C DE7146C1 F5EAAFC2 A0B0DC7D 7FF195DD 6105967F 8EB1ABE8  
79510203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603  
551D2304 18301680 14519620 E52CD148 41A6D69B 8EFEEA88 00ED3CA7 E7301D06  
03551D0E 04160414 519620E5 2CD14841 A6D69B8E FEEA8800 ED3CA7E7 300D0609  
2A864886 F70D0101 04050003 81810076 D12A4112 963CACAA 59657B51 EA051F2F  
4555E416 2FDF32DF 98D031A0 17C88DBB 8ADBD8D9 0770062B 2295D315 7E38C18F  
B4C0CCF6 2555CA47 20D09D2B CED701F9 445AD9B3 322783C8 736F1CEE F2AC2F59  
823B4C75 5985F137 2A1B77AB 8B6007CE 4FC2C29C 228A3BB2 F0B841B8 5CFA5201  
EAEDB235 E35340FD B79542E7 8D754F  
Quit
```

```
!  
voice-card 0  
dspfarm  
dsp services dspfarm12  
!  
!  
voice service pots  
map q850-cause 47 release-source all tone 1  
supplementary-service qsig call-forward  
supplementary-service qsig m wi release-call  
fax rate disable  
!  
voice service voip  
ip address trusted list  
ipv4 172.20.8.38  
ipv4 172.20.8.30  
ipv4 172.20.8.59  
ipv4 172.20.8.85  
ipv4 172.20.8.98  
ip address trusted call-block cause dest-unroutable  
sip  
!  
voice class h323 2  
h225 timeout connect 60  
h225 timeout call-proceeding 5  
h225 timeout ntf 5000  
h225 display-ie ccm-compatible  
call start fast  
call preserve limit-media-detection  
h245 caps suppress nte
```

¹² Enabling DSP sharing and DSp farm services



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



```
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
!
mgcp
mgcp call-agent 172.20.8.38 2427 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
!
mgcp profile default
!
sccp local GigabitEthernet0/0
sccp ccm 172.20.8.42 identifier 11 version 7.0 13
sccp ccm 172.20.8.41 identifier 1 priority 1 version 7.0 14
sccp
!
sccp ccm group 11
associate ccm 11 priority 1
associate profile 11 register CONF-VGW3
!
sccp ccm group 1
associate ccm 1 priority 1
associate profile 1 register XCODE-SJC
associate profile 2 register IOS-MIP-SJC
associate profile 3 register IOS-MIP-SJ-G711
associate profile 7 register G729-Soft
!
dspfarm profile 1 transcode 15
codec g729r8
codec g711ulaw
codec g711alaw
maximum sessions 6
associate application SCCP
!
dspfarm profile 4 transcode
description CME XCODER
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
maximum sessions 2
associate application SCCP
shutdown
```

¹³ For resources registering to the CUCM

¹⁴ For resources registering to the SME

¹⁵ Transcoder resources



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

¹⁶ Conference bridge

¹⁷ Software MTP



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



Fax Gateway Configuration

show ver

Cisco IOS Software, 2800 Software (C2800NM-IPVOICEK9-M), Version 15.0(1)M4, RELEASE SOFTWARE (fc1)

Technical Support: <http://www.cisco.com/techsupport>

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Compiled Thu 28-Oct-10 17:09 by prod_rel_team

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

Cisco 2811 (revision 53.50) with 249856K/12288K bytes of memory.

Processor board ID FTX1040A1LY

2 FastEthernet interfaces

2 Voice FXS interfaces

DRAM configuration is 64 bits wide with parity enabled.

239K bytes of non-volatile configuration memory.

125440K bytes of ATA CompactFlash (Read/Write)

License Info:

License UDI:

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

Configuration register is 0x2102

IOSGW#show run

Building configuration...

Current configuration : 5053 bytes

!

! Last configuration change at 00:50:39 UTC Thu Sep 8 2011

!

version 15.0

service timestamps debug uptime

service timestamps log datetime msec

no service password-encryption

service linenumbers

!

hostname IOSGW

!

boot-start-marker

boot-end-marker

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



```
interface FastEthernet0/1
no ip address
shutdown
duplex auto
speed auto
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 172.16.100.1
ip route 0.0.0.0 0.0.0.0 172.20.109.1
!
!
!
!
control-plane
!
!
voice-port 0/0/0
timeouts ringing infinity
station-id name FAX-GW1
station-id number 4089333473
!
voice-port 0/0/1
timeouts ringing infinity
station-id name 4089336163
station-id number 4089333474
!
!
no mgcp package-capability res-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
!
dial-peer voice 2715 pots
destination-pattern 4089336164
port 0/0/0
forward-digits 0
!
dial-peer voice 2717 pots
destination-pattern 4089336163
port 0/0/1
forward-digits 0
!
dial-peer voice 711 voip18
description outgoing call to International
destination-pattern 7011T
translate-outgoing calling 2
session protocol sipv2
session target ipv4:172.20.8.42
incoming called-number 27..
```

¹⁸ International fax calling



```
dtmf-relay rtp-nte
playout-delay nominal 80
playout-delay mode fixed
codec g711ulaw
no vad
!
dial-peer voice 7 voip19
destination-pattern 71.....
session protocol sipv2
session target ipv4:172.20.8.42
incoming called-number 347.
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 3473 pots
destination-pattern 3473
port 0/0/0
!
dial-peer voice 3474 pots
destination-pattern 3474
port 0/0/1
!
!
sip-ua
protocol mode ipv4
!
!
line con 0
exec-timeout 610 0
password cisco
login
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco
login
!
scheduler allocate 20000 1000
end
```

IOSGW#

¹⁹ Longdistance and local fax calling



Acronyms

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



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