



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

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Table of Contents

Introduction	3
Network Topology	4
Capabilities	4
Limitations	5
Features Supported	6
Features Not Supported and Not Tested	6
System Components	7
Hardware Requirements	7
Software Requirements	7
System Configuration	8
Configuring the Microsoft LYNC 2010 Server Standard Edition	8
Domain Name System Configuration	9
Forward Lookup Zone	9
Reverse Lookup Zones Configuration	10
SRV Records Configuration	11
Lync Server 2010 Site Topology Overview	14
Lync Front End server configuration with FQDN and PSTN Gateway associated with the Mediation Pool	17
Active Directory User Configuration	19
Lync Server 2010 Configuration	23
User Configuration from Control Panel	23
Voice Routing Configuration	25
Microsoft Lync 2010 Enterprise Voice Client Configuration	33
Cisco Unified Communications Manager- Session Manager Edition Configurations	40
Cisco Unified Communications Manager-SME Software release	40
Cisco Unified Communications Manager-SME Region Configuration	41
Cisco Unified Communications Manager-SME Device Pool Configuration	42
Cisco Unified Communications Manager-SME Media Termination Point Configuration	47
Cisco Unified Communications Manager-SME Transcoder Configuration	48
Cisco Unified Communications Manager-SME Media Resource Groups configuration	49
Cisco Unified Communications Manager-SME Media Resource Group Lists configuration	52
Cisco Unified Communications Manager-SME Route Patterns	54
Cisco Unified Communications Manager-SME Route list for PSTN (CUBE) access	64
Cisco Unified Communications Manager-SME Route Group for PSTN Trunk Access	65
Cisco Unified Communications Manager-SME SIP Trunk Configuration	66
Cisco Unified Communications Manager-SME SIP Profile Information	79
Configuring the Cisco Unified Communications Manager	89
Cisco Unified Communications Manager-Software release	89
Cisco Unified Communications Manager-Region Configuration	90
Cisco Unified Communications Manager-Device Pool configuration	92
Cisco Unified Communications Manager-Media Termination Point Configuration	96
Cisco Unified Communications Manager-Conference Bridge Configuration	97
Cisco Unified Communications Manager-Media Resource Group configuration	98
Cisco Unified Communications Manager-Media Resource Group List configuration	99
Cisco Unified Communications Manager-Route Pattern Configuration	100



Cisco Unified Communications Manager-SIP Trunk configuration	105
Cisco Unified Communications Manager-SIP Profile Information.....	115
SCCP/SIP Phone configurations on the Cisco Unified Call Manager.....	125
Cisco IOS Gateway Configurations.....	142
Cisco Unified Border Element Configuration on the ASR.....	142
Acronyms	150
Important Information	151



Introduction

- This application note describes the necessary steps and configurations for connectivity between Cisco Unified Communications Manager-Session Manager Edition (Cisco UCM-SME) 9.0 via Direct SIP to Microsoft Lync 2010 server using Microsoft Mediation Server, and via SIP connection to a Cisco Unified Communications Manager (Cisco UCM) 9.0, and SIP connection to simulated Service Provider (SP) using Cisco Unified Border Element (Cisco UBE) 1.4. The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability between the different leaf nodes (Microsoft Lync 2010, Cisco UCM and Cisco UBE/SP) via the Cisco UCM-SME.
- Features tested are basic call, 3-way (ad-hoc) conference, call transfer (attended and unattended), call forward (all, busy and no answer), hold/resume, DTMF inter-working and voice mail. This test setup also includes a connection to a simulated service provider using SIP trunks. Cisco UBE is used as a session border controller (SBC), providing demarcation, security, and inter-working services between the customer's private network and the service provider's SIP network. Voice mail was tested for Cisco UCM users with mailboxes registered to Unity Connection via a Direct SIP trunk from Cisco UCM-SME. Fax could not be tested (see limitations section).
- During testing, a Cisco ASR 1002 router was used as Cisco UBE, however other Cisco voice gateways can be used and the decision to choose what Cisco gateway model to use is left to the customer. The customer should choose a Cisco IOS gateway model based on the capabilities and the capacity that will be required based on the planned network deployment. Here is a list of Cisco IOS products capable of running Cisco UBE.

[Cisco ASR 1001 Aggregated Services Router](#)

[Cisco ASR 1004 Aggregated Services Router with Route Processor-2](#)

[Cisco ASR 1006 Aggregated Services Router with Route Processor-2](#)

[Cisco 3900 Series Integrated Services Routers](#)

[Cisco 2900 Series Integrated Services Routers](#)

[Cisco AS5350XM Universal Gateway](#)

[Cisco AS5400XM Universal Gateway](#)

[Cisco 1861 Integrated Services Router](#)

[Cisco 881 Integrated Services Router](#)

[Cisco 888 Integrated Services Router](#)

[Cisco IAD880 Series Integrated Access Devices](#)

[Cisco IAD2430 Integrated Access Device](#)

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



Network Topology

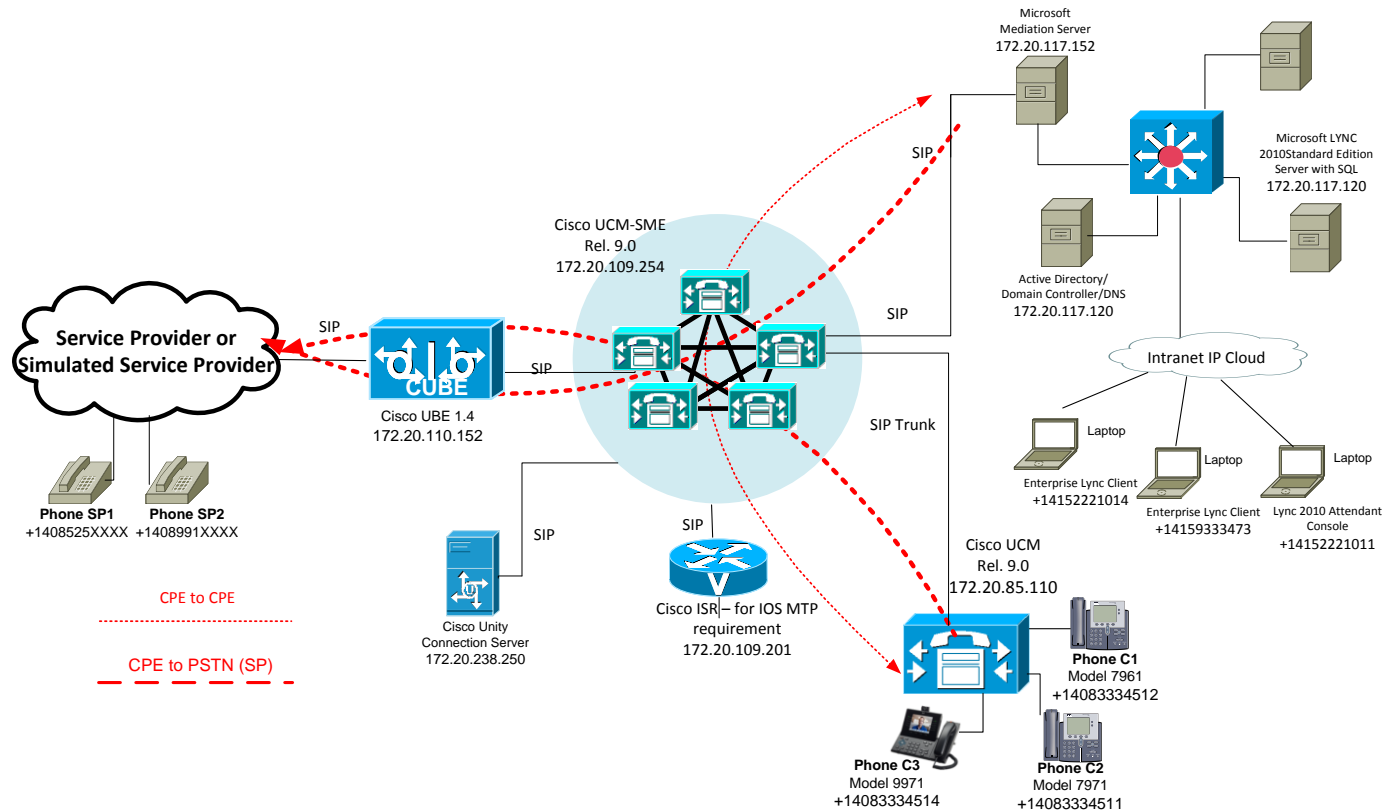


Figure 1. Basic Call Setup

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

Capabilities

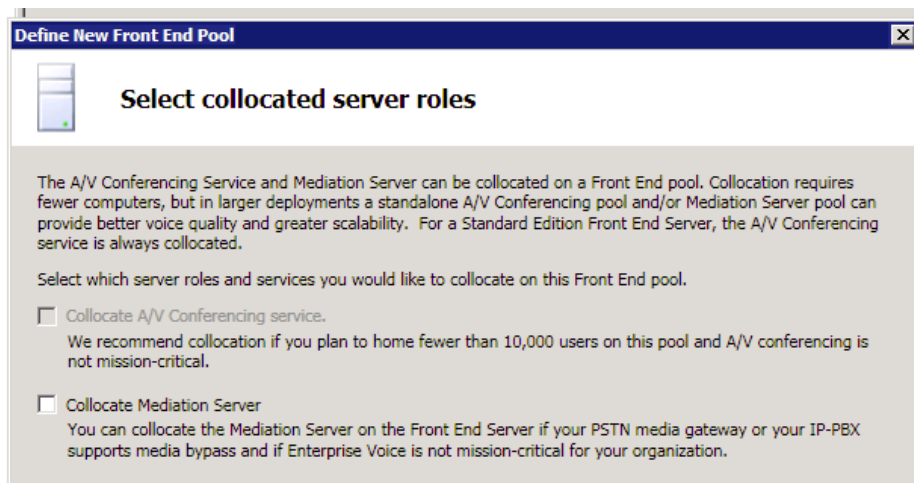
- Voice calls including supplementary services can be successfully established between endpoints controlled by the Microsoft Lync 2010 and endpoints controlled by the Cisco UCM.
- Voice calls including supplementary services can be successfully established between endpoints controlled by the Microsoft Lync 2010 and the PSTN, using Cisco UBE as a session border controller.
- IOS MTPs and transcoders registered to the Cisco UCM-SME can be employed to offer G729 as originating codec option.
- Cisco UCM now supports the use of the Audio Codec Preference and it uses them to select codec or prioritize codec's in the list. System now allows configuring customized Codec Preference List changing the order of codec's of any existing lists (Lossy and Low-Loss) as needed. System also allows the Admin to control if the received preference order of codec for SIP device can be honored by configuring "Accept Audio Codec Preferences in Received Offer" in SIP Profile or Service Parameter.
- Cisco UCM supports URI dialing but with certain new model of IP Phones and other endpoints only.



Limitations

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

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



- Media Bypass works as long as an MTP is required on the SIP trunk. The need for an MTP is due to the following:
 1. Lync supports Media Bypass with early offer, but does not support re-invite without SDP



2. Lync doesn't support Media Bypass with delayed offer
3. Because of #1, hold/resume, transfer, forward, etc won't work without an MTP.

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

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

Features Supported

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

Features Not Supported and Not Tested

- T.38 FAX Relay was not tested. - The Service provider used for this test does not support T.38 FAX.
- G.711 fax pass-through was not tested as Microsoft Lync does not support it.
- Blind call transfers



- Comfort noise is not supported
- Decline a call (CUCM phones divert calls to voicemail, instead of declining the call)

System Components

Hardware Requirements

The following hardware is required:

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

Software Requirements

The following software is required:

- Cisco Unified Communications Manager Release 9.0
- Cisco UCM-Session Manager Edition Release 9.0
- Microsoft Lync Server 2010 Standard Edition, Windows Server 2008 R2 SP1 x64 Enterprise Edition OS
- Windows Active Directory/DNS/Cert Server, Windows Server 2008 R2 SP1 x64 Enterprise Edition OS
- Backend Database: Windows SQL Server 2008 Enterprise Edition, Windows Server 2008 R2 SP1 x64 Enterprise Edition OS
- Microsoft Lync 2010 (build 4.0.7577.4)
- Cisco ASR1K router Release 3.4.2.S

IMPORTANT:

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



System Configuration

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

Configuring the Microsoft LYNC 2010 Server Standard Edition

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

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

http://www.cisco.com/en/US/prod/collateral/voicesw/ps6788/vcallcon/ps5556/ps12515/data_sheet_c78-710875.html

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

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

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

For Microsoft deployment and installation guides, go to:

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

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



Configuration Steps:

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

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

Domain Name System Configuration

Start > Administrative Tools > DNS

Forward Lookup Zone

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
LYNC2010MEDPRIM	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

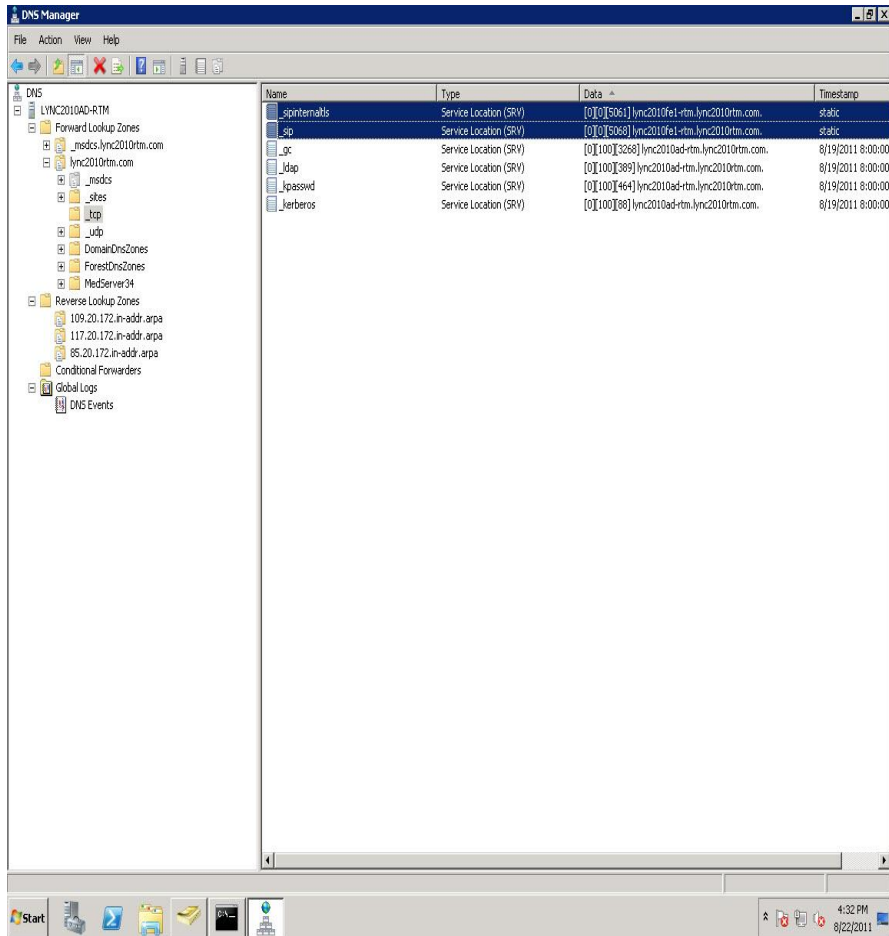
The screenshot shows the DNS Manager console with the following data:

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



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



DNS Manager

File Action View Help

DNS

- LYNC2010AD-RTM
 - Forward Lookup Zones
 - _msdcs.lync2010rtm.com
 - lync2010rtm.com
 - _sites
 - _tcp
 - _udp
 - DomainDnsZones
 - ForestDnsZones
 - MedServer34
 - Reverse Lookup Zones
 - 109.20.172.in-addr.arpa
 - 117.20.172.in-addr.arpa
 - 85.20.172.in-addr.arpa
 - Conditional Forwarders
 - Global Logs
 - DNS Events

Name	Type	Data	Timestamp
_sipinternaltls	Service Location (SRV)	[0][0][5061] lync2010fe1-rtm.lync2010rtm.com.	static
_sip	Service Location (SRV)	[0][0][5068] lync2010fe1-rtm.lync2010rtm.com.	static
_gc	Service Location (SRV)	[0][100][3268] lync2010ad-rtm.lync2010rtm.com.	8/19/2011 8:00:00
_ldap	Service Location (SRV)	[0][100][389] lync2010ad-rtm.lync2010rtm.com.	8/19/2011 8:00:00
_kpasswd	Service Location (SRV)	[0][100][464] lync2010ad-rtm.lync2010rtm.com.	8/19/2011 8:00:00
_kerberos	Service Location (SRV)	[0][100][88] lync2010ad-rtm.lync2010rtm.com.	8/19/2011 8:00:00

_sip Properties

Service Location (SRV) Security

Domain: lync2010rtm.com

Service: _sip

Protocol: _tcp

Priority: 0

Weight: 0

Port number: 5068

Host offering this service:
lync2010fe1-rtm.lync2010rtm.com.

OK Cancel Apply Help

Start

4:34 PM
8/22/2011



DNS Manager

File Action View Help

DNS

- LYNC2010AD-RTM
 - Forward Lookup Zones
 - _msdcs.lync2010rtm.com
 - lync2010rtm.com
 - _sites
 - _tcp
 - 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 | 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 interface. The left pane shows the site hierarchy with the following components highlighted by red boxes:

- LyncRTM** (Site)
- Standard Edition Front End Servers** (Folder)
- LYNC2010FE1-RTM.lync2010rtm.com** (Server)
- Mediation pools** (Folder)
- LYNC2010FE1-RTM.lync2010rtm.com** (Server)
- 172.20.109.254** (IP Address)

The right pane displays the configuration for the **LyncRTM** site:

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 |
|-------------|----------|

The rightmost pane shows the **Actions** menu for the selected site, with the following options:

- New
- Edit Properties...
- Topology
- View
- Delete
- 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
 - 172.20.109.254
 - Monitoring Servers
 - Archiving Servers
 - Edge pools
 - Trusted application servers
 - Branch sites

General

FQDN: LYNC2010FE1-RTM.lync2010rtm.com

IP addresses: Use all configured

Features and functionality

| | |
|---------------------------------|---------|
| Instant messaging and presence: | Enabled |
| Conferencing: | Enabled |
| PSTN conferencing: | Enabled |
| Enterprise Voice: | Enabled |

Associations

SQL store: LYNC2010FE1-RTM.lync2010rtm.com\rtc

File store: \\LYNC2010FE1-RTM.lync2010rtm.com\share

Archiving Server: Not associated

Monitoring Server: Not associated

Edge pool (for media): Not associated

Note: To view the federation route, use the site property page.

Resiliency

Associated backup Registrar pool: Not configured

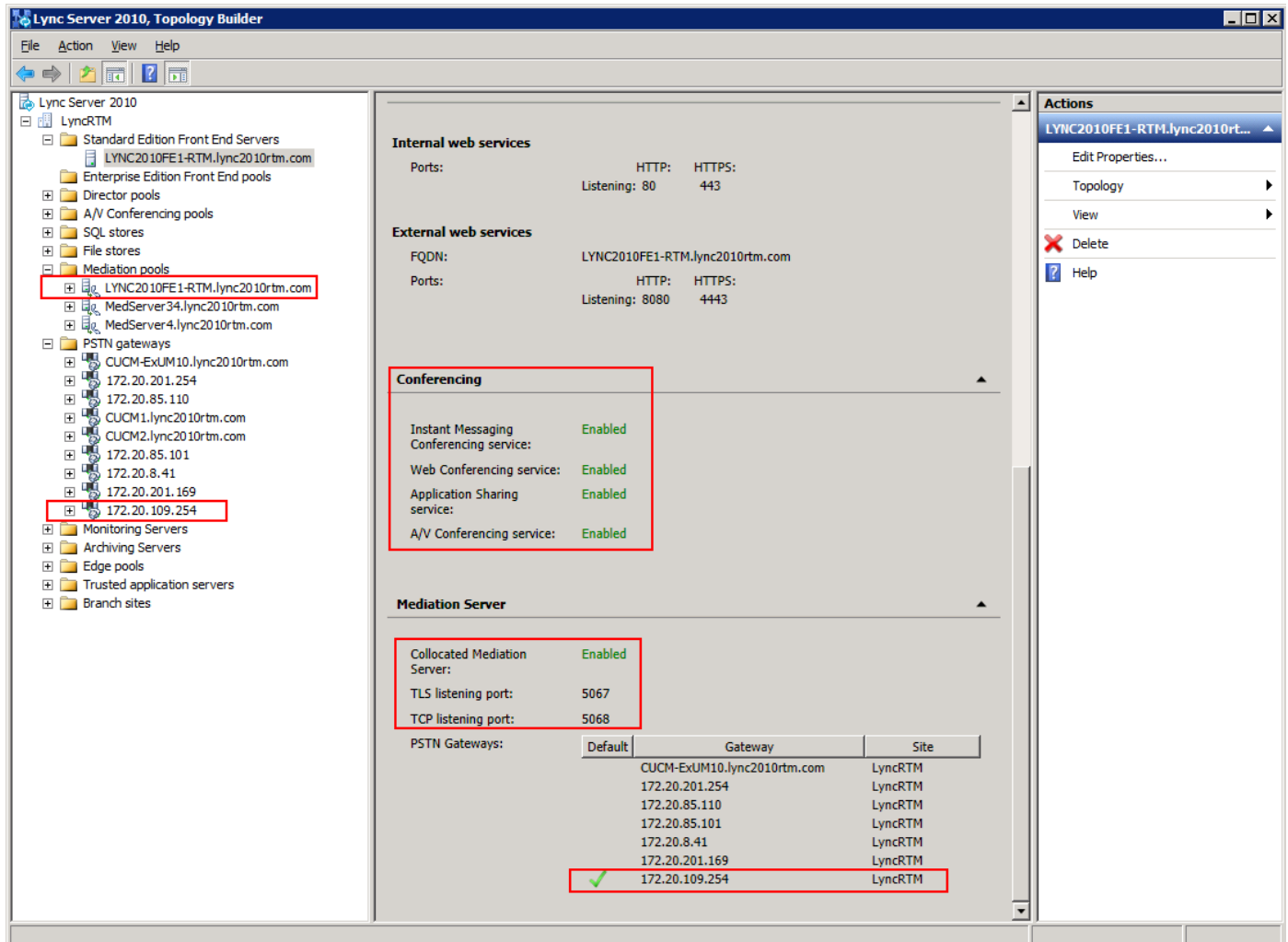
Failover and Failback: Disabled

Web services

Actions

LYNC2010FE1-RTM.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)



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

Lync Server 2010, Topology Builder

File Action View Help

Lync Server 2010

- LyncRTM
 - Standard Edition Front End Servers
 - LYNC2010FE1-RTM.lync2010rtm.com
 - Enterprise Edition Front End pools
 - Director pools
 - A/V Conferencing pools
 - SQL stores
 - File stores
 - Mediation pools
 - LYNC2010FE1-RTM.lync2010rtm.com**
 - MedServer34.lync2010rtm.com
 - MedServer4.lync2010rtm.com
 - PSTN gateways
 - CUCM-ExUM10.lync2010rtm.com
 - 172.20.201.254
 - 172.20.85.110
 - CUCM1.lync2010rtm.com
 - CUCM2.lync2010rtm.com
 - 172.20.85.101
 - 172.20.8.41
 - 172.20.201.169
 - 172.20.109.254
 - Monitoring Servers
 - Archiving Servers
 - Edge pools
 - Trusted application servers
 - Branch sites

Mediation Server PSTN gateway

TLS listening port: 5067

TCP listening port: 5068

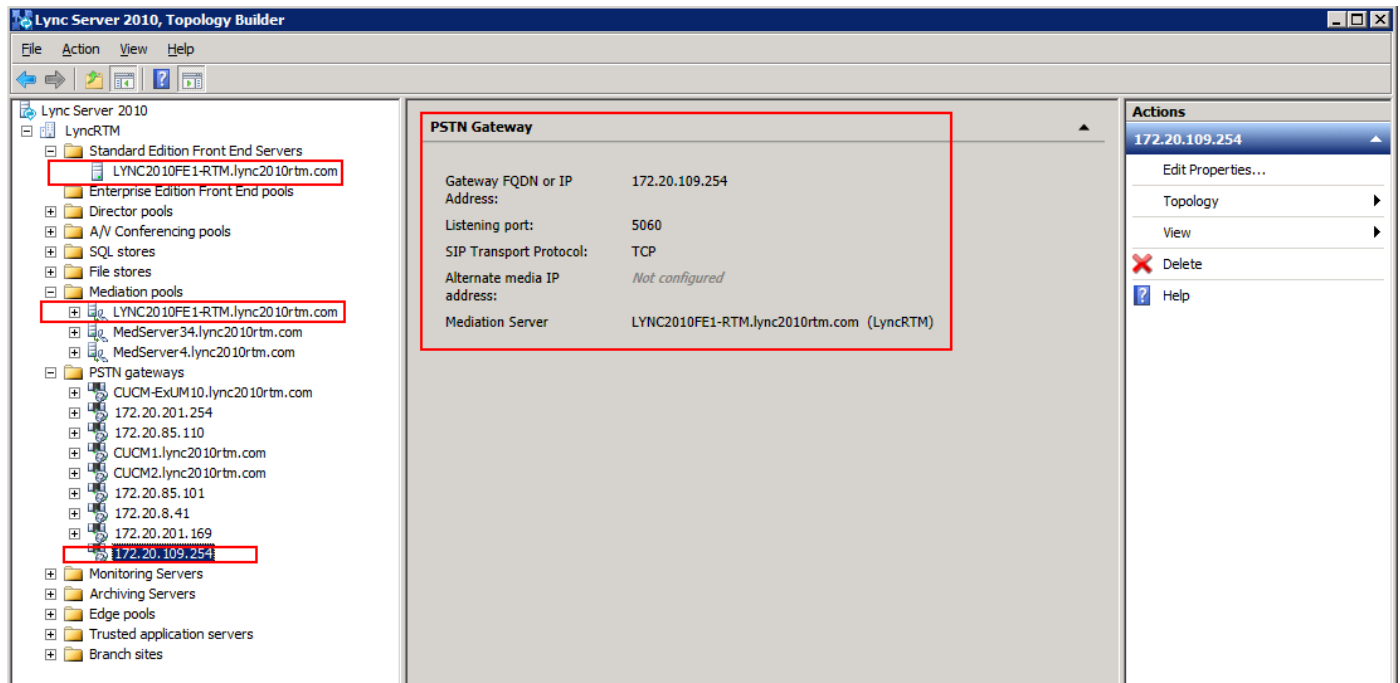
PSTN Gateways:

| Default | Gateway |
|---------|-----------------------------|
| | CUCM-ExUM10.lync2010rtm.com |
| | 172.20.201.254 |
| | 172.20.85.110 |
| | 172.20.85.101 |
| | 172.20.8.41 |
| | 172.20.201.169 |
| ✓ | 172.20.109.254 |

Actions

LYNC2010FE1-RTM.lync2010rtm...

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

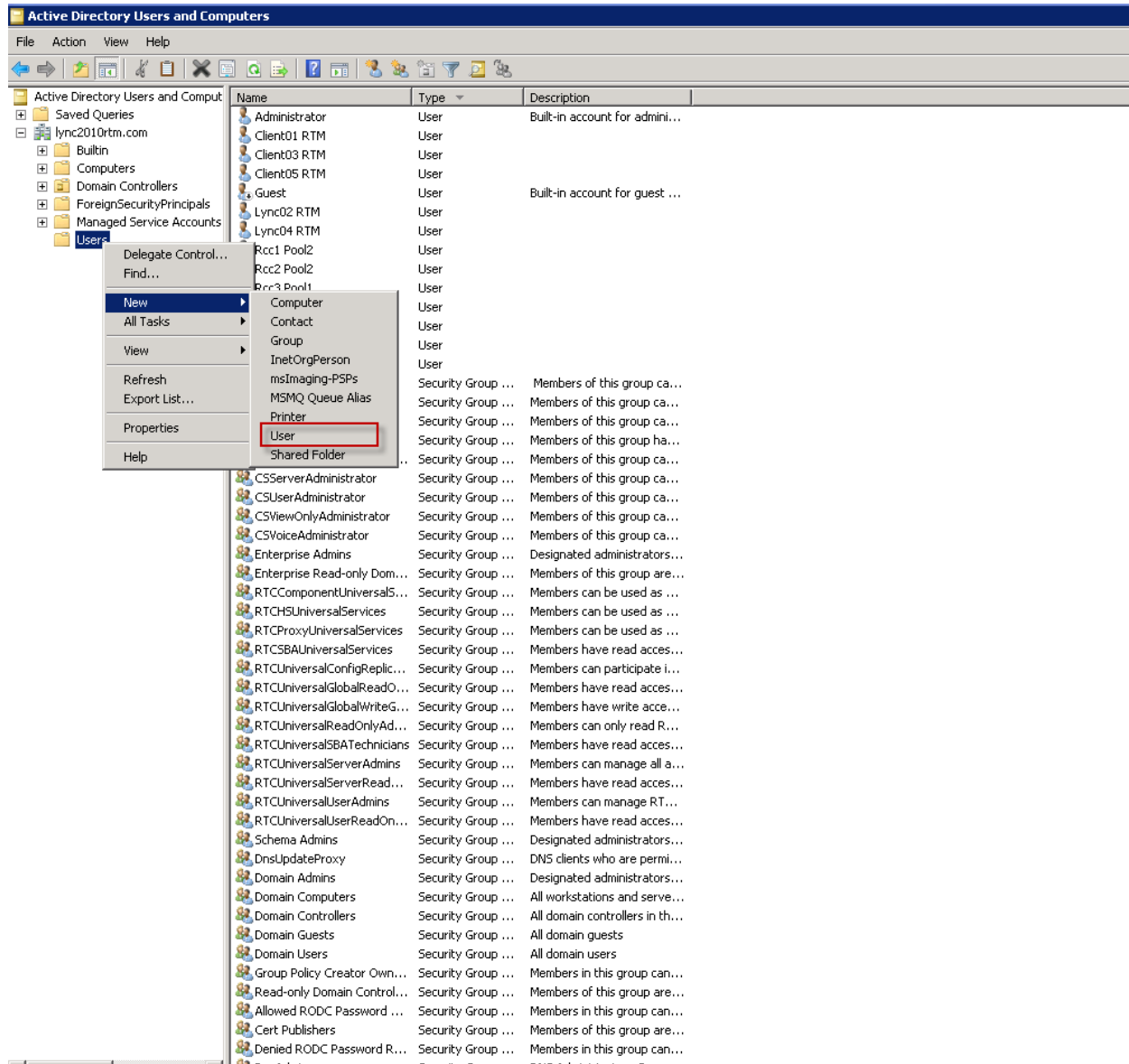


Listening port of the SME is left to default 5060.



Active Directory User Configuration

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





| Active Directory Users and Computers | | | |
|---|------------------------------------|--------------------|--------------------------------|
| File Action View Help | | | |
| Active Directory Users and Computers [LYNC2010] | | | |
| Active Directory Users and Computers [LYNC2010] | Name | Type | Description |
| lync2010rtm.com | Administrator | User | Built-in account for admini... |
| Builtin | Client01 RTM | User | |
| Computers | Client03 RTM | User | |
| Domain Controllers | Client05 RTM | User | |
| ForeignSecurityPrincipals | Guest | User | Built-in account for guest ... |
| Managed Service Accounts | Lync PC1 | User | |
| Users | Lync PC2 | User | |
| | Lync02 RTM | User | |
| | Lync04 RTM | User | |
| | Rcc1 Pool2 | User | |
| | Rcc2 Pool2 | User | |
| | Rcc3 Pool1 | User | |
| | User 1 Pool1 | User | |
| | User 2 Pool1 | User | |
| | CSAdministrator | Security Group ... | Members of this group ca... |
| | CSArchivingAdministrator | Security Group ... | Members of this group ca... |
| | CSHelpDesk | Security Group ... | Members of this group ca... |
| | CSLocationAdministrator | Security Group ... | Members of this group ha... |
| | CSResponseGroupAdministrator | Security Group ... | Members of this group ca... |
| | CSServerAdministrator | Security Group ... | Members of this group ca... |
| | CSUserAdministrator | Security Group ... | Members of this group ca... |
| | CSViewOnlyAdministrator | Security Group ... | Members of this group ca... |
| | CSVoiceAdministrator | Security Group ... | Members of this group ca... |
| | Enterprise Admins | Security Group ... | Designated administrators... |
| | Enterprise Read-only Domain Con... | Security Group ... | Members of this group are... |
| | RTCComponentUniversalServices | Security Group ... | Members can be used as ... |
| | RTCHSUniversalServices | Security Group ... | Members can be used as ... |
| | RTCPProxyUniversalServices | Security Group ... | Members can be used as ... |
| | RTCSBAUniversalServices | Security Group ... | Members have read acces... |
| | RTCUniversalConfigReplicator | Security Group ... | Members can participate i... |
| | RTCUniversalGlobalReadOnlyGroup | Security Group ... | Members have read acces... |
| | RTCUniversalGlobalWriteGroup | Security Group ... | Members have write acces... |
| | RTCUniversalReadOnlyAdmins | Security Group ... | Members can only read R... |
| | RTCUniversalSBATechnicians | Security Group ... | Members have read acces... |
| | RTCUniversalServerAdmins | Security Group ... | Members can manage all a... |
| | RTCUniversalServerReadOnlyGroup | Security Group ... | Members have read acces... |
| | RTCUniversalUserAdmins | Security Group ... | Members can manage RT... |
| | RTCUniversalUserReadOnlyGroup | Security Group ... | Members have read acces... |
| | Schema Admins | Security Group ... | Designated administrators |



Active Directory Users and Computers [LYNC2010rtm.com]

File Action View Help

Active Directory Users and Computers [LYNC2010rtm.com]

- Saved Queries
- lync2010rtm.com
 - Builtin
 - Computers
 - Domain Controllers
 - ForeignSecurityPrincipals
 - Managed Service Accounts
 - Users

| Name | Type | Description |
|---------------------------------|--------------------|--------------------------------|
| Administrator | User | Built-in account for admini... |
| Client01 RTM | User | |
| Client03 RTM | User | |
| Client05 RTM | User | |
| Guest | User | |
| Lync PC1 | | |
| Lync PC2 | | |
| Lync02 RTM | | |
| Lync04 RTM | | |
| Rcc1 Pool2 | | |
| Rcc2 Pool2 | | |
| Rcc3 Pool1 | | |
| User 1 Pool1 | | |
| User 2 Pool1 | | |
| CSAdministrator | | |
| CSArchivingAdministrator | | |
| CSHelpDesk | | |
| CSLocationAdministrator | | |
| CSResponseGroupAdministrator | | |
| CSServerAdministrator | | |
| CSUserAdministrator | | |
| CSViewOnlyAdministrator | | |
| CSVoiceAdministrator | | |
| Enterprise Admins | | |
| Enterprise Read-only Domain Co | | |
| RTCComponentUniversalService | | |
| RTCHSUniversalServices | | |
| RTCPProxyUniversalServices | | |
| RTCSBAUniversalServices | | |
| RTCUniversalConfigReplicator | | |
| RTCUniversalGlobalReadOnlyGr | | |
| RTCUniversalGlobalWriteGroup | | |
| RTCUniversalReadOnlyAdmins | | |
| RTCUniversalSBATechnicians | | |
| RTCUniversalServerAdmins | | |
| RTCUniversalServerReadOnlyGroup | Security Group ... | Members have read acces... |
| RTCUniversalUserAdmins | Security Group ... | Members can manage RT... |
| RTCUniversalUserReadOnlyGroup | Security Group ... | Members have read acces... |
| Schema Admins | Security Group | Designated administrators |

Lync04 RTM Properties

Dial-in | Environment | Sessions | Remote control

Remote Desktop Services Profile | Personal Virtual Desktop | COM+

General | Address | Account | Profile | Telephones | Organization | Member Of

User logon name:
Lync04 @lync2010rtm.com

User logon name (pre-Windows 2000):
LYNC2010RTM\Lync04

Logon Hours... Log On To...

☐ Unlock account

Account options:

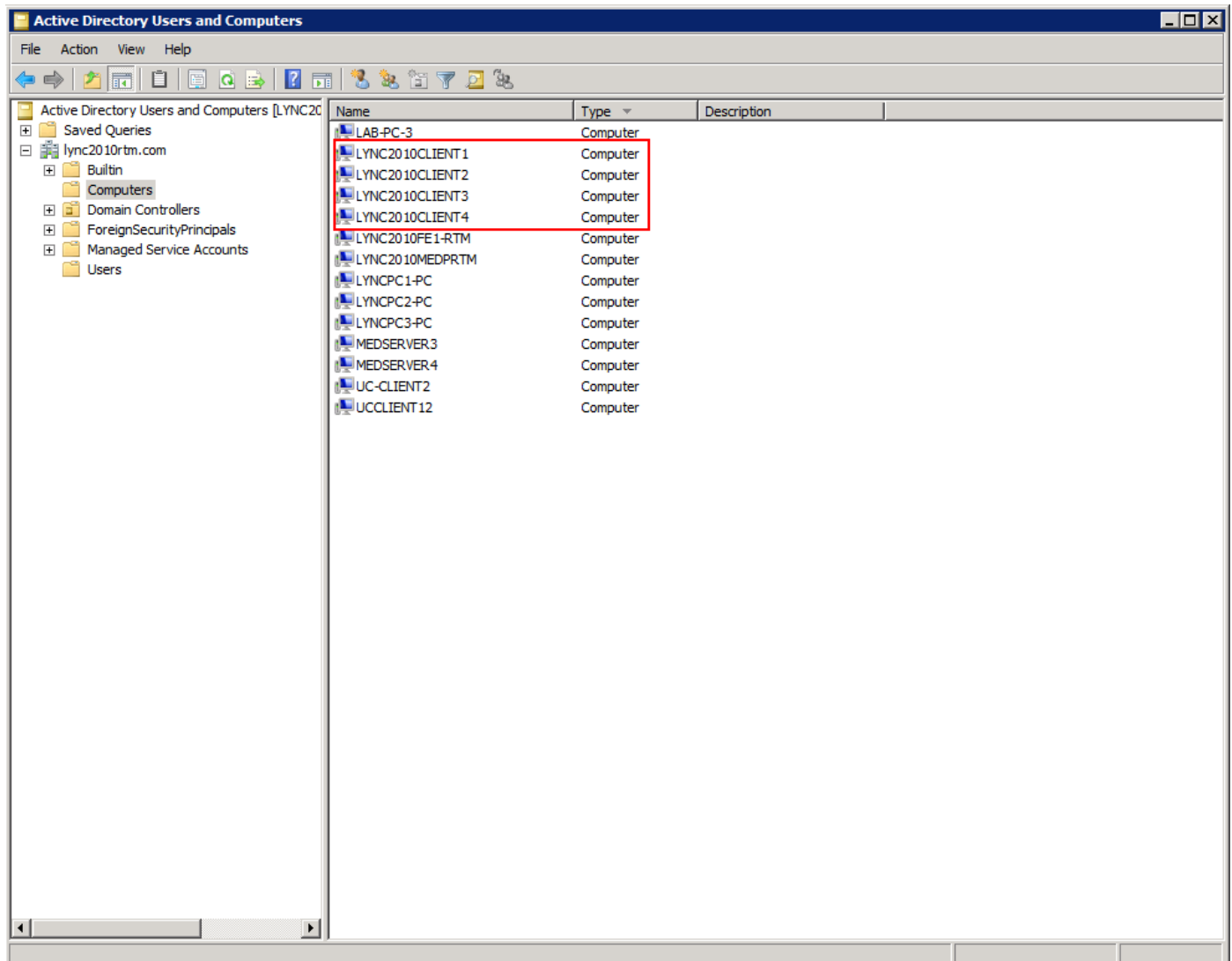
- ☐ User must change password at next logon
- ☒ User cannot change password
- ☒ Password never expires
- ☐ Store password using reversible encryption

Account expires:
☒ Never
☐ End of: Sunday, November 25, 2012

OK Cancel Apply Help



Computers on the Active Directory running LYNC clients





Lync Server 2010 Configuration

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

User Configuration from Control Panel

Start > All Programs > Microsoft Lync Server 2010 > Lync Server Control Panel

To add users go to Home > Enable users for Lync Server and then go to Users > Add and then do a find as shown below.

Microsoft Lync Server 2010 Control Panel

Administrator | Sign out
4.0.7577.0

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

User Search

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 |



Microsoft Lync Server 2010 Control Panel

Administrator | Sign out 4.0.7577.0

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

User Search

Edit Lync Server User - Lync04 RTM

Display name:
Lync04 RTM

☒ **Enabled for Lync Server**

SIP address:*
sip:Lync04 @ lync2010rtm.com

Registrar pool:
LYNC2010FE1-RTM.lync2010rtm.com ?

Telephony:
Enterprise Voice ?

Line URI:
tel:+14152221014 ?

Dial plan policy:
<Automatic> View...

Voice policy:
<Automatic> View...

Conferencing policy:
<Automatic> View...

Client version policy:
<Automatic> View...

PIN policy:
<Automatic> View...

Network Configuration

PIN policy:
<Automatic> View...

External access policy:
<Automatic> View...

Archiving policy:
<Automatic> View...

Location policy:
<Automatic> View...

Client policy:
<Automatic>



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 Control Panel

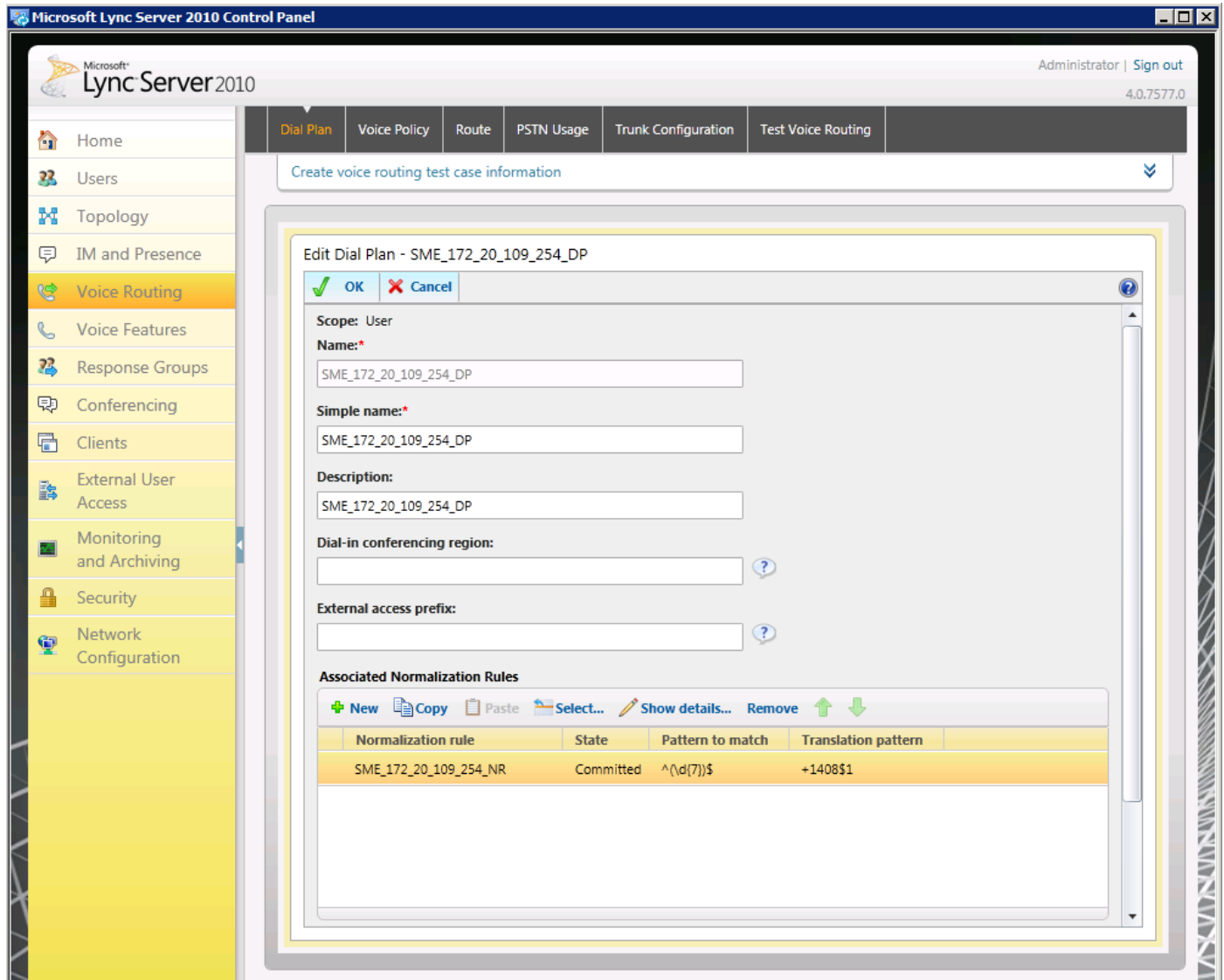
Administrator | Sign out 4.0.7577.0

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

Create voice routing test case information

| Name | Scope | State | Normalization rules | Description |
|-----------------------|--------|-----------|---------------------|-----------------------------|
| Global | Global | Committed | 1 | used to dial to Aparna CUCM |
| SME_172_20_109_254_DP | User | Committed | 1 | SME_172_20_109_254_DP |
| SME_DialPlan | User | Committed | 2 | Use to Dial to SME |
| UserDialPlan | User | Committed | 6 | Dial Plan for CUCM-ExUM10 |

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



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

Click on the Normalization Rule “SME_172_20_109_254_NR” in the above figure.



Microsoft Lync Server 2010 Control Panel

Microsoft Lync Server 2010 Administrator | Sign out 4.0.7577.0

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

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

Create voice routing test case information

Edit Dial Plan Edit Normalization Rule - SME_172_20_109_254_NR

OK Cancel

Name: SME_172_20_109_254_NR

Description: SME_172_20_109_254_NR

Build a Normalization Rule

Fill in the fields that you want to use, or create the rule manually by clicking Edit.

Starting digits:

Length: Exactly 7

Digits to remove: 0

Digits to add: +1408

Pattern to match: ^(\d{7})\$

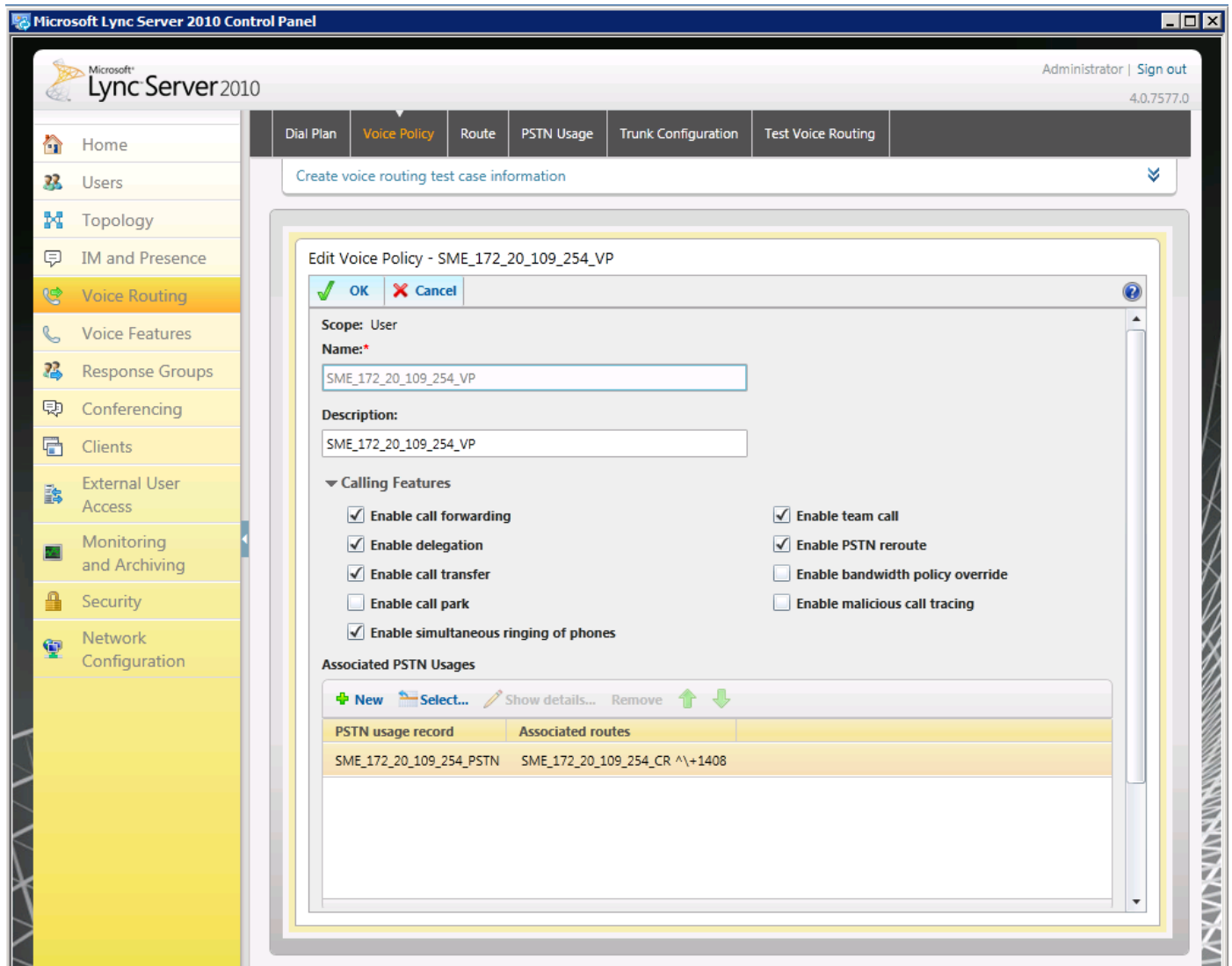
Translation rule: +1408\$1

Edit Reset ?

☐ Internal extension ?

Dialed number to test:

Go



Microsoft Lync Server 2010 Control Panel

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_172_20_109_254_VP

OK Cancel

Scope: User

Name: SME_172_20_109_254_VP

Description: SME_172_20_109_254_VP

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 |
|-------------------------|-------------------------------|
| SME_172_20_109_254_PSTN | SME_172_20_109_254_CR ^\+1408 |

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

Microsoft Lync Server 2010 Control Panel

Administrator | Sign out 4.0.7577.0

Microsoft Lync Server 2010

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

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

Create voice routing test case information

Edit Voice Route - SME_172_20_109_254_CR

OK Cancel

Name: SME_172_20_109_254_CR

Description: SME_172_20_109_254_CR

Build a Pattern to Match

Add the starting digits that you want this route to handle, or create the expression manually by clicking Edit.

Starting digits for numbers that you want to allow:

Type a valid number and then click Add.

+1408

Exceptions Remove

Match this pattern:

^\+1408

Edit Reset ?

☐ Suppress caller ID

Alternate caller ID:

Microsoft Lync Server 2010 Control Panel

Administrator | Sign out | 4.0.7577.0

Microsoft Lync Server 2010

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

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

Create voice routing test case information

Edit Voice Route - SME_172_20_109_254_CR

OK Cancel

☐ Suppress caller ID

Alternate caller ID:

Associated gateways:

PstnGateway:172.20.109.254 Add... Remove

Associated PSTN Usages

Select... Remove ↑ ↓

| PSTN usage record | Associated voice policies |
|-------------------------|---------------------------|
| SME_172_20_109_254_PSTN | SME_172_20_109_254_VP |

Translated number to test:

Go



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

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

The screenshot shows the Microsoft Lync Server 2010 Control Panel interface. The left sidebar contains navigation links: Home, Users, Topology, IM and Presence, Voice Routing (selected), Voice Features, Response Groups, Conferencing, Clients, External User Access, Monitoring and Archiving, Security, and Network Configuration. The main area displays the 'Trunk Configuration' tab for 'PstnGateway:172.20.109.254'. The configuration includes a 'Name' field with the same value, 'Maximum early dialogs supported' set to 20, and 'Encryption support level' set to 'Optional'. Checkboxes for 'Enable media bypass', 'Centralized media processing' (checked), and 'Enable refer support' are visible. Below, the 'Associated Translation Rules' section shows a table with one rule: 'to SME' with a 'Committed' state, matching pattern '^(\d{7})\$', and translation pattern '+1408\$1'.

| Translation rule | State | Pattern to match | Translation pattern |
|------------------|-----------|------------------|---------------------|
| to SME | Committed | ^\d{7}\$ | +1408\$1 |



Lync 2010 Server Management Shell Commands

Lync 2010 Server draining mode

Stop-CsWindowsService -Graceful rtcmedsrv

Lync2010 - enable music on hold:

set-csclientpolicy -EnableClientMusicOnHold \$TRUE

Some Lync2010 useful commands

Get-CsTrunkConfiguration

Set-CsTrunkConfiguration

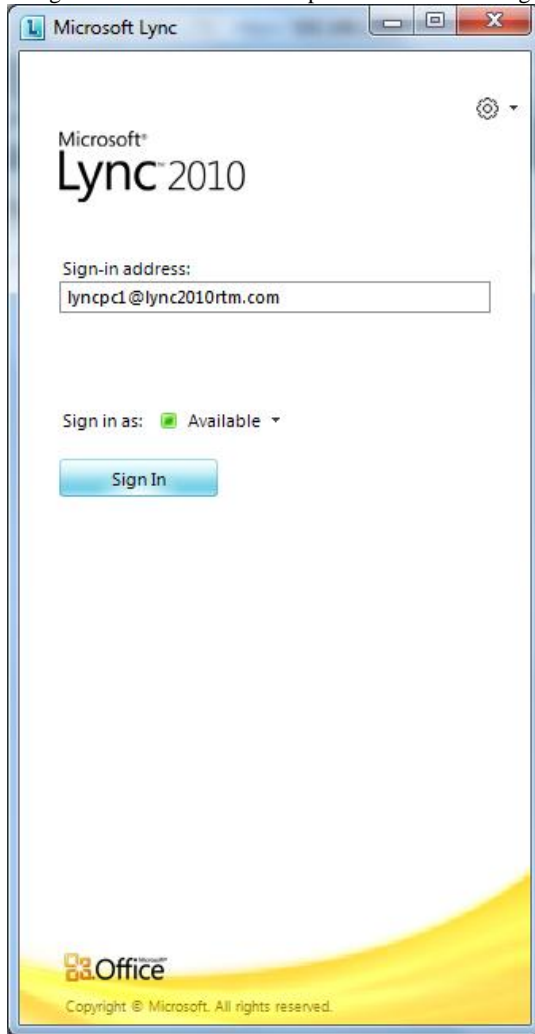
Get-CsMediaConfiguration

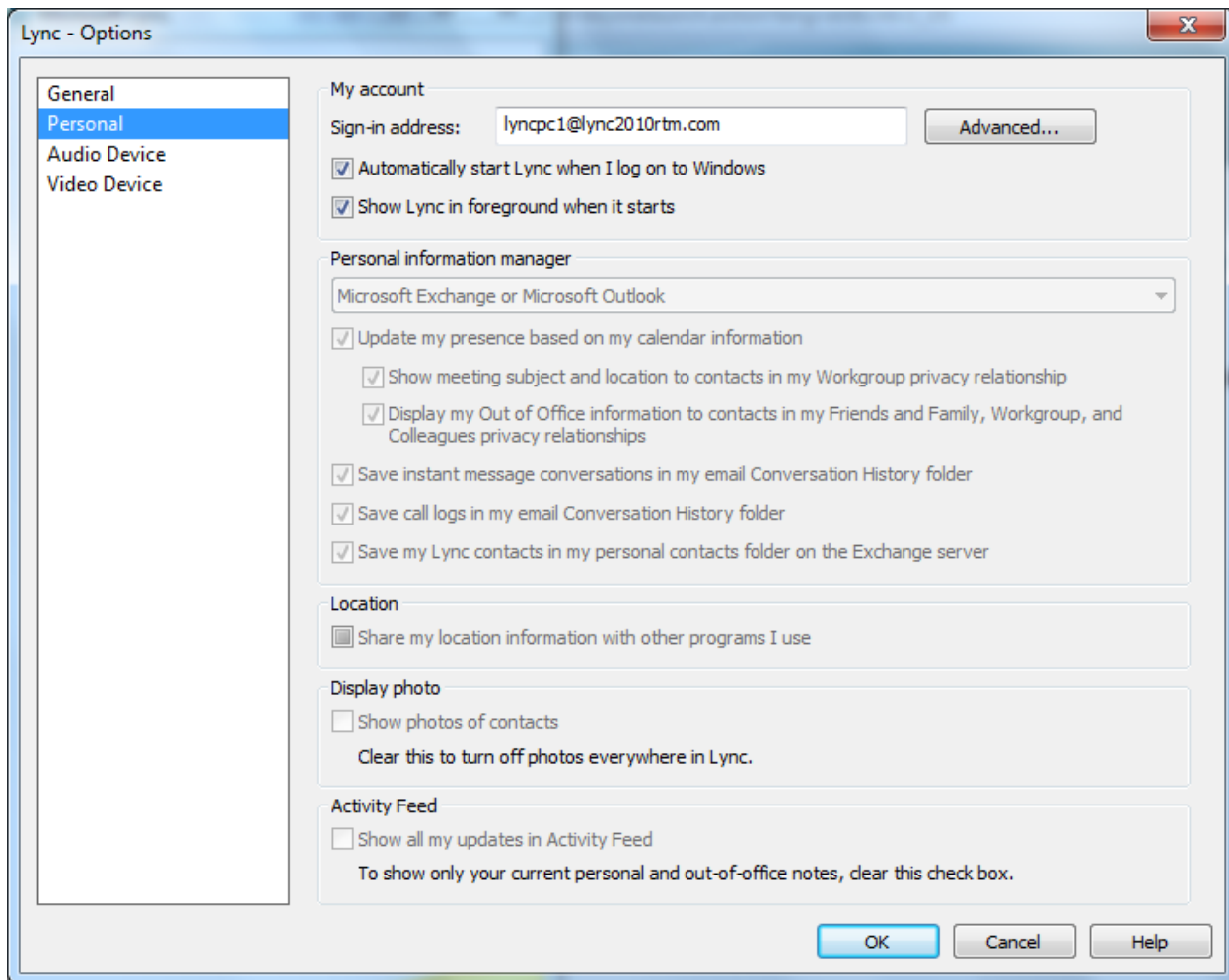
Set-CsMediaConfiguration



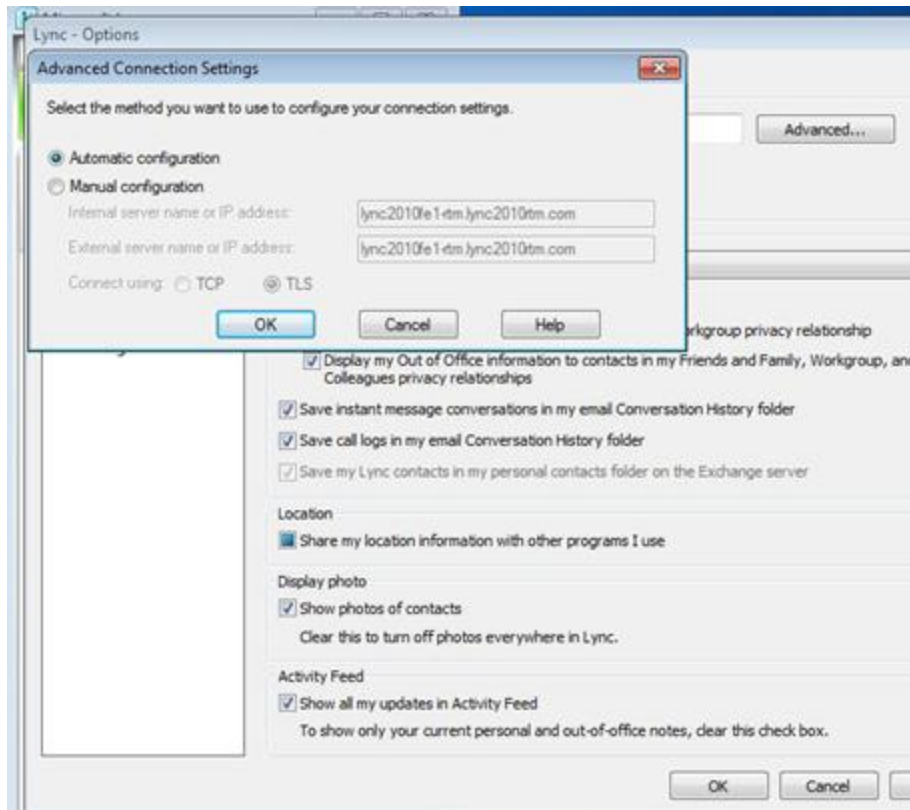
Microsoft Lync 2010 Enterprise Voice Client Configuration

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





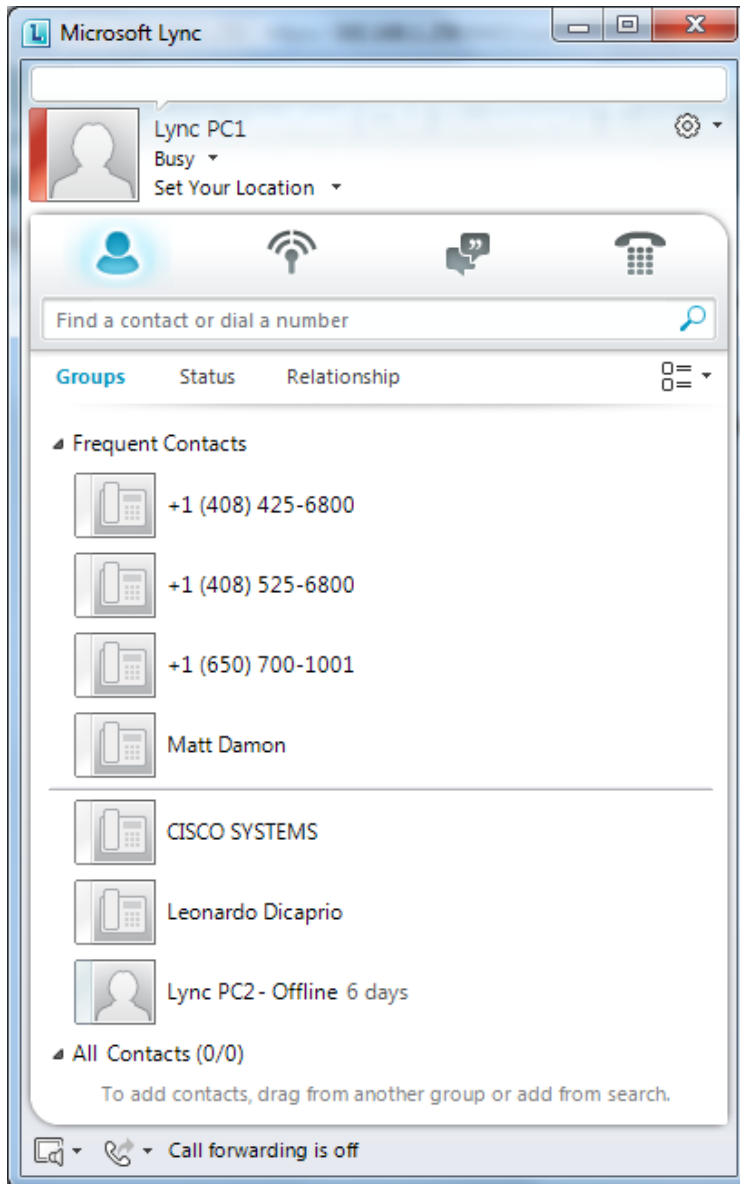
Click Advanced button to select the Advanced Connection Settings.



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 - Options

General
Personal
Status
My Picture
Phones
Alerts
Ringtones and Sounds
Audio Device
Video Device
Call Forwarding
File Saving

My account
Sign-in address: [Advanced...](#)
☒ Automatically start Lync when I log on to Windows
☒ Show Lync in foreground when it starts

Personal information manager

☒ Update my presence based on my calendar information
☒ Show meeting subject and location to contacts in my Workgroup privacy relationship
☒ Display my Out of Office information to contacts in my Friends and Family, Workgroup, and Colleagues privacy relationships
☒ Save instant message conversations in my email Conversation History folder
☒ Save call logs in my email Conversation History folder
☒ Save my Lync contacts in my personal contacts folder on the Exchange server

Location
☒ Share my location information with other programs I use

Display photo
☒ Show photos of contacts
Clear this to turn off photos everywhere in Lync.

Activity Feed
☒ Show all my updates in Activity Feed
To show only your current personal and out-of-office notes, clear this check box.

OK Cancel Help

Lync - Options

General
Personal
Audio Device
Video Device

Audio device
Select the device you want to use for audio calls: [Learn More](#)
 Your computer's default setup

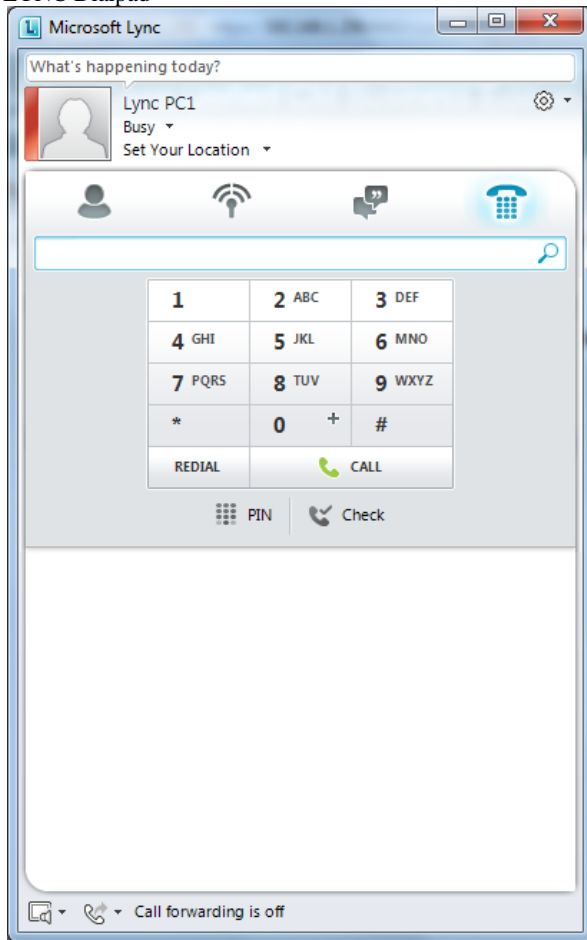
Customize your device
Speaker:
Microphone:
Ringer:

Secondary ringer
☐ Also ring:
☐ Unmute when my phone rings

OK Cancel Help

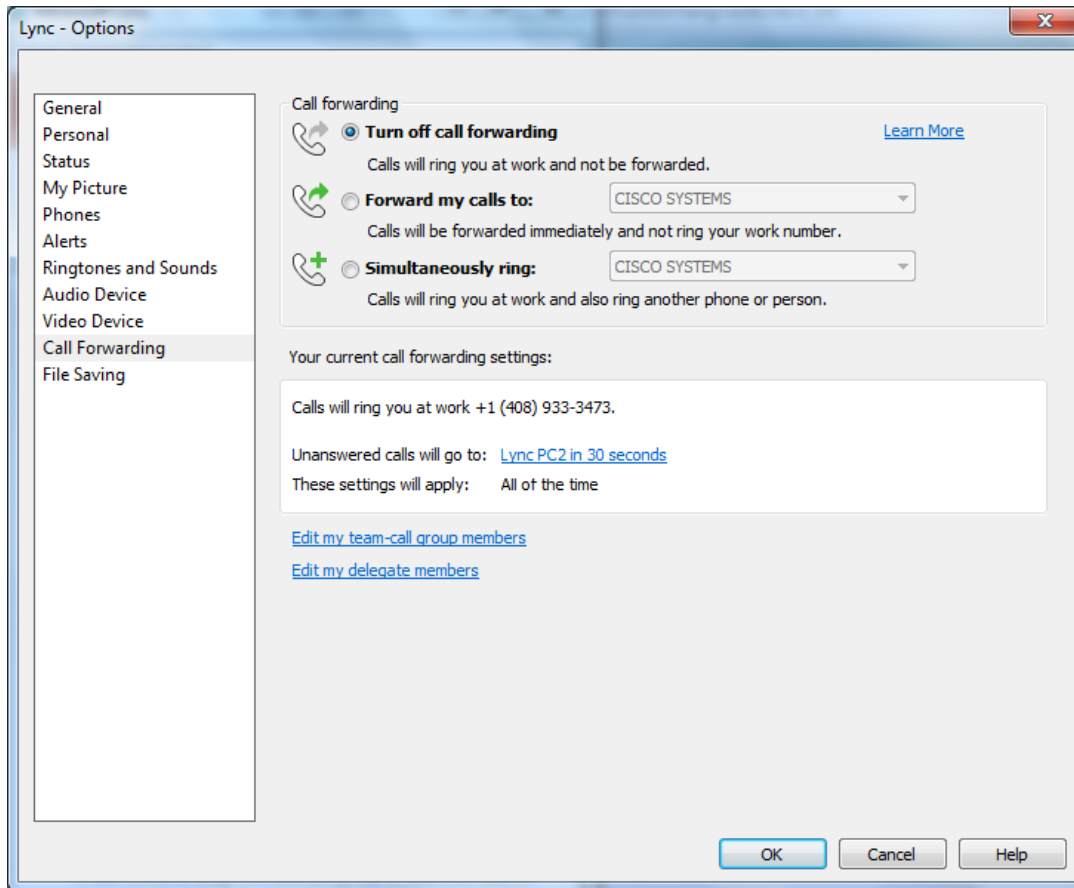


LYNC Dialpad





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-SME Software release

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: [Cisco Unified CM Administration](#) | [CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Cisco Unified CM Administration
System version: 9.0.1.10000-37

Last Successful Logon: Thursday, October 18, 2012 12:46:23 PM PDT

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All rights reserved.

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 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 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-SME Region Configuration

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 ▾

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

Save Delete Reset Apply Config Add New

Region Information

Name*

Region Relationships

| Region | Audio Codec Preference List | Maximum Audio Bit Rate | Maximum Session Bit Rate for Video Calls |
|-----------------|-----------------------------|------------------------|--|
| Region_SME_G711 | Custom Codec List G711 | 64 kbps (G.722, G.711) | 384 |
| Region_SME_G729 | Custom Codec List G711 | 64 kbps (G.722, G.711) | 384 |

NOTE: Regions not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

| Regions | Audio Codec Preference List | Maximum Audio Bit Rate | Maximum Session Bit Rate for Video Calls |
|--|-----------------------------------|-----------------------------------|---|
| <div>ATT_ASR
Default
Region_SME_G711
Region_SME_G729</div> | <div>Keep Current Setting ▾</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.

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



Cisco Unified Communications Manager-SME Device Pool Configuration

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


Navigation [Cisco Unified CM Administration](#)

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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
 4 records found

Device Pool (1 - 4 of 4) Rows per Page

Find Device Pool where ▾ begins with ▾

| <input type="checkbox"/> | Name ^ | Cisco Unified CM Group | Region | Date/Time Group | |
|--------------------------|-------------------------|-------------------------|---------------------------------|-------------------------|---|
| <input type="checkbox"/> | ATT_ASR | Default | ATT_ASR | CMLocal |  |
| <input type="checkbox"/> | Default | Default | Default | CMLocal |  |
| <input type="checkbox"/> | G711-DP | Default | Region_SME_G711 | CMLocal |  |
| <input type="checkbox"/> | G729-DP | Default | Region_SME_G729 | CMLocal |  |



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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

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

Save Delete Copy Reset Apply Config Add New

Status
 Status: Ready

Device Pool Information
Device Pool: G711-DP (4 members**)

Device Pool Settings
Device Pool Name*
Cisco Unified Communications Manager Group*
Calling Search Space for Auto-registration
Adjunct CSS
Reverted Call Focus Priority
Local Route Group
Intercompany Media Services Enrolled Group

Roaming Sensitive Settings
Date/Time Group*
Region*
Media Resource Group List
Location
Network Locale
SRST Reference*
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 configured is used as the prefix unless field is empty in which case there is no prefix assigned.

[Clear Prefix Settings](#) [Default Prefix Settings](#)

| Number Type | Prefix | Strip Digits | Calling Search Space |
|----------------------|--------------------------------------|----------------------|---|
| National Number | <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 as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings

Default Prefix Settings

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

Phone Settings

Inbound Call Settings

Calling Party Transformation CSS

Connected Party Settings

Connected Party Transformation CSS

Redirecting Party Settings

Redirecting Party Transformation CSS



*- indicates required item.



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



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



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

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



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

Cisco Unified CM Administration
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Navigation [Cisco Unified CM Administra](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Device Pool Configuration

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

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

Device Pool Information

Device Pool: G729-DP (4 members**)

Device Pool Settings

Device Pool Name*

Cisco Unified Communications Manager Group*

Calling Search Space for Auto-registration

Adjunct CSS

Reverted Call Focus Priority

Local Route Group

Intercompany Media Services Enrolled Group

Roaming Sensitive Settings

Date/Time Group*

Region*

Media Resource Group List

Location

Network Locale

SRST Reference*

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 configured is used as the prefix unless field is empty in which case there is no prefix assigned.

Clear Prefix Settings

Default Prefix Settings

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

Clear Prefix Settings

Default Prefix Settings

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

Phone Settings

Inbound Call Settings

Calling Party Transformation CSS

Connected Party Settings

Connected Party Transformation CSS

Redirecting Party Settings

Redirecting Party Transformation CSS



*- indicates required item.



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



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



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



Cisco Unified Communications Manager-SME Media Termination Point Configuration

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Navigation Cisco Unified CM Administr...

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

CCMAdministrator | Search Documentation | About

Find and List Media Termination Points

Status
 2 records found

Media Termination Point (1 - 2 of 2) Rows per Page

Find Media Termination Point where Name ▾ begins with ▾ Find Clear Filter

| <input type="checkbox"/> | Name | Description | Device Pool | Status | IP Address | Co |
|--------------------------|--------------------------|---------------------|--------------------|---------------------------|----------------|-------------|
| <input type="checkbox"/> | MTP_2
mtp00233335da20 | MTP_CUCM-SME-Telugu | G711-DP
Default | Registered with CM-Telugu | 172.20.109.254 | Not Allowed |

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Navigation Cisco Unified CM Administrat...

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

CCMAdministrator | Search Documentation | About

Media Termination Point Configuration

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

Status
 Status: Ready

Media Termination Point Information

Registration Registered with Cisco Unified Communications Manager CM-Telugu

IP Address 172.20.109.254

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

Media Termination Point Type* Cisco Media Termination Point Software

Host Server* CM-Telugu

Media Termination Point Name* MTP_2

Description MTP_CUCM-SME-Telugu

Device Pool* G711-DP

☒ Trusted Relay Point

*. indicates required item.



Cisco Unified Communications Manager-SME Transcoder Configuration

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Navigation [Cisco Unified CM Administration](#)

CCMAdministrator | [Search Documentation](#) | [About](#)

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

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 ▾

| <input type="checkbox"/> | Name | Description | Device Pool | Status | IP Address |
|--------------------------|-----------------|-------------|-------------------------|---------------------------|----------------|
| <input type="checkbox"/> | mtp0123456789ab | XCODE-ExtGW | Default | Registered with CM-Telugu | 172.20.109.201 |

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Navigation [Cisco Unified CM Administration](#)

CCMAdministrator | [Search Documentation](#) | [About](#)

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

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

Save

Delete

Copy

Reset

Apply Config

Add New

Transcoder Information**Transcoder:** mtp0123456789ab (XCODE-ExtGW)
Registration Registered with Cisco Unified Communications Manager CM-Telugu
IP Address 172.20.109.201
IPv6 Address 0000:0000:0000:0000:0000:0000:0000:0000**IOS Transcoder Info**Transcoder Type* Cisco IOS Enhanced Media Termination Point
Description XCODE-ExtGW
Device Name* mtp0123456789ab
Device Pool* [Default](#) [View Details](#)
Common Device Configuration < None > [View Details](#)
Special Load Information Leave blank to use default
☒ Trusted Relay Point

*- indicates required item.



Cisco Unified Communications Manager-SME Media Resource Groups configuration

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Navigation: Cisco Unified CM Administration | CCMAdministrator | Search Documentation | About

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

Find and List Media Resource Groups

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

Status
7 records found

Media Resource Group (1 - 7 of 7) Rows per Page

Find Media Resource Group where Name begins with Find Clear Filter

| <input type="checkbox"/> | Name ^ | Description | Multi-cast | |
|--------------------------|--------------------------------|--------------------|------------|--|
| <input type="checkbox"/> | Conf_MRG | | false | |
| <input type="checkbox"/> | Ext_MRG | | false | |
| <input type="checkbox"/> | LYNC-MRG | | false | |
| <input type="checkbox"/> | MRG_IOS_GW_CFB | MRG for IOS GW CFB | false | |
| <input type="checkbox"/> | SIP_TRUNK_MRG | | false | |
| <input type="checkbox"/> | SME-MRG | | false | |
| <input type="checkbox"/> | SoftwareMTP | SW and HW MTP | false | |

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

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Navigation: Cisco Unified CM Administration | CCMAdministrator | Search Documentation | About

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

Media Resource Group Configuration

Related Links: Back To Find/List

Save | Delete | Copy | + Add New

Status
Status: Ready

Media Resource Group Status
Media Resource Group: 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
MOH_2
cfb001122334455
cfb00233335de20

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



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Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Media Resource Group Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Add New

Status
 Status: Ready

Media Resource Group Status
Media Resource Group: SIP_TRUNK_MRG (used by 0 devices)

Media Resource Group Information
Name*
Description

Devices for this Group
Available Media Resources**

CFB_2
MTP_2
cfb001122334455
mtp0123456789ab

Selected Media Resources*

ANN_2 (ANN)
MOH_2 (MOH)
cfb00233335da20 (CFB)
mtp00233335da20 (MTP)

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Save Delete Copy Add New

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



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Navigation [Cisco Unified CM Administra](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Media Resource Group Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Add New

Status
 Status: Ready

Media Resource Group Status
Media Resource Group: SME-MRG (used by 3 devices)

Media Resource Group Information
Name*
Description

Devices for this Group
Available Media Resources**

CFB_2
cfb001122334455
cfb00233335da20

▼ ▲

Selected Media Resources*

ANN_2 (ANN)
MOH_2 (MOH)
MTP_2 (MTP)
mtp00233335da20 (MTP)
mtp0123456789ab (XCODE)

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

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



Cisco Unified Communications Manager-SME Media Resource Group Lists configuration

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Navigation: Cisco Unified CM Administration | CCMAdministrator | Search Documentation | About

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

Find and List Media Resource Group Lists

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

Status
7 records found

Media Resource Group List (1 - 7 of 7)

Find Media Resource Group List where Name begins with Find Clear Filter

| <input type="checkbox"/> | Name ^ | Copy |
|--------------------------|---------------------------------|------|
| <input type="checkbox"/> | Conf_MRGL | |
| <input type="checkbox"/> | Ext_MRGL | |
| <input type="checkbox"/> | Internal-MRGL | |
| <input type="checkbox"/> | Lync-MRGL | |
| <input type="checkbox"/> | MRGL_IOS_GW_CFB | |
| <input type="checkbox"/> | SIP_Trunk_MRGL | |
| <input type="checkbox"/> | SoftwareMGL | |

Add New | Select All | Clear All | Delete Selected

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Navigation: Cisco Unified CM Administration | CCMAdministrator | Search Documentation | About

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

Media Resource Group List Configuration

Save | Delete | Copy | Add New

Status
Status: Ready

Media Resource Group List Status

Media Resource Group List: Internal-MRGL (used by 3 devices)

Media Resource Group List Information

Name*

Media Resource Groups for this List

Available Media Resource Groups

- Conf_MRGL
- Ext_MRGL
- LYNC-MRGL
- MRG_IOS_GW_CFB
- SIP_TRUNK_MRGL

Selected Media Resource Groups

- SME-MRG

Save | Delete | Copy | Add New

*- indicates required item.



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Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Media Resource Group List Configuration Related Links: [Back To Find/List](#)

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

Conf_MRG
Ext_MRG
MRG_IOS_GW_CFB
SIP_TRUNK_MRG
SME-MRG

Selected Media Resource Groups
LYNC-MRG

Save Delete Copy Add New

*- indicates required item.

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Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Media Resource Group List Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Add New

Status
 Status: Ready

Media Resource Group List Status
Media Resource Group List: SoftwareMGL (used by 7 devices)

Media Resource Group List Information
Name*

Media Resource Groups for this List
Available Media Resource Groups

Conf_MRG
Ext_MRG
LYNC-MRG
MRG_IOS_GW_CFB
SIP_TRUNK_MRG

Selected Media Resource Groups
SoftwareMTP


Save Delete Copy Add New

*- indicates required item.

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



Cisco Unified Communications Manager-SME Route Patterns





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Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)



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

Find and List Route Patterns

 Add New  Select All  Clear All  Delete Selected

Status
 18 records found

Route Patterns (1 - 18 of 18) Rows per Page

Find Route Patterns where begins with  

| <input type="checkbox"/> | Pattern ^ | Description | Partition | Route Filter | Associated Device |
|--------------------------|-------------------------------|---|-----------|--------------|---|
| <input type="checkbox"/> | 101X | | | | SME-SIP-trunk-to-CUCM-172-20-85-110 |
| <input type="checkbox"/> | 11XX | | | | S2/SU0/DS1-0@router |
| <input type="checkbox"/> | 1300X | | | | SME-SIP-trunk-to-CUCM-172-20-85-110 |
| <input type="checkbox"/> | 1408XXXXXX | | | | SIP-trunk-to-from-SP-viaCUBE |
| <input type="checkbox"/> | 14XXXXXXX | Outbound calls to ATT PSTN | | | CUCM_TO_ASR-172-20-110-154 |
| <input type="checkbox"/> | 1510123500X | | | | OCS_SIP_trk-172-20-155-50 |
| <input type="checkbox"/> | 1XXX | | | | CUCM-RE-SIP-Trunk-192-168-71-40 |
| <input type="checkbox"/> | 21XXX | | | | CUCM-RE-SIP-Trunk-192-168-71-40 |
| <input type="checkbox"/> | 4014 | | | | SME_SIP_TRUNK_G711-172-20-109-252 |
| <input type="checkbox"/> | 5000 | Route Pattern to ExUM 2010 Voicemail | | | ExUM2010-SIP-trunk-172-20-117-220 |
| <input type="checkbox"/> | 6000 | Route Pattern to ExUM 2010 Auto Attendant | | | ExUM2010-SIP-trunk-172-20-117-220 |
| <input type="checkbox"/> | 70XX | | | | SIEMENS HIPATH SIP TK-172-20-188-13 |
| <input type="checkbox"/> | 710X | | | | SIP_TRUNK_TO_QITT-172-20-109-54 |
| <input type="checkbox"/> | 72XX | For SME 9.0 test case 14.3 | | | SME-SIP-trunk-to-CUCM-172-20-85-110 |
| <input type="checkbox"/> | 9.0111 | | | | SIP-trunk-to-from-SP-viaCUBE |
| <input type="checkbox"/> | 9.@ | | | | SIP-trunk-to-from-SP-viaCUBE |
| <input type="checkbox"/> | \+1408333451X | | | | SME-SIP-trunk-to-CUCM-172-20-85-110 |
| <input type="checkbox"/> | \+1415XXXXXX | | | | SIP-TK-to-Msft-LYNC-172-20-117-152 |



Route Pattern to Service Provider

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation | [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Route Pattern Configuration Related Links: [Back To Find/Li](#)

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition

| | |
|--|---|
| Route Pattern* | <input type="text" value="1408XXXXXXX"/> |
| Route Partition | < None > |
| Description | <input type="text"/> |
| Numbering Plan | -- Not Selected -- |
| Route Filter | < None > |
| MLPP Precedence* | Default |
| Apply Call Blocking Percentage | <input type="text"/> |
| Resource Priority Namespace Network Domain | < None > |
| Route Class* | Default |
| Gateway/Route List* | SIP-trunk-to-from-SP-viaCUBE (Edit) |
| Route Option | <input checked="" type="radio"/> Route this pattern
<input type="radio"/> Block this pattern <input type="text" value="No Error"/> |
| Call Classification* | OffNet |
| <input type="checkbox"/> Allow Device Override | <input checked="" type="checkbox"/> Provide Outside Dial Tone |
| <input type="checkbox"/> Require Forced Authorization Code | <input type="checkbox"/> Allow Overlap Sending |
| <input type="checkbox"/> Urgent Priority | |
| Authorization Level* | <input type="text" value="0"/> |
| <input type="checkbox"/> Require Client Matter Code | |



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Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Route Pattern Configuration Related Links: [Back To Find/](#)

Save Delete Copy Add New

☐ Require Credit 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 the Microsoft LYNC 2010 server

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Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Route Pattern Configuration Related Links: [Back To Find/](#)

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition

| | |
|--|---|
| Route Pattern* | <input type="text" value="+1415XXXXXXX"/> |
| Route Partition | < None > ▾ |
| Description | <input type="text"/> |
| Numbering Plan | -- Not Selected -- ▾ |
| Route Filter | < None > ▾ |
| MLPP Precedence* | Default ▾ |
| <input type="checkbox"/> Apply Call Blocking Percentage | <input type="text"/> |
| Resource Priority Namespace Network Domain | < None > ▾ |
| Route Class* | Default ▾ |
| Gateway/Route List* | SIP-TK-to-Msft-LYNC-172-20-117-152 ▾ (Edit) |
| Route Option | <input checked="" type="radio"/> Route this pattern
<input type="radio"/> Block this pattern <input type="text" value="No Error"/> |
| Call Classification* | OffNet ▾ |
| <input type="checkbox"/> Allow Device Override | <input checked="" type="checkbox"/> Provide Outside Dial Tone |
| <input type="checkbox"/> Require Forced Authorization Code | <input type="checkbox"/> Allow Overlap Sending |
| <input type="checkbox"/> Urgent Priority | |
| Authorization Level* | <input type="text" value="0"/> |
| <input type="checkbox"/> Require Client Matter Code | |



Cisco Unified CM Administration
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Navigation | Cisco Unified CM Administration

CCMAdministrator | Search Documentation | About

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

Route Pattern Configuration

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

Save Delete Copy Add New

☐ Require Client Matter Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Default

Calling Name Presentation*

Default

Calling Party Number Type*

Cisco CallManager

Calling Party Numbering Plan*

Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*

Default

Connected Name Presentation*

Default

Called Party Transformations

Discard Digits

< None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Cisco CallManager

Called Party Numbering Plan*

Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol

-- Not Selected --

Carrier Identification Code

Network Service

-- Not Selected --

Service Parameter Name

< Not Exist >

Service Parameter Value

Save

Delete

Copy

Add New

*. indicates required item.



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

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Navigation [Cisco Unified CM Administration](#)

[System](#) [Call Routing](#) [Media Resources](#) [Advanced Features](#) [Device](#) [Application](#) [User Management](#) [Bulk Administration](#) [Help](#)

Route Pattern Configuration Related Links: [Back To Find/](#)

[Save](#) [Delete](#) [Copy](#) [Add New](#)

Status
 Status: Ready

Pattern Definition
Route Pattern*
Route Partition
Description
Numbering Plan
Route Filter
MLPP Precedence*
☐ Apply Call Blocking Percentage
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



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 ▾

Route Pattern Configuration Related Links: [Back To Find/Lists](#)

Save Delete Copy Add New

☐ Require Client Master Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Default ▾

Calling Name Presentation*

Default ▾

Calling Party Number Type*

Cisco CallManager ▾

Calling Party Numbering Plan*

Cisco CallManager ▾

Connected Party Transformations

Connected Line ID Presentation*

Default ▾

Connected Name Presentation*

Default ▾

Called Party Transformations

Discard Digits

< None > ▾

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Cisco CallManager ▾

Called Party Numbering Plan*

Cisco CallManager ▾

ISDN Network-Specific Facilities Information Element

Network Service Protocol

-- Not Selected -- ▾

Carrier Identification Code

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

*- indicates required item.



International Route Pattern

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Navigation [Cisco Unified CM Administra](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Route Pattern Configuration Related Links: [Back To Find/I](#)

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition

| | |
|--|--|
| Route Pattern* | 9.0111 |
| Route Partition | < None > |
| Description | |
| Numbering Plan | -- Not Selected -- |
| Route Filter | < None > |
| MLPP Precedence* | Default |
| <input type="checkbox"/> Apply Call Blocking Percentage | |
| Resource Priority Namespace Network Domain | < None > |
| Route Class* | Default |
| Gateway/Route List* | SIP-trunk-to-from-SP-viaCUBE (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"/> Require Forced Authorization Code | <input type="checkbox"/> Allow Overlap Sending |
| <input type="checkbox"/> Urgent Priority | |
| Authorization Level* | 0 |
| <input type="checkbox"/> Require Client Matter Code | |



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Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Route Pattern Configuration Related Links: [Back To Find/Lists](#)

Save Delete Copy Add New

☐ Require Client Matter Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Default ▾

Calling Name Presentation*

Default ▾

Calling Party Number Type*

Cisco CallManager ▾

Calling Party Numbering Plan*

Cisco CallManager ▾

Connected Party Transformations

Connected Line ID Presentation*

Default ▾

Connected Name Presentation*

Default ▾

Called Party Transformations

Discard Digits

PreDot ▾

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Cisco CallManager ▾

Called Party Numbering Plan*

Cisco CallManager ▾

ISDN Network-Specific Facilities Information Element

Network Service Protocol

-- Not Selected -- ▾

Carrier Identification Code

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

*- indicates required item.



SIP Route Pattern *

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Navigation: Cisco Unified CM Administration | Search Documentation | About

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

Find and List SIP Route Patterns

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

Status
1 records found

SIP Route Pattern (1 - 1 of 1) Rows per Page

Find SIP Route Pattern where IPv4 Pattern begins with Find Clear Filter

| | Pattern ^ | IPv6 Pattern | Description | Route Partition |
|--------------------------|-----------------|--------------|-------------|-----------------|
| <input type="checkbox"/> | lync2010rtm.com | | | |

Add New | Select All | Clear All | Delete Selected

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Navigation: Cisco Unified CM Administration | Search Documentation | About

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

SIP Route Pattern Configuration

Save | Delete | Copy | Add New

Status
Status: Ready

Pattern Definition

Pattern Usage: Domain Routing

IPv4 Pattern*: lync2010rtm.com

IPv6 Pattern

Description

Route Partition: < None >

SIP Trunk/Route List*: SIP-TK-to-Msft-LYNC-172-20-117-152 (Edit)

☐ Block Pattern

Calling Party Transformations

☐ Use Calling Party's External Phone Mask

Calling Party Transformation Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*: Default

Calling Line Name Presentation*: Default

Connected Party Transformations

Connected Line ID Presentation*: Default

Connected Line Name Presentation*: Default

Save | Delete | Copy | Add New

*- indicates required item.

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



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

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Navigation [Cisco Unified CM Administration](#)

CCMAdministrator | [Search Documentation](#) | [About](#)

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

Find and List Route Lists

Add New

Select All

Clear All

Delete Selected

Reset Selected

Apply Config to Selected

Status
 1 records found

Route List (1 - 1 of 1) Rows per Page

Find Route List where Name ▾ begins with ▾

| <input type="checkbox"/> | Name ^ | Description | Enabled |
|--------------------------|----------------|-------------|---------|
| <input type="checkbox"/> | PSTN-Access-RL | | true |

Cisco Unified CM Administration
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Navigation [Cisco Unified CM Administration](#)

CCMAdministrator | [Search Documentation](#) | [About](#)

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

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

Save

Delete

Copy

Reset

Apply Config

Add New

Status
 Status: Ready**Route List Information**
Registration Unknown
IP Address Unknown
☒ Device is trusted
Name*
Description
Cisco Unified Communications Manager Group*
☒ Enable this Route List (change effective on Save; no reset required)
☐ Run On All Active Unified CM Nodes**Route List Member Information**
Selected Groups**

▼ ▲

Removed Groups***

▼ ▲

 *. indicates required item.
 **Ordered by highest priority
 ***Will be removed from Route List when you click Save



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

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Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

Find and List Route Groups

Add New Select All Clear All Delete Selected

Status
 1 records found

Route Group (1 - 1 of 1) Rows per Page

Find Route Group where Route Group Name

| <input type="checkbox"/> | Name ^ |
|--------------------------|----------------|
| <input type="checkbox"/> | PSTN-Access-RG |

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Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

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

Save Delete Add New

Status
 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-MS-SIP-Trunk-172-20-109-250
IOS_GW_FAX-172-20-109-201

Port(s)

Current Route Group Members
Selected Devices (ordered by priority)*


IOS_GW_FAX-172-20-109-201 (All Ports)

Removed Devices***

Route Group Members
 [IOS_GW_FAX-172-20-109-201](#)



Cisco Unified Communications Manager-SME SIP Trunk Configuration






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


Navigation Cisco Unified CM Administration

CCMAdministrator | Search Documentation | About



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

Find and List Trunks






















 Add New  Select All  Clear All  Delete Selected  Reset Selected






Status
 20 records found

Trunks (1 - 20 of 20) Rows per Page 50

Find Trunks where Device Name ▾ begins with ▾ Find Clear Filter  

Select item or enter search text ▾

| <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-MS-SIP-Trunk-172-20-109-250 | to MS CUBE | | Default | | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | CUCM-RE-SIP-Trunk-192-168-71-40 | to RE CUCM | | Default | 21XXX | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | CUCM-RE-SIP-Trunk-192-168-71-40 | to RE CUCM | | Default | 1XXX | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | CUCM TO ASR-172-20-110-154 | SIP TRUNK TO ASR | | ATT ASR | 14XXXXXXXXXX | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | ExUM2010-SIP-trunk-172-20-117-220 | SIP Trunk to ExUM 2010 Server | | Default | 5000 | | | | SIP Trunk | Non Secure SIP Profile for ExUM |
| <input type="checkbox"/> |  | ExUM2010-SIP-trunk-172-20-117-220 | SIP Trunk to ExUM 2010 Server | | Default | 6000 | | | | SIP Trunk | Non Secure SIP Profile for ExUM |
| <input type="checkbox"/> |  | IOS_GW_FAX-172-20-109-201 | To IOS GW for Fax | | Default | | | PSTN-Access-RG | 1 | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | OCS SIP trk-172-20-155-50 | SIP trunk to OCS Mediation Server | | Default | 1510123500X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | SIEMENS_HIPATH SIP TK-172-20-188-13 | SIP trunk to Siemens HiPath 4000 Rel 5 | | Default | 70XX | | | | SIP Trunk | Non Secure SIP Profile for Siemens |
| <input type="checkbox"/> |  | SIP-TK-to-MsfL-LYNC-172-20-117-152 | SIP-TK-to-MsfL-LYNC-EOWithPRACKsup | | G711-DP | \+1415XXXXXXXX | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | SIP-TK-to-MsfL-LYNC-172-20-117-152 | SIP-TK-to-MsfL-LYNC-EOWithPRACKsup | | G711-DP | lync2010rtm.com | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | SIP-trunk-to-from-SP-viaCUBE | SIP-TK-to-from-SP-viaCUBE-172-20-110-152 | | G711-DP | 9.011 | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | SIP-trunk-to-from-SP-viaCUBE | SIP-TK-to-from-SP-viaCUBE-172-20-110-152 | | G711-DP | 1408XXXXXXXX | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | SIP-trunk-to-from-SP-viaCUBE | SIP-TK-to-from-SP-viaCUBE-172-20-110-152 | | G711-DP | 9.@ | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | SIP TRUNK TO OITT-172-20-109-54 | SIP trunk to OITT | | Default | 710X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | SME-SIP-trunk-to-CUCM-172-20-85-110 | SME SIP Trunk-to-CUCM-StrdSIP-with-no-PRACK | | G711-DP | 72XX | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | SME-SIP-trunk-to-CUCM-172-20-85-110 | SME SIP Trunk-to-CUCM-StrdSIP-with-no-PRACK | | G711-DP | 101X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | SME-SIP-trunk-to-CUCM-172-20-85-110 | SME SIP Trunk-to-CUCM-StrdSIP-with-no-PRACK | | G711-DP | \+1408333451X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | SME-SIP-trunk-to-CUCM-172-20-85-110 | SME SIP Trunk-to-CUCM-StrdSIP-with-no-PRACK | | G711-DP | 1300X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> |  | SME SIP TRUNK G711-172-20-109-252 | SIP TRUNK TO SME OVER G711-StrdSIPprofile | | Default | 4014 | | | | SIP Trunk | Non Secure SIP Profile |

 Add New  Select All  Clear All  Delete Selected  Reset Selected



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

| | | | | | |
|--|--|-------------------------|----------------------------|-----------|--|
| SIP-trunk-to-from-SP-viaCUBE | SIP-TK-to-from-SP-viaCUBE-172-20-110-152 | G711-DP | 1408XXXXXX | SIP Trunk | Non Secure SIP Trunk Profile |
|--|--|-------------------------|----------------------------|-----------|--|

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation [Cisco Unified CM Administration](#)

CCMAdministrator | [Search Documentation](#) | [About](#)

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

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

Save Delete Reset Add New

Status
 Status: Ready

Device Information

| | |
|---|---|
| Product: | SIP Trunk |
| Device Protocol: | SIP |
| Trunk Service Type | None(Default) |
| Device Name* | <input type="text" value="SIP-trunk-to-from-SP-viaCUBE"/> |
| Description | <input type="text" value="SIP-TK-to-from-SP-viaCUBE-172-20-110-152"/> |
| Device Pool* | <input type="text" value="G711-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=" Internal-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 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* | Default |
| Use Trusted Relay Point* | Default |
| <input checked="" type="checkbox"/> PSTN Access | |
| <input type="checkbox"/> Run On All Active Unified CM Nodes | |

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

Call Routing Information

☐ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type* PAI

SIP Privacy* Default

Inbound Calls

Significant Digits* All

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings Default Prefix Settings

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|----------------------|----------------|-----------------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <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 Line ID Presentation* Default

Calling Name Presentation* Default

Calling and Connected Party Info Format* Deliver DN only in connected party

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name



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



SIP Information

Destination

☐ Destination Address is an SRV

| | Destination Address | Destination Address IPv6 | Destination Port | | |
|-----|---------------------|--------------------------|------------------|---|---|
| 1 * | 172.20.110.152 | | 5060 |  |  |

MTP Preferred Originating Codec*
711ulaw

BLF Presence Group*
Standard Presence group

SIP Trunk Security Profile*
Non Secure SIP Trunk Profile

Rerouting Calling Search Space
< None >

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

SUBSCRIBE Calling Search Space
< None >



SIP Profile*
Standard SIP Profile

DTMF Signaling Method*
RFC 2833

Normalization Script

Normalization Script
< None >

☐ Enable Trace

| | Parameter Name | Parameter Value | | |
|---|----------------|-----------------|---|---|
| 1 | | |  |  |

Geolocation Configuration

Geolocation
< None >

Geolocation Filter
< None >

☐ Send Geolocation Information

Save

Delete

Reset

Add New

 *. indicates required item.

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



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

| | | | | | | |
|-----|-------------------------------------|---|---------|---------------|-----------|------------------------|
| SIP | SME-SIP-trunk-to-CUCM-172-20-85-110 | SME SIP Trunk-to-CUCM-StrdSIP-with-no-PRACK | G711-DP | \+1408333451X | SIP Trunk | Non Secure SIP Profile |
|-----|-------------------------------------|---|---------|---------------|-----------|------------------------|

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

CCMAdministrator | Search Documentation | About

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

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

Save Delete Reset Add New

Status
 Status: Ready

Device Information

| | |
|-----------------------------|--|
| Product: | SIP Trunk |
| Device Protocol: | SIP |
| Trunk Service Type | None(Default) |
| Device Name* | <input type="text" value="SME-SIP-trunk-to-CUCM-172-20-85-110"/> |
| Description | <input type="text" value="SME SIP Trunk-to-CUCM-StrdSIP-with-no-PRACK"/> |
| Device Pool* | <input type="text" value="G711-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="Internal-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"/> |

☐ 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* | Default |
| Use Trusted Relay Point* | Default |
| <input checked="" type="checkbox"/> PSTN Access | |
| <input type="checkbox"/> Run On All Active Unified CM Nodes | |

Intercompany Media Engine (IME)

| | |
|------------------------------|----------|
| E.164 Transformation Profile | < None > |
|------------------------------|----------|

Multilevel Precedence and Preemption (MLPP) Information

| | |
|-------------|----------|
| MLPP Domain | < None > |
|-------------|----------|

Call Routing Information

| | |
|---|---------|
| <input type="checkbox"/> Remote-Party-Id | |
| <input checked="" type="checkbox"/> Asserted-Identity | |
| Asserted-Type* | PAI |
| SIP Privacy* | Default |

Inbound Calls

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

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings

Default Prefix Settings

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|---------|--------------|----------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <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 Line ID Presentation* | Default |
| Calling Name Presentation* | Default |
| Calling and Connected Party Info Format* | Deliver DN only in connected party |

☒ Redirecting Diversion Header Delivery - Outbound

| | |
|--------------------------------------|----------|
| Redirecting Party Transformation CSS | < None > |
|--------------------------------------|----------|

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

| | |
|--------------|--|
| Caller ID DN | |
| Caller Name | |

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



SIP Information

Destination

☐ Destination Address is an SRV

1 *

Destination Address

172.20.85.110

Destination Address IPv6

Destination Port

5060

+

-

MTP Preferred Originating Codec*

711ulaw

BLF Presence Group*

Standard Presence group

SIP Trunk Security Profile*

Non Secure SIP Trunk Profile

Rerouting Calling Search Space

< None >

Out-Of-Dialog Refer Calling Search Space

< None >

SUBSCRIBE Calling Search Space

< None >

SIP Profile*

Standard SIP Profile

DTMF Signaling Method*

RFC 2833

Normalization Script

Normalization Script

< None >

☐ Enable Trace

1

Parameter Name

Parameter Value

+

-

Geolocation Configuration

Geolocation

< None >

Geolocation Filter

< None >

☐ Send Geolocation Information

Save

Delete

Reset

Add New

i

*. indicates required item.

i

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



Cisco Unified Communications Manager-SME SIP trunk to Microsoft Lync 2010 Server Standard Edition

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation [Cisco Unified CM Administra](#)

CCMAdministrator | [Search Documentation](#) | [About](#)

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

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

Save Delete Reset Add New

Status
 Status: Ready

Device Information

| | |
|-----------------------------|---|
| Product: | SIP Trunk |
| Device Protocol: | SIP |
| Trunk Service Type | None(Default) |
| Device Name* | <input type="text" value="SIP-TK-to-Msft-LYNC-172-20-117-152"/> |
| Description | <input type="text" value="SIP-TK-to-Msft-LYNC-EOwithPRACKsup"/> |
| Device Pool* | <input type="text" value="G711-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"/> |

☒ 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* | Default |
| Use Trusted Relay Point* | Default |
| <input checked="" type="checkbox"/> PSTN Access | |
| <input type="checkbox"/> Run On All Active Unified CM Nodes | |

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type* PAI

SIP Privacy* Default

Inbound Calls

Significant Digits* All

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings Default Prefix Settings

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|---------|--------------|-----------------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <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 Line ID Presentation* Default

Calling Name Presentation* Default

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

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

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



SIP Information

Destination

☐ Destination Address is an SRV

| | Destination Address | Destination Address IPv6 | Destination Port | | |
|-----|---------------------|--------------------------|------------------|--|--|
| 1 * | 172.20.117.152 | | 5068 | | |

MTP Preferred Originating Codec*

711ulaw

BLF Presence Group*

Standard Presence group

SIP Trunk Security Profile*

Non Secure SIP Trunk Profile

Rerouting Calling Search Space

< None >

Out-Of-Dialog Refer Calling Search Space

< None >

SUBSCRIBE Calling Search Space

< None >

SIP Profile*

EO SIP Profile with no PRACK

DTMF Signaling Method*

RFC 2833

Normalization Script

Normalization Script

Remove_TIAS_Friendly_Patch

☐ Enable Trace

| | Parameter Name | Parameter Value | | |
|---|----------------|-----------------|--|--|
| 1 | | | | |

Geolocation Configuration

Geolocation

< None >

Geolocation Filter

< None >

☐ Send Geolocation Information

Save

Delete

Reset

Add New

*- indicates required item.

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

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Page 75 of 152



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

| | | | | | | |
|--|--|------------------------------------|-------------------------|---------------------------------|-----------|--|
| | SIP-TK-to-Msft-LYNC-172-20-117-152 | SIP-TK-to-Msft-LYNC-EOwithPRACKsup | G711-DP | lync2010rtm.com | SIP Trunk | Non Secure SIP Profile |
|--|--|------------------------------------|-------------------------|---------------------------------|-----------|--|

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

CCMAdministrator | Search Documentation | About

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

Trunk Configuration

Related Links: Back To Find/List

Save Delete Reset Add New

Status

Status: Ready

Device Information

Product:

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*

SIP Trunk

SIP

None(Default)

SIP-TK-to-Msft-LYNC-172-20-117-152

SIP-TK-to-Msft-LYNC-EOwithPRACKsup

G711-DP

< None >

Use System Default

Lync-MRGL

Hub_None

< None >

None

No Changes

No Changes

None

0

When using both sRTP and TLS



| | |
|---|---------|
| Route Class Signaling Enabled* | Default |
| Use Trusted Relay Point* | Default |
| <input checked="" type="checkbox"/> PSTN Access | |
| <input type="checkbox"/> Run On All Active Unified CM Nodes | |

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type* PAI

SIP Privacy* Default

Inbound Calls

Significant Digits* All

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings

Default Prefix Settings

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|--------------------------------------|--------------------------------|-----------------------------|-------------------------------------|
| Incoming Number | <input type="text" value="Default"/> | <input type="text" value="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 Line ID Presentation* Default

Calling Name Presentation* Default

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

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

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



SIP Information

Destination

☐ Destination Address is an SRV

| | Destination Address | Destination Address IPv6 | Destination Port | | |
|-----|---------------------|--------------------------|------------------|--|--|
| 1 * | 172.20.117.152 | | 5068 | | |

MTP Preferred Originating Codec*

711ulaw

▼

BLF Presence Group*

Standard Presence group

▼

SIP Trunk Security Profile*

Non Secure SIP Trunk Profile

▼

Rerouting Calling Search Space

< None >

▼

Out-Of-Dialog Refer Calling Search Space

< None >

▼

SUBSCRIBE Calling Search Space

< None >

▼

SIP Profile*

EO SIP Profile with no PRACK

▼

DTMF Signaling Method*

RFC 2833

▼

Normalization Script

Normalization Script

Remove_TIAS_Friendly_Patch

▼

☐ Enable Trace

| | Parameter Name | Parameter Value | | |
|---|----------------|-----------------|--|--|
| 1 | | | | |

Geolocation Configuration

Geolocation

< None >

▼

Geolocation Filter

< None >

▼

☐ Send Geolocation Information

Save

Delete

Reset


Add New

*- indicates required item.

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



Cisco Unified Communications Manager-SME SIP Profile Information





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Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)



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

Find and List SIP Profiles

 Add New  Select All  Clear All  Delete Selected

Status
 11 records found

SIP Profile (1 - 11 of 11) Rows per Page

Find SIP Profile where begins with  

| <input type="checkbox"/> | Name ^ | Description |
|--------------------------|--|--|
| <input type="checkbox"/> | EO SIP Profile for OCS 2007 R2 | Default SIP Profile |
| <input type="checkbox"/> | EO SIP Profile with no PRACK | SIP Profile EO with no PRACK-To MedSrv-117-152 |
| <input type="checkbox"/> | EO with PRACK supported | EO with PRACK supported |
| <input type="checkbox"/> | Early Offer SIP Profile for Siemens | Default SIP Profile |
| <input type="checkbox"/> | Standard SIP Profile | Default SIP Profile |
| <input type="checkbox"/> | Standard SIP Profile Avaya | Default SIP Profile |
| <input type="checkbox"/> | Standard SIP Profile For Cisco VCS | Default SIP Profile For Cisco Video Communication Server |
| <input type="checkbox"/> | Standard SIP Profile For TelePresence Conferencing | Default SIP Profile For Cisco TelePresence Conferencing |
| <input type="checkbox"/> | Standard SIP Profile Msft Fax | Default SIP Profile |
| <input type="checkbox"/> | Standard SIP Profile for ATT | Default SIP Profile |
| <input type="checkbox"/> | Standard SIP Profile with PRACK supported | Default SIP Profile |



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

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

SIP Profile Configuration

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

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

Name*

EO SIP Profile with no PRACK

Description

SIP Profile EO with no PRACK-To MedSrv-117-152

Default MTP Telephony Event Payload Type*

101

Early Offer for G.Clear Calls*

Disabled

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*

TIAS and AS

User-Agent and Server header information*

Send Unified CM Version Information as User-Ager

Accept Audio Codec Preferences in Received Offer*

Default

Dial String Interpretation*

Phone number consists of characters 0-9, *, #, and

☐ Redirect by Application

☐ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Enable ANAT

☐ Require SDP Inactive Exchange for Mid-Call Media Change

☒ Use Fully Qualified Domain Name in SIP Requests

Required for URI dialing

☐ Assured Services SIP conformance

Parameters used in Phone

Timer Invite Expires (seconds)*

180

Timer Register Delta (seconds)*

5

Timer Register Expires (seconds)*

3600

Timer T1 (msec)*

500

Timer T2 (msec)*

4000



| | |
|---|--|
| Retry INVITE* | <input type="text" value="6"/> |
| Retry Non-INVITE* | <input type="text" value="10"/> |
| Start Media Port* | <input type="text" value="16384"/> |
| Stop Media Port* | <input type="text" value="32766"/> |
| Call Pickup URI* | <input type="text" value="x-cisco-serviceuri-pickup"/> |
| Call Pickup Group Other URI* | <input type="text" value="x-cisco-serviceuri-opickup"/> |
| Call Pickup Group URI* | <input type="text" value="x-cisco-serviceuri-gpickup"/> |
| Meet Me Service URI* | <input type="text" value="x-cisco-serviceuri-meetme"/> |
| User Info* | <input type="text" value="None"/> |
| DTMF DB Level* | <input type="text" value="Nominal"/> |
| Call Hold Ring Back* | <input type="text" value="Off"/> |
| Anonymous Call Block* | <input type="text" value="Off"/> |
| Caller ID Blocking* | <input type="text" value="Off"/> |
| Do Not Disturb Control* | <input type="text" value="User"/> |
| Telnet Level for 7940 and 7960* | <input type="text" value="Disabled"/> |
| Resource Priority Namespace | <input type="text" value=" < None >"/> |
| Timer Keep Alive Expires (seconds)* | <input type="text" value="120"/> |
| Timer Subscribe Expires (seconds)* | <input type="text" value="120"/> |
| Timer Subscribe Delta (seconds)* | <input type="text" value="5"/> |
| Maximum Redirections* | <input type="text" value="70"/> |
| Off Hook To First Digit Timer (milliseconds)* | <input type="text" value="15000"/> |
| Call Forward URI* | <input type="text" value="x-cisco-serviceuri-cfwdall"/> |
| Speed Dial (Abbreviated Dial) URI* | <input type="text" value="x-cisco-serviceuri-abbrdial"/> |
| <input checked="" type="checkbox"/> Conference Join Enabled | |
| <input type="checkbox"/> RFC 2543 Hold | |
| <input checked="" type="checkbox"/> Semi Attended Transfer | |
| <input type="checkbox"/> Enable VAD | |

- ☐ Stutter Message Waiting
- ☐ MLPP User Authorization

Normalization Script

Normalization Script

☐ Enable Trace

| | Parameter Name | Parameter Value | |
|---|----------------------|----------------------|---|
| 1 | <input type="text"/> | <input type="text"/> | <input type="button" value="+"/> <input type="button" value="-"/> |

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*

Never

RSVP Over SIP*

Local RSVP

Resource Priority Namespace List

< None >

☒ Fall back to local RSVP

SIP Rel1XX Options*

Disabled

Video Call Traffic Class*

Mixed

Calling Line Identification Presentation*

Default

☐ Deliver Conference Bridge Identifier

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

☒ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

SIP OPTIONS Ping

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

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

60

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

120

Ping Retry Timer (milliseconds)*

500

Ping Retry Count*

6

Save

Delete

Copy

Reset

Apply Config

Add New

i

*- indicates required item.



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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation [Cisco Unified CM Administrat](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

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

Copy | Reset | Apply Config | Add New

Status
 Status: Ready
 All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information
Name*
Description
Default MTP Telephony Event Payload Type*
Early Offer for G.Clear Calls*
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*
User-Agent and Server header information*
Accept Audio Codec Preferences in Received Offer*
Dial String Interpretation*
☐ Redirect by Application
☐ Disable Early Media on 180
☐ Outgoing T.38 INVITE include audio mline
☐ Enable ANAT
☐ Require SDP Inactive Exchange for Mid-Call Media Change
☐ Use Fully Qualified Domain Name in SIP Requests
☐ Assured Services SIP conformance

Parameters used in Phone
Timer Invite Expires (seconds)*
Timer Register Delta (seconds)*
Timer Register Expires (seconds)*
Timer T1 (msec)*
Timer T2 (msec)*



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

- ☒ Conference Join Enabled
- ☐ RFC 2543 Hold
- ☒ Semi Attended Transfer
- ☐ Enable VAD
- ☐ Stutter Message Waiting
- ☐ MLPP User Authorization

Normalization Script

Normalization Script

☐ Enable Trace

| | Parameter Name | Parameter Value | |
|---|----------------------|----------------------|---|
| 1 | <input type="text"/> | <input type="text"/> | <input type="button" value="+"/> <input type="button" value="-"/> |

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*

Never

RSVP Over SIP*

Local RSVP

Resource Priority Namespace List

< None >

☒ Fall back to local RSVP

SIP Rel1XX Options*

Disabled

Video Call Traffic Class*

Mixed

Calling Line Identification Presentation*

Default

☐ Deliver Conference Bridge Identifier

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

☐ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow IX Application Media

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

SIP OPTIONS Ping

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

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

60

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

120

Ping Retry Timer (milliseconds)*

500

Ping Retry Count*

6

Copy

Reset

Apply Config

Add New

*- indicates required item.



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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation [Cisco Unified CM Administration](#)

[CCMAdministrator](#) | [Search Documentation](#) | [About](#)

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

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

Copy | Reset | Apply Config | Add New

Status
 Status: Ready
 All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information
Name*
Description
Default MTP Telephony Event Payload Type*
Early Offer for G.Clear Calls*
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*
User-Agent and Server header information*
Accept Audio Codec Preferences in Received Offer*
Dial String Interpretation*
☐ Redirect by Application
☐ Disable Early Media on 180
☐ Outgoing T.38 INVITE include audio mline
☐ Enable ANAT
☐ Require SDP Inactive Exchange for Mid-Call Media Change
☐ Use Fully Qualified Domain Name in SIP Requests
☐ Assured Services SIP conformance

Parameters used in Phone
Timer Invite Expires (seconds)*
Timer Register Delta (seconds)*
Timer Register Expires (seconds)*
Timer T1 (msec)*
Timer T2 (msec)*



| | |
|---|--|
| Retry INVITE* | <input type="text" value="6"/> |
| Retry Non-INVITE* | <input type="text" value="10"/> |
| Start Media Port* | <input type="text" value="16384"/> |
| Stop Media Port* | <input type="text" value="32766"/> |
| Call Pickup URI* | <input type="text" value="x-cisco-serviceuri-pickup"/> |
| Call Pickup Group Other URI* | <input type="text" value="x-cisco-serviceuri-opickup"/> |
| Call Pickup Group URI* | <input type="text" value="x-cisco-serviceuri-gpickup"/> |
| Meet Me Service URI* | <input type="text" value="x-cisco-serviceuri-meetme"/> |
| User Info* | <input type="text" value="None"/> |
| DTMF DB Level* | <input type="text" value="Nominal"/> |
| Call Hold Ring Back* | <input type="text" value="Off"/> |
| Anonymous Call Block* | <input type="text" value="Off"/> |
| Caller ID Blocking* | <input type="text" value="Off"/> |
| Do Not Disturb Control* | <input type="text" value="User"/> |
| Telnet Level for 7940 and 7960* | <input type="text" value="Disabled"/> |
| Resource Priority Namespace | <input type="text" value=" < None >"/> |
| Timer Keep Alive Expires (seconds)* | <input type="text" value="120"/> |
| Timer Subscribe Expires (seconds)* | <input type="text" value="120"/> |
| Timer Subscribe Delta (seconds)* | <input type="text" value="5"/> |
| Maximum Redirections* | <input type="text" value="70"/> |
| Off Hook To First Digit Timer (milliseconds)* | <input type="text" value="15000"/> |
| Call Forward URI* | <input type="text" value="x-cisco-serviceuri-cfwdall"/> |
| Speed Dial (Abbreviated Dial) URI* | <input type="text" value="x-cisco-serviceuri-abbrdial"/> |
| <input checked="" type="checkbox"/> Conference Join Enabled | |
| <input type="checkbox"/> RFC 2543 Hold | |
| <input checked="" type="checkbox"/> Semi Attended Transfer | |
| <input type="checkbox"/> Enable VAD | |
| <input type="checkbox"/> Stutter Message Waiting | |
| <input type="checkbox"/> MLPP User Authorization | |

Normalization Script

Normalization Script

☐ Enable Trace

| | Parameter Name | Parameter Value | |
|---|----------------------|----------------------|---|
| 1 | <input type="text"/> | <input type="text"/> | <input type="button" value="+"/> <input type="button" value="-"/> |

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*

Never

RSVP Over SIP*

Local RSVP

Resource Priority Namespace List

< None >

☒ Fall back to local RSVP

SIP Rel1XX Options*

Disabled

Video Call Traffic Class*

Mixed

Calling Line Identification Presentation*

Default

☐ Deliver Conference Bridge Identifier

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

☐ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow IX Application Media

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

SIP OPTIONS Ping

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

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

60

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

120

Ping Retry Timer (milliseconds)*

500

Ping Retry Count*

6

Copy

Reset

Apply Config

Add New

*- indicates required item.



Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager-Software release

Cisco Unified CM Administration
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Navigation **Cisco Unified CM Administration**

administrator | Search Documentation | About

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

Cisco Unified CM Administration
System version: 9.0.1.10000-37

Last Successful Logon: Friday, October 26, 2012 3:09:18 PM PDT

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



Cisco Unified Communications Manager-Region Configuration

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For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About

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

Find and List Regions

+ Add New ☐ Select All ☐ Clear All ☒ Delete Selected

Status
3 records found

Regions (1 - 3 of 3) Rows per Page

Find Regions where Name begins with Find Clear Filter

| <input type="checkbox"/> | Name ^ |
|--------------------------|----------------------------|
| <input type="checkbox"/> | Default |
| <input type="checkbox"/> | g711region |
| <input type="checkbox"/> | g729region |

Add New Select All Clear All Delete Selected

Cisco UCM Regions Configuration – G711 Region

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For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About

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

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

Save ☒ Delete ☐ Reset ☐ Apply Config ☐ Add New

Region Information
Name*

Region Relationships

| Region | Audio Codec Preference List | Maximum Audio Bit Rate | Maximum Session Bit Rate for Video Calls |
|------------|---|------------------------|--|
| Default | Use System Default (Factory Default low loss) | 64 kbps (G.722, G.711) | 384 |
| g711region | Custom List G711 | 64 kbps (G.722, G.711) | 384 |
| g729region | Custom List G729 | 8 kbps (G.729) | 384 |

NOTE: Regions not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

| Regions | Audio Codec Preference List | Maximum Audio Bit Rate | Maximum Session Bit Rate for Video Calls |
|-------------------------------------|-----------------------------|------------------------|---|
| Default
g711region
g729region | Keep Current Setting ▾ | 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 |

Save Delete Reset Apply Config Add New

i *- indicates required item.



Cisco UCM Regions Configuration – G729 Region

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Navigation [Cisco Unified CM Administration](#)

administrator | [Search Documentation](#) | [About](#)

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

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

Save Delete Reset Apply Config Add New

Region Information

Name*

Region Relationships

| Region | Audio Codec Preference List | Maximum Audio Bit Rate | Maximum Session Bit Rate for Video Call |
|------------|---|------------------------|---|
| Default | Use System Default (Factory Default low loss) | 64 kbps (G.722, G.711) | 384 |
| g711region | Custom List G729 | 8 kbps (G.729) | 384 |
| g729region | Custom List G729 | 8 kbps (G.729) | 384 |

NOTE: Regions not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

| Regions | Audio Codec Preference List | Maximum Audio Bit Rate | Maximum Session Bit Rate for Video Call |
|--|-----------------------------------|-----------------------------------|---|
| <div>Default
g711region
g729region</div> | <div>Keep Current Setting ▾</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 Unified Communications Manager-Device Pool configuration

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Navigation: Cisco Unified CM Administration
administrator | Search Documentation | About

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: 5

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

| <input type="checkbox"/> | Name ^ | Cisco Unified CM Group | Region | Date/Time Group | |
|--------------------------|----------------|------------------------|------------|-----------------|--|
| <input type="checkbox"/> | Default | Default | Default | CMLocal | |
| <input type="checkbox"/> | g711devicepool | Default | g711region | CMLocal | |
| <input type="checkbox"/> | g729devicepool | Default | g729region | CMLocal | |

Add New Select All Clear All Delete Selected

Cisco UCM Device Pool for G711

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Navigation: Cisco Unified CM Administration
administrator | Search Documentation | About

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

Device Pool Configuration Related Links: Back To Find/List

Save Delete + Add New

Status
Status: Ready

Device Pool Information
Device Pool: g711devicepool (9 members)**

Device Pool Settings

Device Pool Name* g711devicepool

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* g711region ▾

Media Resource Group List SJC-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 > |
| AAR Group | < None > |
| Calling Party Transformation CSS | < None > |
| Called Party Transformation CSS | < None > |

Geolocation Configuration

| | |
|--------------------|----------|
| Geolocation | < None > |
| Geolocation Filter | < None > |

Call Routing Information**Incoming Calling Party Settings**

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

[Clear Prefix Settings](#)[Default Prefix Settings](#)

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

Incoming Called Party Settings

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

[Clear Prefix Settings](#)[Default Prefix Settings](#)

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

Phone Settings**Inbound Call Settings**

| | |
|----------------------------------|----------|
| Calling Party Transformation CSS | < None > |
|----------------------------------|----------|

Connected Party Settings

| | |
|------------------------------------|----------|
| Connected Party Transformation CSS | < None > |
|------------------------------------|----------|

Redirecting Party Settings

| | |
|--------------------------------------|----------|
| Redirecting Party Transformation CSS | < None > |
|--------------------------------------|----------|

[Save](#) [Delete](#) [Copy](#) [Reset](#) [Apply Config](#) [Add New](#)

*- indicates required item.



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



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



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



Cisco UCM Device Pool for G729

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Navigation Cisco Unified CM Administrat

administrator | Search Documentation | About

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

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

Save Delete Copy Reset Apply Config Add New

Status
 Status: Ready

Device Pool Information
Device Pool: g729devicepool (0 members**)

Device Pool Settings
Device Pool Name* g729devicepool
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* g729region ▾
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
Geolocation < None > ▾
Geolocation Filter < None > ▾

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

Clear Prefix Settings Default Prefix Settings

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



Incoming Called Party Settings

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

[Clear Prefix Settings](#)

[Default Prefix Settings](#)

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

Phone Settings

Inbound Call Settings

Calling Party Transformation CSS

Connected Party Settings

Connected Party Transformation CSS

Redirecting Party Settings

Redirecting Party Transformation CSS

[Save](#) [Delete](#) [Copy](#) [Reset](#) [Apply Config](#) [Add New](#)



*- indicates required item.



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



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



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



Cisco Unified Communications Manager-Media Termination Point Configuration

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Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About

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

Find and List Media Termination Points

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

Status
1 records found

Media Termination Point (1 - 1 of 1) Rows per Page

Find Media Termination Point where: Name ▾ begins with ▾ Find Clear Filter

| Name | Description | Device Pool | Status | IP Address | Co |
|-------|-----------------|-------------|-----------------------------|---------------|-------------|
| MTP_2 | MTP_CUCM-ExUM10 | Default | Registered with CUCM-ExUM10 | 172.20.85.110 | Not Allowed |

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

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Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About

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

Media Termination Point Configuration Related Links: Back To Find/List

Save Reset Apply Config

Status
Status: Ready

Media Termination Point Information

Registration: Registered with Cisco Unified Communications Manager CUCM-ExUM10
IP Address: 172.20.85.110
IPv6 Address: 0000:0000:0000:0000:0000:0000:0000:0000
Media Termination Point Type*: Cisco Media Termination Point Software
Host Server*: CUCM-ExUM10
Media Termination Point Name*: MTP_2
Description: MTP_CUCM-ExUM10
Device Pool*: Default
☐ Trusted Relay Point

Save Reset Apply Config

*- indicates required item.



Cisco Unified Communications Manager-Conference Bridge Configuration

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Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About

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

Find and List Conference Bridges

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

Status
1 records found

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

Find Conference Bridges where Name begins with Find Clear Filter

| Conference Bridge Name | Description | Device Pool | Status | IP Address |
|------------------------|-----------------|----------------|-----------------------------|---------------|
| CFB_2 | CFB_CUCM-ExUM10 | g711devicepool | Registered with CUCM-ExUM10 | 172.20.85.110 |

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

Software Conference Bridge

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Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About

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

Conference Bridge Configuration Related Links: Back To Find/List

Save Reset Apply Config

Status
Status: Ready

Conference Bridge Information
Conference Bridge : CFB_2 (CFB_CUCM-ExUM10)
Registration Registered with Cisco Unified Communications Manager CUCM-ExUM10
IP Address 172.20.85.110

Software Conference Bridge Info
Conference Bridge Type* Cisco Conference Bridge Software
Host Server CUCM-ExUM10
⚠ Device is not trusted
Conference Bridge Name* CFB_2
Description CFB_CUCM-ExUM10
Device Pool* g711devicepool
Common Device Configuration < None >
Location* Hub_None
Use Trusted Relay Point* Default

Save Reset Apply Config

*- indicates required item.



Cisco Unified Communications Manager-Media Resource Group configuration

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Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About

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

Find and List Media Resource Groups

+ Add New ☐ Select All ☐ Clear All ☒ Delete Selected

Status
6 records found

Media Resource Group (1 - 6 of 6) Rows per Page

Find Media Resource Group where Name ▾ begins with ▾ Find Clear Filter

| <input type="checkbox"/> | Name ^ | Description | Multi-cast | |
|--------------------------|---------------------------------------|-----------------------|------------|--|
| <input type="checkbox"/> | MRG for ExUM 2007 Fax | MRG for ExUM 2007 Fax | false | |
| <input type="checkbox"/> | MRG for ExUM 2010 | MRG for ExUM 2010 | false | |
| <input type="checkbox"/> | MRG_HW_MTP | MRG HW MTP | false | |
| <input type="checkbox"/> | MRG_SW_MTP | MRG SW MTP | false | |
| <input type="checkbox"/> | MRG_SW_noMTP | MRG SW no MTP | false | |
| <input type="checkbox"/> | SJC-MRG | SJC-MRG | false | |

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Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About

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

Media Resource Group Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Add New

Status
Status: Ready

Media Resource Group Status
Media Resource Group: SJC-MRG (used by 7 devices)

Media Resource Group Information

Name*
Description

Devices for this Group

Available Media Resources**

⌵ ⌶

Selected Media Resources*

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Legend:
*- 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-Media Resource Group List configuration

Cisco Unified CM Administration
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Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About

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

Find and List Media Resource Group Lists

+ Add New ☐ Select All ☐ Clear All ☒ Delete Selected

Status
6 records found

Media Resource Group List (1 - 6 of 6) Rows per Page

Find Media Resource Group List where Name begins with Find Clear Filter

| <input type="checkbox"/> | Name ^ | Copy |
|--------------------------|--|------|
| <input type="checkbox"/> | MRGL for ExUM 2007 Fax | |
| <input type="checkbox"/> | MRGL for ExUM 2010 | |
| <input type="checkbox"/> | MRGL_HW_MTP | |
| <input type="checkbox"/> | MRGL_SW_MTP | |
| <input type="checkbox"/> | MRGL_SW_noMTP | |
| <input type="checkbox"/> | SJC-MRGL | |

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For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About

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

Media Resource Group List Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Add New

Status
Status: Ready

Media Resource Group List Status
Media Resource Group List: SJC-MRGL (used by 7 devices)

Media Resource Group List Information
Name*

Media Resource Groups for this List

Available Media Resource Groups

MRG for ExUM 2007 Fax
MRG for ExUM 2010
MRG_HW_MTP
MRG_SW_MTP
MRG_SW_noMTP

▼ ^

Selected Media Resource Groups


SJC-MRGL

▼ ^

***** indicates required item.



Cisco Unified Communications Manager-Route Pattern Configuration





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


Navigation Cisco Unified CM Administration

administrator | Search Documentation | About



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



















Find and List Route Patterns

 Add New  Select All  Clear All  Delete Selected

Status
 27 records found

Route Patterns (1 - 27 of 27) Rows per Page 50

Find Route Patterns where begins with  

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



Route Pattern-101X

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Navigation [Cisco Unified CM Administration](#)

administrator | [Search Documentation](#) | [About](#)

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

Route Pattern Configuration Related Links: [Back To Find/](#)

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition

Route Pattern*101X

Route Partition< None >

DescriptionIncoming from CUCM endpoint to SME and to LYNC

Numbering Plan-- Not Selected --

Route Filter< None >

MLPP Precedence*Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain< None >

Route Class*Default

Gateway/Route List*SIP-Trunk-to-SME [\(Edit\)](#)

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

Connected Party Transformations

Connected Line ID Presentation*Default

Connected Name Presentation*Default

Called Party Transformations

Discard Digits< None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)+1415222

Called Party Number Type*Cisco CallManager

Called Party Numbering Plan*Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol-- Not Selected --

Carrier Identification Code

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

Save Delete Copy Add New

*. indicates required item.



Route Pattern-1408XXXXXXX

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Navigation [Cisco Unified CM Administration](#)

administrator | [Search Documentation](#) | [About](#)

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

Route Pattern Configuration Related Links: [Back To Find/](#)

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition
Route Pattern* 1408XXXXXXX
Route Partition < None >
Description Incoming from CUCM endpoint to SME and to SP
Numbering Plan -- Not Selected --
Route Filter < None >
MLPP Precedence* Default
☐ Apply Call Blocking Percentage
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* SIP-Trunk-to-SME [\(Edit\)](#)
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 4082733982
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling Party Number Type* Cisco CallManager
Calling Party Numbering Plan* Cisco CallManager

Connected Party Transformations
Connected Line ID Presentation* Default
Connected Name Presentation* Default

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

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

Save

Delete

Copy

Add New

*- indicates required item.



Route Pattern- 141522210XX

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Navigation Cisco Unified CM Administration

administrator | Search Documentation | About

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

Route Pattern Configuration

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

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition
Route Pattern*
Route Partition
Description
Numbering Plan
Route Filter
MLPP Precedence*
☐ Apply Call Blocking Percentage
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*

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

Save

*- indicates required item.



SIP Route Pattern*

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Navigation [Cisco Unified CM Administration](#)

administrator | [Search Documentation](#) | [About](#)

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

Find and List SIP Route Patterns

Add New Select All Clear All Delete Selected

Status
 1 records found

SIP Route Pattern (1 - 1 of 1) Rows per Page

Find SIP Route Pattern where IPv4 Pattern ▾ begins with ▾

| <input type="checkbox"/> | Pattern ^ | IPv6 Pattern | Description | Route Partition | |
|--------------------------|-----------------|--------------|-------------|-----------------|--|
| <input type="checkbox"/> | lync2010rtm.com | | | | |

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For Cisco Unified Communications Solutions

Navigation [Cisco Unified CM Administration](#)

administrator | [Search Documentation](#) | [About](#)

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

SIP Route Pattern Configuration Related Links: [Back To Find/L](#)

Save Delete Copy Add New

Status
 Status: Ready**Pattern Definition**

Pattern Usage Domain Routing

IPv4 Pattern*

IPv6 Pattern

Description

Route Partition

SIP Trunk/Route List* [\(Edit\)](#)

☐ Block Pattern

Calling Party Transformations

☐ Use Calling Party's External Phone Mask

Calling Party Transformation Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* ▾

Calling Line Name Presentation* ▾

Connected Party Transformations

Connected Line ID Presentation* ▾

Connected Line Name Presentation* ▾

*- indicates required item.

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



Cisco Unified Communications Manager-SIP Trunk configuration

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Navigation Cisco Unified CM Administration

administrator | Search Documentation | About

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

Find and List Trunks

Status

26 records found

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

Find Trunks where Device Name begins with Find Clear Filter

Select item or enter search text

| | | Name ^ | Description | Calling Search Space | Device Pool | Route Pattern | Partition | Route Group | Priority | Trunk Type | SIP Trunk Security Profile |
|--------------------------|--|--|---|----------------------|--------------------------------|---------------------------------|-----------|-----------------------------|----------|------------|---|
| <input type="checkbox"/> | | CUCM-SUMMER-SIP-trunk | CUCM-SUMMER-SIP-trunk | | Default | 63XX | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | ExUM07_SIP_trunk | ExUM07_SIP_trunk | | Default | 610XX | | | | SIP Trunk | Non Secure SIP Profile for ExUM |
| <input type="checkbox"/> | | ExUM10_Trunk | Trunk to ExUM 2010 Server | | Default | 11000 | | | | SIP Trunk | Non Secure SIP Profile for ExUM |
| <input type="checkbox"/> | | ExUM10_Trunk | Trunk to ExUM 2010 Server | | Default | 5000 | | | | SIP Trunk | Non Secure SIP Profile for ExUM |
| <input type="checkbox"/> | | ExUM10_Trunk | Trunk to ExUM 2010 Server | | Default | 5000 | | | | SIP Trunk | Non Secure SIP Profile for ExUM |
| <input type="checkbox"/> | | Lync-MedServer34-Cluster | Med12 Cluster | | Default | | | Failover RG | 1 | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | Lync2010-RTM_SIP_Trunk | SIP Trunk to Lync 2010 RTM colocated Med Server | | q711devicepool | 141522210XX | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | Lync2010_SIP_Trunk | SIP Trunk to Lync2010 colocated med server | | Default | | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | LyncIT_SIP_Trunk | SIP Trunk to LyncIT client | | q711devicepool | 222101X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | MS_GW1 | SIP Trunk to SIP GW for PSTN connection | | Default | 775X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | QCSR2-MSPBX-SIP-Trunk | QCSR2-MSPBX-SIP-Trunk | | Default | 7101X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | QCSR2-MEDSRV_SIP_Trunk | SIP Trunk to QCSR2 Mediation Server | | Default | 100X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | QCSR2_OITT_SIP_Trunk | SIP Trunk to QCSR2 OITT | | Default | | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SIP-Trunk-to-SME | SIP Trunk to CUCM-SME-172-20-109-254 | | q711devicepool | 101X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SIP-Trunk-to-SME | SIP Trunk to CUCM-SME-172-20-109-254 | | q711devicepool | 770X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SIP-Trunk-to-SME | SIP Trunk to CUCM-SME-172-20-109-254 | | q711devicepool | lvnc2010rtm.com | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SIP-Trunk-to-SME | SIP Trunk to CUCM-SME-172-20-109-254 | | q711devicepool | 555770X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SIP-Trunk-to-SME | SIP Trunk to CUCM-SME-172-20-109-254 | | q711devicepool | 1408XXXXXXX | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SIP-Trunk-to-SME | SIP Trunk to CUCM-SME-172-20-109-254 | | q711devicepool | 650X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SIP-Trunk-to-SME | SIP Trunk to CUCM-SME-172-20-109-254 | | q711devicepool | 660X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SIP-Trunk-to-SME | SIP Trunk to CUCM-SME-172-20-109-254 | | q711devicepool | 347X | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SME-SIP-Trunk | SME SIP Trunk | | Default | | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SP-ASR-SIP-trunk | SIP trunk to SP VZ ASR | | q711devicepool | 9@ | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SP-ASR-SIP-trunk | SIP trunk to SP VZ ASR | | q711devicepool | 6411 | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SP-ASR-SIP-trunk | SIP trunk to SP VZ ASR | | q711devicepool | 72XX | | | | SIP Trunk | Non Secure SIP Profile |
| <input type="checkbox"/> | | SP-ASR-SIP-trunk | SIP trunk to SP VZ ASR | | q711devicepool | 44XXXXXXXXXX | | | | SIP Trunk | Non Secure SIP Profile |

Add New

Select All

Clear All

Delete Selected

Reset Selected



SIP trunk to SME with Route pattern 101X

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Navigation [Cisco Unified CM Administrat](#)

administrator | [Search Documentation](#) | [About](#)

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

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

Save Delete Reset Add New

Status

Status: Ready

Device Information

| | |
|-----------------------------|--------------------------------------|
| Product: | SIP Trunk |
| Device Protocol: | SIP |
| Trunk Service Type | None(Default) |
| Device Name* | SIP-Trunk-to-SME |
| Description | SIP Trunk to CUCM-SME-172-20-109-254 |
| Device Pool* | g711devicepool |
| Common Device Configuration | < None > |
| Call Classification* | Use System Default |
| Media Resource Group List | SJC-MRGL |
| Location* | Hub_None |
| AAR Group | < None > |
| Tunneled Protocol* | None |
| QSIG Variant* | No Changes |
| ASN.1 ROSE OID Encoding* | No Changes |
| Packet Capture Mode* | None |
| Packet Capture Duration | 0 |

☐ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Path Replacement Support

☐ Transmit UTF-8 for Calling Party Name

☐ Transmit UTF-8 Names in QSIG APDU

☐ Unattended Port



☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*

When using both sRTP and TLS

Route Class Signaling Enabled*

Default

Use Trusted Relay Point*

Default

☒ PSTN Access

☐ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

Call Routing Information

☐ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type* PAI

SIP Privacy* Default

Inbound Calls

Significant Digits* All

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

[Clear Prefix Settings](#)

[Default Prefix Settings](#)

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|----------------------|----------------|-----------------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <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* Last Redirect Number (External)

Calling Line ID Presentation* Default

Calling Name Presentation* Default

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

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

☐ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

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



SIP Information

Destination

☐ Destination Address is an SRV

| | Destination Address | Destination Address IPv6 | Destination Port |
|-----|---------------------|--------------------------|------------------|
| 1 * | 172.20.109.254 | | 5060 |

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

Normalization Script

Normalization Script: remove-tias

☐ Enable Trace

| | Parameter Name | Parameter Value |
|---|----------------|-----------------|
| 1 | | |

Geolocation Configuration

| | |
|--------------------|----------|
| Geolocation | < None > |
| Geolocation Filter | < None > |

☐ Send Geolocation Information

*- indicates required item.

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



SIP trunk to SME with Route Pattern lync2010rtm.com

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Navigation [Cisco Unified CM Administra](#)

administrator | [Search Documentation](#) | [About](#)

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

Trunk Configuration

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

Save Delete Reset Add New

Status

Status: Ready

Device Information

| | |
|---|--------------------------------------|
| Product: | SIP Trunk |
| Device Protocol: | SIP |
| Trunk Service Type | None(Default) |
| Device Name* | SIP-Trunk-to-SME |
| Description | SIP Trunk to CUCM-SME-172-20-109-254 |
| Device Pool* | g711devicepool |
| Common Device Configuration | < None > |
| Call Classification* | Use System Default |
| Media Resource Group List | SJC-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 |
| <input checked="" type="checkbox"/> PSTN Access | |
| <input type="checkbox"/> Run On All Active Unified CM Nodes | |

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

Call Routing Information

☐ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type* PAI

SIP Privacy* Default

Inbound Calls

Significant Digits* All

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings

Default Prefix Settings

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|---------|--------------|----------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <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* Last Redirect Number (External)

Calling Line ID Presentation* Default

Calling Name Presentation* Default

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

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

☐ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

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



SIP Information

Destination

☐ Destination Address is an SRV

| | Destination Address | Destination Address IPv6 | Destination Port |
|-----|---------------------|--------------------------|------------------|
| 1 * | 172.20.109.254 | | 5060 |

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

Normalization Script

Normalization Script: remove-tias

☐ Enable Trace

| | Parameter Name | Parameter Value |
|---|----------------|-----------------|
| 1 | | |

Geolocation Configuration

| | |
|--------------------|----------|
| Geolocation | < None > |
| Geolocation Filter | < None > |

☐ Send Geolocation Information

*. indicates required item.

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



SIP trunk to SME with Route Pattern 1408XXXXXXX

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Navigation Cisco Unified CM Administration

administrator | Search Documentation | About

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

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

Save Delete Reset Add New

Status
 Status: Ready

Device Information

| | |
|---|---|
| Product: | SIP Trunk |
| Device Protocol: | SIP |
| Trunk Service Type | None(Default) |
| Device Name* | <input type="text" value="SIP-Trunk-to-SME"/> |
| Description | <input type="text" value="SIP Trunk to CUCM-SME-172-20-109-254"/> |
| Device Pool* | <input type="text" value="g711devicepool"/> |
| 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="SJC-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 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* | Default |
| Use Trusted Relay Point* | Default |
| <input checked="" type="checkbox"/> PSTN Access | |
| <input type="checkbox"/> Run On All Active Unified CM Nodes | |

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

Call Routing Information

☐ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type* PAI

SIP Privacy* Default

Inbound Calls

Significant Digits* All

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

[Clear Prefix Settings](#) [Default Prefix Settings](#)

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|---------|--------------|----------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <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* Last Redirect Number (External)

Calling Line ID Presentation* Default

Calling Name Presentation* Default

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

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

☐ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

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



SIP Information

Destination

☐ Destination Address is an SRV

| | Destination Address | Destination Address IPv6 | Destination Port |
|-----|---------------------|--------------------------|------------------|
| 1 * | 172.20.109.254 | | 5060 |

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

Normalization Script

Normalization Script: remove-tias

☐ Enable Trace

| | Parameter Name | Parameter Value |
|---|----------------|-----------------|
| 1 | | |

Geolocation Configuration

| | |
|--------------------|----------|
| Geolocation | < None > |
| Geolocation Filter | < None > |


☐ Send Geolocation Information

*- indicates required item.

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



Cisco Unified Communications Manager-SIP Profile Information





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Navigation [Cisco Unified CM Administration](#)

administrator | [Search Documentation](#) | [About](#)



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

Find and List SIP Profiles

 Add New  Select All  Clear All  Delete Selected

Status
 13 records found

SIP Profile (1 - 13 of 13) Rows per Page

Find SIP Profile where begins with  

| <input type="checkbox"/> | Name ^ | Description |
|--------------------------|--|--|
| <input type="checkbox"/> | EO SIP Profile for PRACK required | EO SIP Profile for PRACK required |
| <input type="checkbox"/> | EO SIP Profile for PRACK supported | EO SIP Profile for PRACK supported |
| <input type="checkbox"/> | EO for PRACK supported and Audio Codec Pref ON | EO for PRACK supported and Audio Codec Pref ON |
| <input type="checkbox"/> | EO with Audio Codec Pref ON | EO with Audio Codec Pref ON |
| <input type="checkbox"/> | Early Offer SIP Profile | Default SIP Profile |
| <input type="checkbox"/> | Lync Failover profile | Default SIP Profile |
| <input type="checkbox"/> | Standard SIP Profile | Default SIP Profile |
| <input type="checkbox"/> | Standard SIP Profile For Cisco VCS | Default SIP Profile For Cisco Video Communication Server |
| <input type="checkbox"/> | Standard SIP Profile For TelePresence Conferencing | Default SIP Profile For Cisco TelePresence Conferencing |
| <input type="checkbox"/> | Standard SIP Profile for ExUM FAX | Default SIP Profile for ExUM FAX |
| <input type="checkbox"/> | Standard SIP Profile for PRACK required | Default SIP Profile for PRACK required |
| <input type="checkbox"/> | Standard SIP Profile for PRACK supported | Default SIP Profile for PRACK supported |
| <input type="checkbox"/> | Standard SIP Profile with FQDN in SIP Requests | Default SIP Profile |



Standard SIP Profile

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For Cisco Unified Communications Solutions

Navigation [Cisco Unified CM Administration](#)

[administrator](#) | [Search Documentation](#) | [About](#)

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

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

Copy Reset Apply Config Add New

Status

Status: Ready

All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

| | |
|--|--|
| Name* | Standard SIP Profile |
| Description | Default SIP Profile |
| Default MTP Telephony Event Payload Type* | 101 |
| Early Offer for G.Clear Calls* | Disabled |
| SDP Session-level Bandwidth Modifier for Early Offer and Re-invites* | TIAS and AS |
| User-Agent and Server header information* | Send Unified CM Version Information as User-Ager |
| Accept Audio Codec Preferences in Received Offer* | Default |
| Dial String Interpretation* | Phone number consists of characters 0-9, *, #, and |
| <input type="checkbox"/> Redirect by Application | |
| <input type="checkbox"/> Disable Early Media on 180 | |
| <input type="checkbox"/> Outgoing T.38 INVITE include audio mline | |
| <input type="checkbox"/> Enable ANAT | |
| <input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change | |
| <input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests | |
| <input type="checkbox"/> Assured Services SIP conformance | |

Parameters used in Phone

| | |
|-----------------------------------|------|
| Timer Invite Expires (seconds)* | 180 |
| Timer Register Delta (seconds)* | 5 |
| Timer Register Expires (seconds)* | 3600 |
| Timer T1 (msec)* | 500 |
| Timer T2 (msec)* | 4000 |



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

- ☒ Conference Join Enabled
- ☐ RFC 2543 Hold
- ☒ Semi Attended Transfer
- ☐ Enable VAD
- ☐ Stutter Message Waiting
- ☐ MLPP User Authorization

Normalization Script

Normalization Script < None >

☐ Enable Trace

| | Parameter Name | Parameter Value | | |
|---|----------------|-----------------|---|---|
| 1 | | | + | - |

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*

Never

RSVP Over SIP*

Local RSVP

Resource Priority Namespace List

< None >

☒ Fall back to local RSVP

SIP Rel1XX Options*

Disabled

Video Call Traffic Class*

Mixed

Calling Line Identification Presentation*

Default

☐ Deliver Conference Bridge Identifier

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

☐ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

SIP OPTIONS Ping

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

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

60

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

120

Ping Retry Timer (milliseconds)*

500

Ping Retry Count*

6

Copy

Reset

Apply Config

Add New

*- indicates required item.



Standard SIP Profile for PRACK supported

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Navigation [Cisco Unified CM Administration](#)

administrator | [Search Documentation](#) | [About](#)

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

SIP Profile Configuration

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

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

Name*

Standard SIP Profile for PRACK supported

Description

Default SIP Profile for PRACK supported

Default MTP Telephony Event Payload Type*

101

Early Offer for G.Clear Calls*

Disabled

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*

TIAS and AS

User-Agent and Server header information*

Send Unified CM Version Information as User-Ager

Accept Audio Codec Preferences in Received Offer*

Default

Dial String Interpretation*

Phone number consists of characters 0-9, *, #, and

☐ Redirect by Application

☐ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Enable ANAT

☐ Require SDP Inactive Exchange for Mid-Call Media Change

☐ Use Fully Qualified Domain Name in SIP Requests

☐ Assured Services SIP conformance

Parameters used in Phone

Timer Invite Expires (seconds)*

180

Timer Register Delta (seconds)*

5

Timer Register Expires (seconds)*

3600

Timer T1 (msec)*

500

Timer T2 (msec)*

4000



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

- ☒ Conference Join Enabled
- ☐ RFC 2543 Hold
- ☒ Semi Attended Transfer
- ☐ Enable VAD
- ☐ Stutter Message Waiting
- ☐ MLPP User Authorization

Normalization Script

Normalization Script

☐ Enable Trace

| | Parameter Name | Parameter Value | |
|---|----------------------|----------------------|---|
| 1 | <input type="text"/> | <input type="text"/> | <input type="button" value="+"/> <input type="button" value="-"/> |

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*

Never

RSVP Over SIP*

Local RSVP

Resource Priority Namespace List

< None >

☒ Fall back to local RSVP

SIP Rel1XX Options*

Send PRACK if 1xx Contains SDP

Video Call Traffic Class*

Mixed

Calling Line Identification Presentation*

Default

☐ Deliver Conference Bridge Identifier

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

☐ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

SIP OPTIONS Ping

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

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

60

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

120

Ping Retry Timer (milliseconds)*

500

Ping Retry Count*

6

Save

Delete

Copy

Reset

Apply Config

Add New

*- indicates required item.



Standard SIP Profile with FQDN in SIP requests

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 ▾

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

Save Delete Copy Reset Apply Config Add New

Status
 Status: Ready
 All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information
Name*
Description
Default MTP Telephony Event Payload Type*
Early Offer for G.Clear Calls*
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*
User-Agent and Server header information*
Accept Audio Codec Preferences in Received Offer*
Dial String Interpretation*
☐ Redirect by Application
☐ Disable Early Media on 180
☐ Outgoing T.38 INVITE include audio mline
☐ Enable ANAT
☐ Require SDP Inactive Exchange for Mid-Call Media Change
☒ Use Fully Qualified Domain Name in SIP Requests Required for SIP URI Dialing
☐ Assured Services SIP conformance

Parameters used in Phone
Timer Invite Expires (seconds)*
Timer Register Delta (seconds)*
Timer Register Expires (seconds)*
Timer T1 (msec)*
Timer T2 (msec)*



| | |
|---|--|
| Retry INVITE* | <input type="text" value="6"/> |
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| Anonymous Call Block* | <input type="text" value="Off"/> |
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| Do Not Disturb Control* | <input type="text" value="User"/> |
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☐ Enable VAD
☐ Stutter Message Waiting
☐ MLPP User Authorization

Normalization Script
Normalization Script
☐ Enable Trace

| | Parameter Name | Parameter Value | |
|---|----------------------|----------------------|---|
| 1 | <input type="text"/> | <input type="text"/> | <input type="button" value="+"/> <input type="button" value="−"/> |

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Caller Name



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60

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

120

Ping Retry Timer (milliseconds)*

500

Ping Retry Count*

6

Save

Delete

Copy

Reset

Apply Config

Add New

*- indicates required item.



SCCP/SIP Phone configurations on the Cisco Unified Call Manager

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Navigation: Cisco Unified CM Administration
administrator | Search Documentation | About

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

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

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

Status
5 records found

Phone (1 - 5 of 5) Rows per Page

Find Phone where Device Name begins with Find Clear Filter

Select item or enter search text

| | Device Name(Line) ^ | Description | Device Pool | Device Protocol | Status | IP Address | Copy |
|--------------------------|---------------------------------|--|--------------------------------|-----------------|-----------------------------|-------------------------------|------|
| <input type="checkbox"/> | SEP001D705F9036 | 7971 SCCP Phone - James Cameroon X4511 | q711devicepool | SCCP | Registered with CUCM-ExUM10 | 172.20.109.57 | |
| <input type="checkbox"/> | SEP0023339CA1C3 | 7975 SIP Phone - John Smith x4511 | Default | SIP | Unknown | Unknown | |
| <input type="checkbox"/> | SEP00235E18F219 | 7961 SCCP Phone - David Lynch x4512 | q711devicepool | SCCP | Registered with CUCM-ExUM10 | 172.20.109.58 | |
| <input type="checkbox"/> | SEP1C17D337D1B8 | 9971 Sip Phone - Diana Wesley x4514 | q711devicepool | SIP | Registered with CUCM-ExUM10 | 172.20.109.59 | |
| <input type="checkbox"/> | SEP1C17D337D3F0 | Auto 1003 | Default | SIP | Registered with CUCM-ExUM10 | 172.20.109.38 | |

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



| | | | |
|---|--|---|---|
| 22 | None | Network Locale | < None > |
| 23 | None | Built In Bridge* | Default |
| 24 | None | Privacy* | Default |
| 25 | None | Device Mobility Mode* | Default View Current Device Mobility Settings |
| 26 | None | Owner User ID | < None > |
| 27 | None | Phone Personalization* | Default |
| 28 | None | Services Provisioning* | Default |
| 29 | None | Phone Load Name | |
| 30 | None | Single Button Barge | Default |
| 31 | None | Join Across Lines | Default |
| 32 | None | Use Trusted Relay Point* | Default |
| 33 | None | BLF Audible Alert Setting (Phone Idle)* | Default |
| 34 | None | BLF Audible Alert Setting (Phone Busy)* | Default |
| 35 | None | Always Use Prime Line* | Default |
| 36 | None | Always Use Prime Line for Voice Message* | Default |
| ----- Unassigned Associated Items ----- | | Geolocation | < None > |
| 37 | Add a new SD | <input checked="" type="checkbox"/> Retry Video Call as Audio | |
| 38 | Add a new SURF | <input type="checkbox"/> Ignore Presentation Indicators (internal calls only) | |
| 39 | Add a new BLF SD | <input checked="" type="checkbox"/> Allow Control of Device from CTI | |
| 40 | Add a new BLF Directed Call Park | <input checked="" type="checkbox"/> Logged Into Hunt Group | |
| 41 | CallBack | <input type="checkbox"/> Remote Device | |
| 42 | Call Park | <input type="checkbox"/> Protected Device**** | |
| 43 | Call Pickup | <input type="checkbox"/> Hot line Device***** | |
| 44 | Conference List | <input type="checkbox"/> Require off-premise location | |
| 45 | Do Not Disturb | | |
| 46 | End Call | | |

| | | | |
|----|---|--|---|
| 47 | Forward All | Call Routing Information | |
| 48 | Group Call Pickup | Inbound Calls | |
| 49 | Hold | Calling Party Transformation CSS | < None > |
| 50 | Hunt Group Logout | <input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS | |
| 51 | Intercom [1] - Add a new Intercom | Outbound Calls | |
| 52 | Malicious Call Identification | Calling Party Transformation CSS | < None > |
| 53 | Meet Me Conference | <input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS | |
| 54 | Mobility | | |
| 55 | New Call | Protocol Specific Information | |
| 56 | Other Pickup | Packet Capture Mode* | None |
| 57 | Quality Reporting Tool | Packet Capture Duration | 0 |
| 58 | Redial | BLF Presence Group* | Standard Presence group |
| 59 | Remove Last Participant | Device Security Profile* | Cisco 7971 - Standard SCCP Non-Secure Profile |
| 60 | Transfer | SUBSCRIBE Calling Search Space | < None > |
| 61 | Video Mode | <input type="checkbox"/> Unattended Port | |
| 62 | Queue Status | <input type="checkbox"/> Require DTMF Reception | |
| 63 | Privacy | <input type="checkbox"/> RFC2833 Disabled | |
| 64 | None | | |

| | |
|--|-------------------------------|
| Certification Authority Proxy Function (CAPF) Information | |
| Certificate Operation* | No Pending Operation |
| Authentication Mode* | By Null String |
| Authentication String | |
| Generate String | |
| Key Size (Bits)* | 1024 |
| Operation Completes By | 2012 11 15 12 (YYYY:MM:DD:HH) |
| Certificate Operation Status: None | |
| Note: Security Profile Contains Addition CAPF Settings. | |



Expansion Module Information

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

External Data Locations Information (Leave blank to use default)

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

Extension Information

| | |
|--|-----------------------------------|
| <input type="checkbox"/> Enable Extension Mobility | |
| Log Out Profile | -- Use Current Device Settings -- |
| Log in Time | < None > |
| Log out Time | < None > |

MLPP Information

| | |
|------------------|----------|
| MLPP Domain | < None > |
| MLPP Indication* | Default |
| MLPP Preemption* | Default |

Do Not Disturb

| | |
|---|----------------------------------|
| <input type="checkbox"/> 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 | Override Comm Settings |
|---|-----------------------------|--------------------------|
| <input type="checkbox"/> Disable Speakerphone | | |
| <input type="checkbox"/> Disable Speakerphone and Headset | | |
| Forwarding Delay * | Disabled | |
| PC Port * | Enabled | |
| Settings Access * | Enabled | <input type="checkbox"/> |
| Gratuitous ARP * | Disabled | |
| PC Voice VLAN Access * | Enabled | |
| Video Capabilities * | Disabled | <input type="checkbox"/> |
| Auto Line Select * | Disabled | |
| Web Access * | Disabled | <input type="checkbox"/> |
| Days Display Not Active | Sunday
Monday
Tuesday | <input type="checkbox"/> |
| Display On Time | 07:30 | <input type="checkbox"/> |



| | | |
|--|-----------------------------|--------------------------|
| Display On Duration | 10:30 | <input type="checkbox"/> |
| Display Idle Timeout | 01:00 | <input type="checkbox"/> |
| Enable Power Save Plus | Sunday
Monday
Tuesday | <input type="checkbox"/> |
| Phone On Time | 00:00 | <input type="checkbox"/> |
| Phone Off Time | 24:00 | <input type="checkbox"/> |
| Phone Off Idle Timeout* | 60 | <input type="checkbox"/> |
| <input type="checkbox"/> Enable Audible Alert | | <input type="checkbox"/> |
| EnergyWise Domain | | <input type="checkbox"/> |
| EnergyWise Endpoint Security Secret | | <input type="checkbox"/> |
| <input type="checkbox"/> Allow EnergyWise Overrides | | <input type="checkbox"/> |
| Span to PC Port* | Disabled | |
| Logging Display* | PC Controlled | |
| Load Server | | <input type="checkbox"/> |
| Recording Tone* | Disabled | |
| Recording Tone Local Volume* | 100 | |
| Recording Tone Remote Volume* | 50 | |
| Recording Tone Duration | | |
| Display On When Incoming Call* | Disabled | <input type="checkbox"/> |
| RTCP* | Disabled | <input type="checkbox"/> |
| "more" Soft Key Timer | 5 | |
| Auto Call Select* | Enabled | |
| Log Server | | <input type="checkbox"/> |
| 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 | |
| IPv6 Load Server | | <input type="checkbox"/> |
| IPv6 Log Server | | |
| 802.1x Authentication* | User Controlled | <input type="checkbox"/> |



| | | |
|---------------------------------------|------------------------|--------------------------|
| Detect Unified CM Connection Failure* | Normal | <input type="checkbox"/> |
| Minimum Ring Volume* | 0-Silent | |
| Headset Sidetone Level* | Use Phone Default | |
| HTTPS Server* | http and https Enabled | <input type="checkbox"/> |
| Enbloc Dialing* | Enabled | |
| Switch Port Remote Configuration* | Disabled | <input type="checkbox"/> |
| PC Port Remote Configuration* | Disabled | <input type="checkbox"/> |
| Automatic Port Synchronization* | Disabled | <input type="checkbox"/> |
| SSH Access* | Disabled | <input type="checkbox"/> |
| LOGIN Access* | Enabled | <input type="checkbox"/> |
| FIPS Mode* | Disabled | <input type="checkbox"/> |
| 80-bit SRTP* | Disabled | <input type="checkbox"/> |

i *- indicates required item.

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

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

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

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



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

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Navigation [Cisco Unified CM Administrat](#)

administrator | [Search Documentation](#) | [About](#)

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

Directory Number Configuration Related Links: [Configure Device \(SEP001D705F9036\)](#)

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

Edit Device

Edit Line Appearance

▼ ▲

Dissociate Devices

Directory Number Settings

Voice Mail Profile (Choose <None> to use system default)

Calling Search Space

BLF Presence Group*

User Hold MOH Audio Source

Network Hold MOH Audio Source

Auto Answer*

☐ Reject Anonymous Calls



| Directory URIs | | | |
|--|----------------------|-----------|--------------------------|
| Primary | URI | Partition | Remote |
| <input type="checkbox"/> | <input type="text"/> | < None > | <input type="checkbox"/> |
| <input type="button" value="Add Row"/> | | | |

| AAR Settings | | | |
|--|------------|----------------------|-----------|
| AAR | Voice Mail | AAR Destination Mask | AAR Group |
| <input type="checkbox"/> | or | <input type="text"/> | < None > |
| <input checked="" type="checkbox"/> 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 | | | |
| Forward All | <input type="checkbox"/> or | <input type="text"/> | Use System Default |
| Secondary Calling Search Space for Forward All | | | < None > |
| Forward Busy Internal | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward Busy External | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward No Answer Internal | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward No Answer External | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward No Coverage Internal | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward No Coverage External | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward on CTI Failure | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward Unregistered Internal | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward Unregistered External | <input type="checkbox"/> or | <input type="text"/> | < None > |
| No Answer Ring Duration (seconds) | | <input type="text"/> | |
| Call Pickup Group | | < None > | |

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

| MLPP Alternate Party Settings | |
|--|----------------------|
| Target (Destination) | <input type="text"/> |
| MLPP Calling Search Space | < None > |
| MLPP No Answer Ring Duration (seconds) | <input type="text"/> |

| Line Settings for All Devices | |
|---|----------------------|
| Hold Reversion Ring Duration (seconds) | <input type="text"/> |
| Setting the Hold Reversion Ring Duration to zero will disable the feature | |
| Hold Reversion Notification Interval (seconds) | <input type="text"/> |
| Setting the Hold Reversion Notification Interval to zero will disable the feature | |
| Party Entrance Tone* | Default |



Line 1 on Device SEP001D705F9036

| | | |
|--|--|--|
| Display (Caller ID) | <input type="text" value="James Cameroon"/> | Display text for a line appearance is intended for displaying text such as a name instead of a directory number for
If you specify a number, the person receiving a call may not see the proper identity of the caller. |
| ASCII Display (Caller ID) | <input type="text" value="James Cameroon"/> | |
| Line Text Label | <input type="text" value="James Cameroon"/> | |
| ASCII Line Text Label | <input type="text" value="James Cameroon"/> | |
| External Phone Number Mask | <input type="text"/> | |
| 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="Ring"/> | |
| 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 SEP001D705F9036

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 SEP001D705F9036

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

Users Associated with Line

*- indicates required item.

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



Device Name: [SEP1C17D337D1B8](#)

Device Protocol: SIP

Cisco

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administrator | Search Documentation | About

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

Phone Configuration

Related Links: Back To Find/List

Save X Delete Copy Reset Apply Config Add New

Status

Status: Ready

Association Information

Modify Button Items

1 7971 Line [1] - \+14083334514 (no partition)

2 7971 Line [2] - 34514 (no partition)

3 Add a new SD

4 Add a new SD

5 Add a new SD

6 Add a new SD

----- Unassigned Associated Items -----

7 7971 Line [3] - Add a new DN

8 Add a new SD

9 All Calls

10 Add a new BLF Directed Call Park

11 Call Park

12 Call Pickup

13 CallBack

14 Group Call Pickup

15 Hunt Group Logout

16 7971 Intercom [1] - Add a new Intercom

17 Malicious Call Identification

18 Meet Me Conference

19 Mobility

20 Other Pickup

21 Quality Reporting Tool

22 Redial

23 Add a new SURF

24 Add a new BLF SD

25 Answer Oldest

26 Do Not Disturb

27 Services

28 Record

29 Alerting Calls

30 Queue Status

31 Privacy

32 None

Phone Type

Product Type: Cisco 9971

Device Protocol: SIP

Device Information

Registered with Cisco Unified Communications Manager CUCM-ExUM10

IP Address 172.20.109.59

Active Load ID sip9971.9-3-1-33

Inactive Load ID sip9971.9-3-0RT1-85

Download Status Successful

Device is Active

Device is trusted

MAC Address* 1C17D337D1B8

Description 9971 Sip Phone - Diana Wesley x4514

Device Pool* g711devicepool View Details

Common Device Configuration < None > View Details

Phone Button Template* Standard 9971 SIP

Common Phone Profile* Standard Common Phone Profile

Calling Search Space < None >

AAR Calling Search Space < None >

Media Resource Group List SJC-MRGL

User Hold MOH Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

Location* Hub_None

AAR Group < None >

User Locale < None >

Network Locale < None >

Built In Bridge* Default

Privacy* Default

Device Mobility Mode* On View Current Device Mobility Settings

Owner User ID User05

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

Geolocation < None >

Feature Control Policy < None >

☐ Ignore Presentation Indicators (internal calls only)

☒ Allow Control of Device from CTI

☒ Logged Into Hunt Group

☐ Remote Device

☐ Protected Device****

☐ Require off-premise location



Call Routing Information

Inbound Calls

Calling Party Transformation CSS < None >

☒ Use Device Pool Calling Party Transformation CSS

Outbound Calls

Calling Party Transformation CSS < None >

☒ Use Device Pool Calling Party Transformation CSS

Protocol Specific Information

| | |
|---|---|
| Packet Capture Mode* | None |
| Packet Capture Duration | <input type="text" value="0"/> |
| BLF Presence Group* | Standard Presence group |
| SIP Dial Rules | < None > |
| MTP Preferred Originating Codec* | 711ulaw |
| Device Security Profile* | Cisco 9971 - Standard SIP Non-Secure Profile |
| Rerouting Calling Search Space | < None > |
| SUBSCRIBE Calling Search Space | < None > |
| SIP Profile* | Standard SIP Profile |
| Digest User | < None > |
| <input type="checkbox"/> Media Termination Point Required | |
| <input type="checkbox"/> Unattended Port | |
| <input type="checkbox"/> Require DTMF Reception | |

Certification Authority Proxy Function (CAPF) Information

| | |
|---|--|
| Certificate Operation* | No Pending Operation |
| Authentication Mode* | By Null String |
| Authentication String | <input type="text"/> |
| <input type="button" value="Generate String"/> | |
| Key Size (Bits)* | 1024 |
| Operation Completes By | 2012 11 16 12 (YYYY:MM:DD:HH) |
| Certificate Operation Status: None | |
| Note: Security Profile Contains Addition CAPF Settings. | |

Expansion Module Information

| | |
|--------------------|-----------------------------|
| Module 1 | < None > |
| Module 1 Load Name | <input type="text"/> |
| Module 2 | < None > |
| Module 2 Load Name | <input type="text"/> |
| Module 3 | < None > |
| Module 3 Load Name | <input type="text"/> |



| External Data Locations Information (Leave blank to use default) | |
|--|----------------------|
| Information | <input type="text"/> |
| Directory | <input type="text"/> |
| Messages | <input type="text"/> |
| Services | <input type="text"/> |
| Authentication Server | <input type="text"/> |
| Proxy Server | <input type="text"/> |
| Idle | <input type="text"/> |
| Idle Timer (seconds) | <input type="text"/> |
| Secure Authentication URL | <input type="text"/> |
| Secure Directory URL | <input type="text"/> |
| Secure Idle URL | <input type="text"/> |
| Secure Information URL | <input type="text"/> |
| Secure Messages URL | <input type="text"/> |
| Secure Services URL | <input type="text"/> |

| Extension Information | |
|--|-----------------------------------|
| <input type="checkbox"/> Enable Extension Mobility | |
| Log Out Profile | -- Use Current Device Settings -- |
| Log in Time | < None > |
| Log out Time | < None > |

| MLPP Information | |
|------------------|----------|
| MLPP Domain | < None > |
| MLPP Indication* | Off |
| MLPP Preemption* | Disabled |

| Do Not Disturb | | |
|---|----------------------------------|--|
| <input type="checkbox"/> Do Not Disturb | | |
| DND Option* | Use Common Phone Profile Setting | |
| DND Incoming Call Alert | < None > | |

| Secure Shell Information | | |
|--------------------------|-------|--|
| Secure Shell User | cisco | |
| Secure Shell Password | | |

| Product Specific Configuration Layout | | | |
|---|---|---|-------------------------------------|
| | ? | Param | Override Comr Settings |
| <input type="checkbox"/> Disable Speakerphone | | | |
| <input type="checkbox"/> Disable Speakerphone and Headset | | | |
| PC Port * | | Enabled | |
| Back USB Port* | | Enabled | <input type="checkbox"/> |
| Side USB Port* | | Enabled | <input type="checkbox"/> |
| Cisco Camera* | | Enabled | <input checked="" type="checkbox"/> |
| Video Capabilities* | | Enabled | <input checked="" type="checkbox"/> |
| Enable/Disable USB Classes | | Mass Storage
Human Interface Device
Audio Class | <input type="checkbox"/> |
| SDIO * | | Disabled | <input type="checkbox"/> |
| Bluetooth * | | Enabled | <input type="checkbox"/> |
| Wifi * | | Enabled | <input type="checkbox"/> |
| Bluetooth Profiles* | | Handsfree
Human Interface Device | <input type="checkbox"/> |
| Settings Access* | | Enabled | <input type="checkbox"/> |
| Gratuitous ARP* | | Disabled | |



| | | |
|---|-----------------------------|-------------------------------------|
| PC Voice VLAN Access* | Enabled | |
| Web Access* | Disabled | <input type="checkbox"/> |
| Show All Calls on Primary Line* | Disabled | |
| Days Display Not Active | Sunday
Monday
Tuesday | <input type="checkbox"/> |
| Display On Time | 07:30 | <input type="checkbox"/> |
| Display On Duration | 10:30 | <input type="checkbox"/> |
| Display Idle Timeout | 01:00 | <input type="checkbox"/> |
| HTTPS Server* | http and https Enabled | <input checked="" type="checkbox"/> |
| Enable Power Save Plus | Sunday
Monday
Tuesday | <input type="checkbox"/> |
| Phone On Time | 00:00 | <input type="checkbox"/> |
| Phone Off Time | 24:00 | <input type="checkbox"/> |
| Phone Off Idle Timeout* | 60 | <input type="checkbox"/> |
| <input type="checkbox"/> Enable Audible Alert | | <input type="checkbox"/> |
| EnergyWise Domain | | <input type="checkbox"/> |
| EnergyWise Endpoint Security Secret | | <input type="checkbox"/> |
| <input type="checkbox"/> Allow EnergyWise Overrides | | <input type="checkbox"/> |
| Span to PC Port* | Disabled | |
| Logging Display* | Disabled | |
| Load Server | | <input type="checkbox"/> |
| Recording Tone* | Disabled | |
| Recording Tone Local Volume* | 100 | |
| Recording Tone Remote Volume* | 50 | |
| Recording Tone Duration | | |

| | | |
|--|--------------------|-------------------------------------|
| Display On When Incoming Call* | Enabled | <input checked="" type="checkbox"/> |
| RTCP* | Enabled | <input checked="" type="checkbox"/> |
| Log Server | | |
| Advertise G.722 and iSAC Codecs * | Use System Default | |
| Wideband Headset UI Control* | Enabled | |
| Wideband Headset* | Enabled | |
| 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 | |
| 802.1x Authentication* | User Controlled | <input type="checkbox"/> |
| Detect Unified CM Connection Failure* | Normal | <input checked="" type="checkbox"/> |
| Switch Port Remote Configuration* | Disabled | <input type="checkbox"/> |
| PC Port Remote Configuration* | Disabled | <input type="checkbox"/> |
| Automatic Port Synchronization* | Disabled | <input type="checkbox"/> |
| Power Negotiation* | Enabled | <input type="checkbox"/> |
| Restrict Data Rates* | Disabled | |
| SSH Access* | Disabled | <input type="checkbox"/> |
| Incoming Call Toast Timer* | 5 | <input type="checkbox"/> |
| Provide Dial Tone from Release Button* | Disabled | <input type="checkbox"/> |
| Hide Video By Default* | Disabled | <input type="checkbox"/> |
| Background Image | | <input type="checkbox"/> |



| | | |
|-----------------------------------|-------------|--------------------------|
| Simplified New Call UI* | Disabled | <input type="checkbox"/> |
| Enable VXC VPN for MAC | | |
| VXC VPN Option* | Dual Tunnel | <input type="checkbox"/> |
| VXC Challenge* | Challenge | <input type="checkbox"/> |
| VXC-M Servers | | <input type="checkbox"/> |
| Revert to All Calls* | Disabled | <input type="checkbox"/> |
| 80-bit SRTP* | Disabled | <input type="checkbox"/> |
| RTCP for Video* | Enabled | <input type="checkbox"/> |
| Record Call Log from Shared Line* | Disabled | <input type="checkbox"/> |

i *- indicates required item.

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

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

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

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



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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation [Cisco Unified CM Administration](#)

[administrator](#) | [Search Documentation](#) | [About](#)

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

Directory Number Configuration Related Links: [Configure Device \(SEP1C17D337D1B8\)](#)

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

BLF Presence Group*

User Hold MOH Audio Source

Network Hold MOH Audio Source

Auto Answer*

☐ Reject Anonymous Calls



| Directory URIs | | | |
|--|----------------------|-----------|--------------------------|
| Primary | URI | Partition | Rem |
| <input type="checkbox"/> | <input type="text"/> | < None > | <input type="checkbox"/> |
| <input type="button" value="Add Row"/> | | | |

| AAR Settings | | | |
|---|-----------------------------|----------------------|-----------|
| AAR | Voice Mail | AAR Destination Mask | AAR Group |
| <input type="checkbox"/> | <input type="checkbox"/> or | <input type="text"/> | < None > |
| <input type="checkbox"/> 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 | <input type="text"/> | < None > |
| Secondary Calling Search Space for Forward All | | | < None > |
| Forward Busy Internal | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward Busy External | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward No Answer Internal | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward No Answer External | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward No Coverage Internal | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward No Coverage External | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward on CTI Failure | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward Unregistered Internal | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Forward Unregistered External | <input type="checkbox"/> or | <input type="text"/> | < None > |
| No Answer Ring Duration (seconds) | | <input type="text"/> | |
| Call Pickup Group | | < None > | |

| Park Monitoring | | | |
|--|-----------------------------|----------------------|---|
| | Voice Mail | Destination | Calling Search Space |
| Park Monitoring Forward No Retrieve Destination External | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Park Monitoring Forward No Retrieve Destination Internal | <input type="checkbox"/> or | <input type="text"/> | < None > |
| Park Monitoring Reversion Timer | | <input type="text"/> | A blank value will use value set in Park Monitoring Reversion Timer service parameter |

| MLPP Alternate Party Settings | |
|--|----------------------|
| Target (Destination) | <input type="text"/> |
| MLPP Calling Search Space | < None > |
| MLPP No Answer Ring Duration (seconds) | <input type="text"/> |

| Line Settings for All Devices | |
|--|--|
| Hold Reversion Ring Duration (seconds) | <input type="text"/> Setting the Hold Reversion Ring Duration to zero will disable the feature |
| Hold Reversion Notification Interval (seconds) | <input type="text"/> Setting the Hold Reversion Notification Interval to zero will disable the feature |
| Party Entrance Tone* | Default |

| Line 1 on Device SEP1C17D337D1B8 | |
|---|----------------------|
| Display (Caller ID) | Diana Wesley |
| If you specify a number, the person receiving a call may not see the proper identity of the caller. | |
| ASCII Display (Caller ID) | Diana Wesley |
| Line Text Label | Diana Wesley |
| ASCII Line Text Label | Diana Wesley |
| External Phone Number Mask | <input type="text"/> |
| 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 > | |
| <input checked="" type="checkbox"/> Log Missed Calls | | |

Multiple Call/Call Waiting Settings on Device SEP1C17D337D1B8

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

| | | |
|--------------------------|---|------------------------------------|
| Maximum Number of Calls* | 4 | |
| Busy Trigger* | 2 | (Less than or equal to Max. Calls) |

Multiple Call/Call Waiting Settings on Device SEP1C17D337D1B8

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

| | | |
|--------------------------|---|------------------------------------|
| Maximum Number of Calls* | 4 | |
| Busy Trigger* | 2 | (Less than or equal to Max. Calls) |

Forwarded Call Information Display on Device SEP1C17D337D1B8

| |
|---|
| <input checked="" type="checkbox"/> Caller Name |
| <input type="checkbox"/> Caller Number |
| <input type="checkbox"/> Redirected Number |
| <input checked="" type="checkbox"/> 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.



Cisco IOS Gateway Configurations

Cisco Unified Border Element Configuration on the ASR

```
CUBE-ASR1K_Vz_152>en
Password:
CUBE-ASR1K_Vz_152#
CUBE-ASR1K_Vz_152#
CUBE-ASR1K_Vz_152#
CUBE-ASR1K_Vz_152#
CUBE-ASR1K_Vz_152#sho ver
CUBE-ASR1K_Vz_152#sho version
Cisco IOS Software, IOS-XE Software (PPC_LINUX_IOSD-ADVENTERPRISE-M), Version 15.1(3)S2, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2011 by Cisco Systems, Inc.
Compiled Mon 12-Dec-11 15:09 by mcpre
```

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ROM: IOS-XE ROMMON

```
CUBE-ASR1K_Vz_152 uptime is 2 days, 16 hours, 8 minutes
Uptime for this control processor is 2 days, 16 hours, 10 minutes
System returned to ROM by reload
System image file is "bootflash:asr1000rp1-adventerprise.03.04.02.S.151-3.S2.bin"
Last reload reason: Reload Command
```

```
cisco ASR1002 (2RU) processor with 1708551K/6147K bytes of memory.
4 Gigabit Ethernet interfaces
32768K bytes of non-volatile configuration memory.
4194304K bytes of physical memory.
7798783K bytes of eUSB flash at bootflash:.
```

Configuration register is 0x2102

```
CUBE-ASR1K_Vz_152#sho run
CUBE-ASR1K_Vz_152#sho running-config
Building configuration...
```

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



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



```

sip-profiles 2
no call service stop
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class sip-profiles 1
response ANY sip-header Allow-Header modify "invite, options, bye, cancel, ack, prack, update, refer, subscribe, notify, info, register" "invite,
bye, cancel, ack, prack, subscribe, notify, info, register"
response 183 sip-header Allow-Header modify "ACK, BYE, 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"
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 /81/ /1\1/
!
voice translation-rule 2
rule 1 /^84089550468/ /14089550468/
!
!
voice translation-profile Outgoing-Fax-G711
translate called 1
!
voice translation-profile RICUBcallingnum
translate calling 2
translate called 2
!
!
!
!
redundancy
mode none
!
!
!
no ip ftp passive
ip ftp username ios_user
ip ftp password cisco
!
!
!
```




```
!  
interface GigabitEthernet0/0/0  
ip address 172.20.110.152 255.255.255.0  
negotiation auto  
!  
interface GigabitEthernet0/0/1  
no ip address  
standby delay minimum 30 reload 60  
standby version 2  
standby 6 priority 50  
standby 6 track 1 decrement 10  
shutdown  
negotiation auto  
bfd interval 500 min_rx 500 multiplier 3  
!  
interface GigabitEthernet0/0/2  
no ip address  
shutdown  
negotiation auto  
!  
interface GigabitEthernet0/0/3  
no ip address  
shutdown  
negotiation auto  
!  
interface GigabitEthernet0  
vrf forwarding Mgmt-intf  
no ip address  
shutdown  
negotiation auto  
!  
ip forward-protocol nd  
!  
no ip http server  
ip route 172.20.0.0 255.255.0.0 172.20.110.1  
ip route 172.30.218.0 255.255.255.0 172.20.110.150  
ip route 172.31.124.0 255.255.255.0 172.20.110.150  
!  
!  
!  
!  
control-plane  
!  
!  
!  
dial-peer voice 9000 voip1  
description to Service Provider - SP facing  
destination-pattern 1T  
session protocol sipv2  
session target sip-server  
session transport udp  
voice-class sip asserted-id pai  
dtmf-relay rtp-nte  
codec g711ulaw  
fax protocol pass-through g711ulaw
```

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



```
!  
dial-peer voice 9001 voip  
description Incoming to SME - SME facing  
destination-pattern 408933....  
session protocol sipv2  
session target ipv4:172.20.236.252  
session transport tcp  
dtmf-relay rtp-nte  
fax protocol pass-through g711ulaw  
!  
dial-peer voice 577 voip  
description Outbound to Fax Machines  
destination-pattern 408577....  
session protocol sipv2  
session target sip-server  
session transport udp  
incoming called-number 408577....  
codec g711ulaw  
fax protocol pass-through g711ulaw  
!  
dial-peer voice 6170 voip  
description To Fax Machines on Siemens  
destination-pattern 408933617.  
session protocol sipv2  
session target ipv4:172.20.236.252  
session transport udp  
incoming called-number 408933617.  
codec g711ulaw  
fax protocol pass-through g711ulaw  
!  
dial-peer voice 3470 voip  
description To Fax Machines on CUCM  
destination-pattern 40893334..  
session protocol sipv2  
session target ipv4:172.20.236.252  
session transport udp  
incoming called-number 40893334..  
codec g711ulaw  
fax protocol pass-through g711ulaw  
!  
dial-peer voice 408 voip  
description To Service Provider 7-digit dialing - SP facing  
shutdown  
destination-pattern .....  
session protocol sipv2  
session target sip-server  
session transport udp  
voice-class sip asserted-id pai  
dtmf-relay rtp-nte  
fax protocol pass-through g711ulaw  
!  
dial-peer voice 9003 voip  
description Fax calls using G.711u - SP facing  
translation-profile outgoing Outgoing-Fax-G711  
destination-pattern 81.....  
session protocol sipv2  
session target sip-server  
voice-class sip early-offer forced  
dtmf-relay rtp-nte
```



```
codec g711ulaw
fax protocol pass-through g711ulaw
!
dial-peer voice 9900 voip2
description to Service Provider - SME facing
session protocol sipv2
session transport udp
incoming called-number .....
voice-class sip asserted-id pai
dtmf-relay rtp-nte
codec g711ulaw
fax protocol pass-through g711ulaw
!
dial-peer voice 9901 voip
description Incoming to SME - SP facing
session protocol sipv2
session target sip-server
session transport tcp
incoming called-number 408933....
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 4408 voip
description To Service Provider 7-digit dialing - SME facing
shutdown
session protocol sipv2
session target sip-server
session transport udp
incoming called-number .....
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 9903 voip
description Fax calls using G.711u - SME facing
session protocol sipv2
session target sip-server
incoming called-number 81.....
voice-class sip early-offer forced
dtmf-relay rtp-nte
codec g711ulaw
fax protocol pass-through g711ulaw
!
dial-peer voice 9004 voip
description To Service Provider International calls - SP facing
destination-pattern 011T
session protocol sipv2
session target sip-server
session transport udp
voice-class sip early-offer forced
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 9904 voip
description To Service Provider International calls - SME facing
session protocol sipv2
```

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



```
session target sip-server
session transport udp
incoming called-number 011T
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 237 voip
description Outgoing to CM-Polaris
shutdown
destination-pattern 408273....
session protocol sipv2
session target ipv4:172.20.236.50
session transport tcp
!
dial-peer voice 408273 voip
description incoming call from VZ to SME
destination-pattern 408273398[12]
session protocol sipv2
session target ipv4:172.20.85.110:5060
session transport tcp
voice-class codec 1
no voice-class sip early-offer forced
dtmf-relay rtp-nte
!
dial-peer voice 1111 voip3
description Incoming-to-SME-172-20-109-254 SME-facing
destination-pattern 408273398.
session protocol sipv2
session target ipv4:172.20.109.254
session transport tcp
dtmf-relay rtp-nte
codec g711ulaw
!
dial-peer voice 1112 voip4
description Incoming-to-SME-172-20-109-254 SP-facing
session protocol sipv2
session target sip-server
session transport tcp
incoming called-number 408273398.
dtmf-relay rtp-nte
codec g711ulaw
!
!
sip-ua
set pstn-cause 1 sip-status 503
set pstn-cause 102 sip-status 503
retry invite 2
retry bye 2
retry cancel 2
timers trying 1000
sip-server ipv4:172.30.218.49:5147
g729-annexb override
!
!
line con 0
```

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

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



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

CUBE-ASR1K_Vz_152#



Acronyms

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



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