

Lync Server 2013 using SIP trunk to Cisco Unified Communications Manager Release 10.0

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Introduction

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 10.0 to interoperate with the Lync Server 2013 using SIP. End points are configured on both Cisco UCM and Lync Server with connectivity to PSTN. A SIP Trunk is configured between Cisco UCM and Cisco Unity for Voicemail connectivity, Lync Server users access Cisco Unity Voicemail via Cisco UCM.

Key Points:

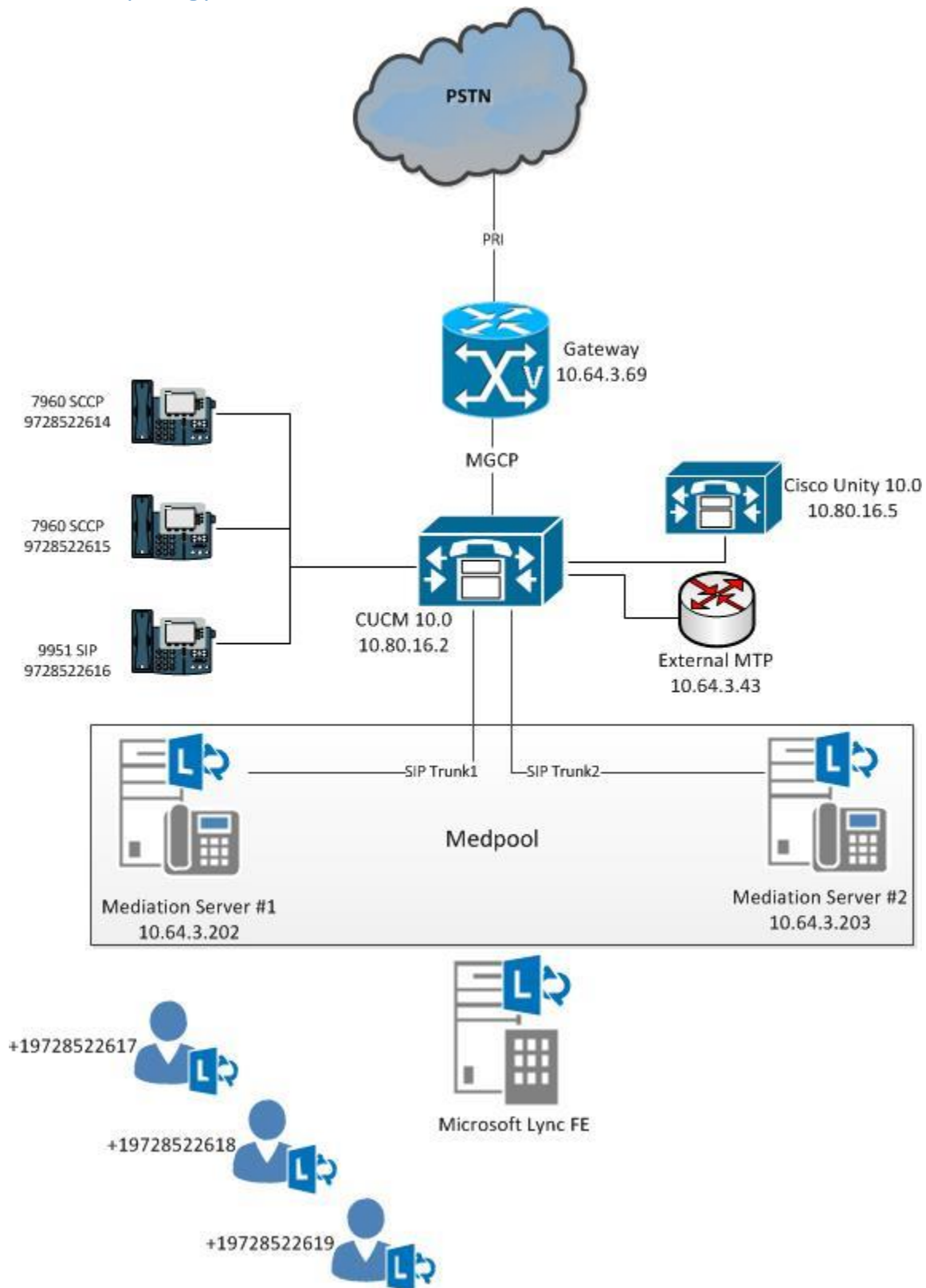
- The testing has been performed with only IPv4 using TCP for signaling.
- CISCO UCM is connected to the PSTN network via MGCP Gateway, as seen in the topology diagram.
- Basic call, call transfer with and without refer, call forwarding, conference call, call hold and resume, call park, RTCP, PRACK, Voice Mail work successfully.
- Testing was performed with Cisco UCM 10.0(1), but a later release fixing the defects CSCum00523 and CSCun13435 is required to resolve REFER and Call Hold issues.
- A Lua script is used to modify the bandwidth line during call hold, to manipulate the user=phone parameter in SIP URIs, to change History-Info headers in inbound INVITEs to Diversion headers, to change Referred-By headers to Diversion headers and to provide ring back at the call originator when PRACK is enabled on the SIP trunk.
- An external MTP is used on CISCO UCM to enable RTCP from CISCO UCM.
- "IP RTCP Interval threshold" on MGCP gateway is changed to 5000 to prevent Lync from dropping the call while the call is on hold with MOH enabled.
- Configuration of multiple SIP trunks and associated routing in Cisco UCM is necessary to support redundant Lync Mediation servers.

The following items were tested:

- Basic outbound and inbound calls between Lync and PSTN through Unified Communications Manager and verification of voice path.
- SIP Headers: E.164 and non-E.164, phone-context, long Request-URI
- Anonymous caller representation
- Codecs: G.711ulaw, G.711alaw, DTMF, Comfort Noise
- Early Media: PRACK, IVR
- RTP and RTCP
- Call transfer: attended, early unattended (only for Cisco endpoints) and blind (only for Lync endpoints).

- Call Park and retrieve
- Shared Lines on Cisco endpoints.
- Call forwarding: Call Forward Unconditional (CFU), simultaneous ring, Call Forward No Answer, Call Forward Busy (only for Cisco endpoints)
- Hold and resume with music on hold and without music on hold.
- Three-way conferencing
- Voice Mail

Network Topology



System Components

Hardware Requirements

The following hardware was tested

- Cisco UCS-C240-M3S VMWare Host running ESXi 5.5
- Microsoft Windows Server 2012 running Hyper-V
- Cisco 3925 Chassis as MGCP Gateway

Software Requirements

The following software was tested:

- Lync Server release 2013 and version 5.0.8308.556
- Cisco Unified Communications Manager version 10.0.1.10000-24
- Cisco Gateway version 15.4(1)T
- Cisco Unity version 10.0.1.10000-24
- A later release of Cisco UCM fixing the defects CSCum00523 and CSCun13435 is required to resolve REFER and Call Hold issues observed in testing.

Features

This section lists supported and unsupported features. Deviance from the configuration presented in this guide is not supported by Cisco. Please see the Limitations section below for more information.

Features Supported:

- Attended call transfer.
- Blind call transfer(only for Lync Endpoints)
- Early unattended transfer (only for Cisco Endpoints)
- Call forwarding unconditional
- Call Forward No Answer
- Call Forward Busy
- Hold and resume with and without refer
- Conference call
- Audio Codecs G711uLaw and G711aLaw
- RTCP
- Call Park
- Failover
- Early Media

- MWI on Cisco Phones
- Shared lines on Cisco Endpoints
- Voice Mail Deposit and Retrieval
- Message Waiting Indicator (only for Cisco Endpoints)

Features Not Supported or Not Tested:

- Message Waiting Indicator on Lync Endpoints

Limitations

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

- When simultaneous ring is set on Lync client to an IVR and PSTN user makes an inbound call to Lync, the call originator does not hear the early media from IVR.
- No message waiting indicator on Lync for voice mail. Lync rejects the NOTIFY from Unity as it does not have 'Notify' as either Supported or Allowed on the call leg to Unity
- Lync users do not receive Comfort Noise. Cisco provides local Comfort Noise via Cisco IP Phones and Gateways
- The external MTP configured on Unified Communications Manager does not pass-through the RTCP packets coming from Lync or the MGCP Gateway when it receives a=inactive in the SDP on call hold from Lync.
- When Unified Communications Manager and external MTP are configured for G.729 only and if it receives a call with G.711, Unified Communications Manager sends back a "503 Service Unavailable"

Cisco Configuration

The goal of this guide is to provide an overview of the integration between Cisco Unified Communication Manager, Cisco Unity and Lync Server. The deployment will interconnect the UC systems using SIP and using MGCP Gateway for PSTN connectivity to Unified Communications Manager. The following sections provide the required configurations for a successful integration.

Cisco Unified Communications Manager Configuration

SIP Trunk Security Profile Configuration

Navigation: System -> Security -> SIP Trunk Security Profile

1. Set **Name**: Enter a name for the security profile.
When you save the new profile, the name displays in the SIP Trunk Security Profile drop-down list box in the Trunk Configuration window.
2. Set **Description**: Enter a description relevant to your security profile
3. Confirm **Accept unsolicited notification**: is checked
If you want Cisco Unified Communications Manager to accept incoming non-INVITE, unsolicited notification messages that come via the SIP trunk, check this check box.
4. Confirm **Accept replaces header**: is checked
If you want Cisco Unified Communications Manager to accept new SIP dialogs, which have replaced existing SIP dialogs, check this check box

SIP Trunk Security Profile Configuration

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

Save

Delete

Copy

Reset

Apply Config

Add New

Status

Status: Ready

SIP Trunk Security Profile Information

Name*

Lync Security Profile

Description

Lync SIP Trunk Security Profile

Device Security Mode

Non Secure

Incoming Transport Type*

TCP+UDP

Outgoing Transport Type

TCP

☐ Enable Digest Authentication

Nonce Validity Time (mins)*

600

X.509 Subject Name

Incoming Port*

5060

☐ Enable Application level authorization

☐ Accept presence subscription

☐ Accept out-of-dialog refer**

☒ Accept unsolicited notification

☒ Accept replaces header

☐ Transmit security status

☐ Allow charging header

SIP V.150 Outbound SDP Offer Filtering*

Use Default Filter

Save

Delete

Copy

Reset

Apply Config

Add New

Figure 1: SIP Trunk Security Profile

SIP Trunk Security Profile Configuration

Related Links:
Back To Find/List
Go

Save
Delete
Copy
Reset
Apply Config
Add New

Status

Update successful
Reset of the trunk is required to have changes take effect.

SIP Trunk Security Profile Information

Name*

Lync to UC Security Profile

Description

Lync to UC SIP Trunk Security Profile

Device Security Mode

Non Secure

Incoming Transport Type*

TCP+UDP

Outgoing Transport Type

TCP

☐ Enable Digest Authentication

Nonce Validity Time (mins)*

600

X.509 Subject Name

Incoming Port*

5060

☐ Enable Application level authorization

☒ Accept presence subscription

☐ Accept out-of-dialog refer**

☒ Accept unsolicited notification

☒ Accept replaces header

☐ Transmit security status

☐ Allow charging header

SIP V.150 Outbound SDP Offer Filtering*

Use Default Filter

Save
Delete
Copy
Reset
Apply Config
Add New

Figure 2: Unity Connection Security Profile

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SIP Profile Configuration

Navigation: Device -> Device Settings -> SIP Profile

SIP Profile Information

1. Set **Name**: Enter a name for the SIP Profile
When you save the new profile, the name displays in the SIP Profile drop-down list box in the Trunk Configuration window
2. Set **Description**: Enter a description relevant to your profile
3. Set **SDP Session-level Bandwidth Modifier for Early Offer and Re-invites**: TIAS and AS
The Session Level Bandwidth Modifier specifies the maximum amount of bandwidth needed when all the media streams are used.
4. Confirm **Fall back to local RSVP**: is checked
5. Set **SIP Rel1XX Options**: Send PRACK if 1xx contains SDP
6. Confirm **Early Offer support for voice and video calls**: is unchecked
Check this check box to make this trunk support early offer.
7. Confirm **Send send-receive SDP in mid-call INVITE**: is unchecked
8. Confirm **SIP OPTIONS Ping**: is enabled

SIP Profile Configuration

Related Links:
Back To Find/List
Go

Save
Delete
Copy
Reset
Apply Config
Add New

Status

i Status: Ready

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

SIP Profile Information

Name*
Lync SIP Profile

Description

Default MTP Telephony Event Payload Type*
101

Early Offer for G.Clear Calls*
Disabled

User-Agent and Server header information*
Send Unified CM Version Information as User-Ager

Version in User Agent and Server Header*
Major And Minor

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

Confidential Access Level Headers*
Disabled

☐ Redirect by Application

☐ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Use Fully Qualified Domain Name in SIP Requests

☐ Assured Services SIP conformance

SDP Information

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

SDP Transparency Profile
Pass all unknown SDP attributes

Accept Audio Codec Preferences in Received Offer*
Default

☐ Require SDP Inactive Exchange for Mid-Call Media Change

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

Figure 3: SIP Profile Configuration -1

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
<input checked="" type="checkbox"/> Conference Join Enabled	
<input type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input checked="" type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	
<input type="checkbox"/> MLPP User Authorization	

Figure 4: SIP Profile Configuration-2

Normalization Script								
Normalization Script	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> < None > ▼ </div>							
<input type="checkbox"/> Enable Trace								
<table border="1" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="width: 5%;"></th> <th style="width: 40%;">Parameter Name</th> <th style="width: 40%;">Parameter Value</th> <th style="width: 15%;"></th> </tr> </thead> <tbody> <tr> <td style="text-align: center;">1</td> <td style="height: 20px;"></td> <td style="height: 20px;"></td> <td style="text-align: center;"> <div style="border: 1px solid #ccc; padding: 2px; display: inline-block;">+</div> <div style="border: 1px solid #ccc; padding: 2px; display: inline-block;">-</div> </td> </tr> </tbody> </table>		Parameter Name	Parameter Value		1			<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;">+</div> <div style="border: 1px solid #ccc; padding: 2px; display: inline-block;">-</div>
	Parameter Name	Parameter Value						
1			<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;">+</div> <div style="border: 1px solid #ccc; padding: 2px; display: inline-block;">-</div>					

Incoming Requests FROM URI Settings	
Caller ID DN	<div style="border: 1px solid #ccc; height: 20px;"></div>
Caller Name	<div style="border: 1px solid #ccc; height: 20px;"></div>

Trunk Specific Configuration	
Reroute Incoming Request to new Trunk based on*	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> Never ▼ </div>
RSVP Over SIP*	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> Local RSVP ▼ </div>
Resource Priority Namespace List	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> < None > ▼ </div>
<input checked="" type="checkbox"/> Fall back to local RSVP	
SIP Rel1XX Options*	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> Send PRACK if 1xx Contains SDP ▼ </div>
Video Call Traffic Class*	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> Mixed ▼ </div>
Calling Line Identification Presentation*	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> Default ▼ </div>
Session Refresh Method*	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> Invite ▼ </div>
<input type="checkbox"/> Enable ANAT	
<input type="checkbox"/> Deliver Conference Bridge Identifier	
<input type="checkbox"/> Early Offer support for voice and video calls (insert MTP if needed)	
<input type="checkbox"/> Allow Passthrough of Configured Line Device Caller Information	
<input type="checkbox"/> Reject Anonymous Incoming Calls	
<input type="checkbox"/> Reject Anonymous Outgoing Calls	
<input type="checkbox"/> Send ILS Learned Destination Route String	

SIP OPTIONS Ping	
<input checked="" type="checkbox"/> 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)*	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> 60 </div>
Ping Interval for Out-of-service Trunks (seconds)*	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> 120 </div>
Ping Retry Timer (milliseconds)*	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> 500 </div>
Ping Retry Count*	<div style="border: 1px solid #ccc; padding: 2px; display: inline-block;"> 6 </div>

SDP Information	
<input type="checkbox"/> Send send-receive SDP in mid-call INVITE	
<input type="checkbox"/> Allow Presentation Sharing using BFCP	
<input type="checkbox"/> Allow iX Application Media	
<input type="checkbox"/> Allow multiple codecs in answer SDP	

Save

Delete

Copy

Reset

Apply Config

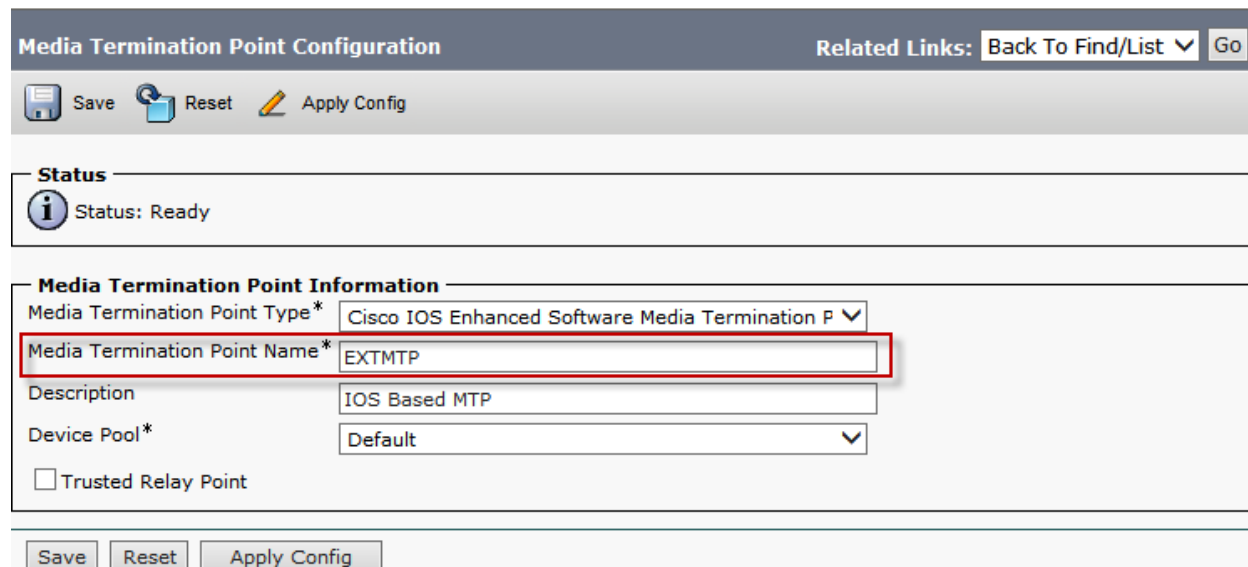
Add New

Figure 5: SIP Profile Configuration-3

Media Termination Point

Navigation: Media Resources->Media Termination Point

1. Set **Media Termination Point Name**: Enter the name of the external media termination point
2. Set **Device Pool**: Select the device pool, default device pool is used in this configuration.



Media Termination Point Configuration Related Links: [Back To Find/List](#) [Go](#)

[Save](#) [Reset](#) [Apply Config](#)

Status
Status: Ready

Media Termination Point Information

Media Termination Point Type* Cisco IOS Enhanced Software Media Termination P ▼

Media Termination Point Name* EXTMTP

Description IOS Based MTP

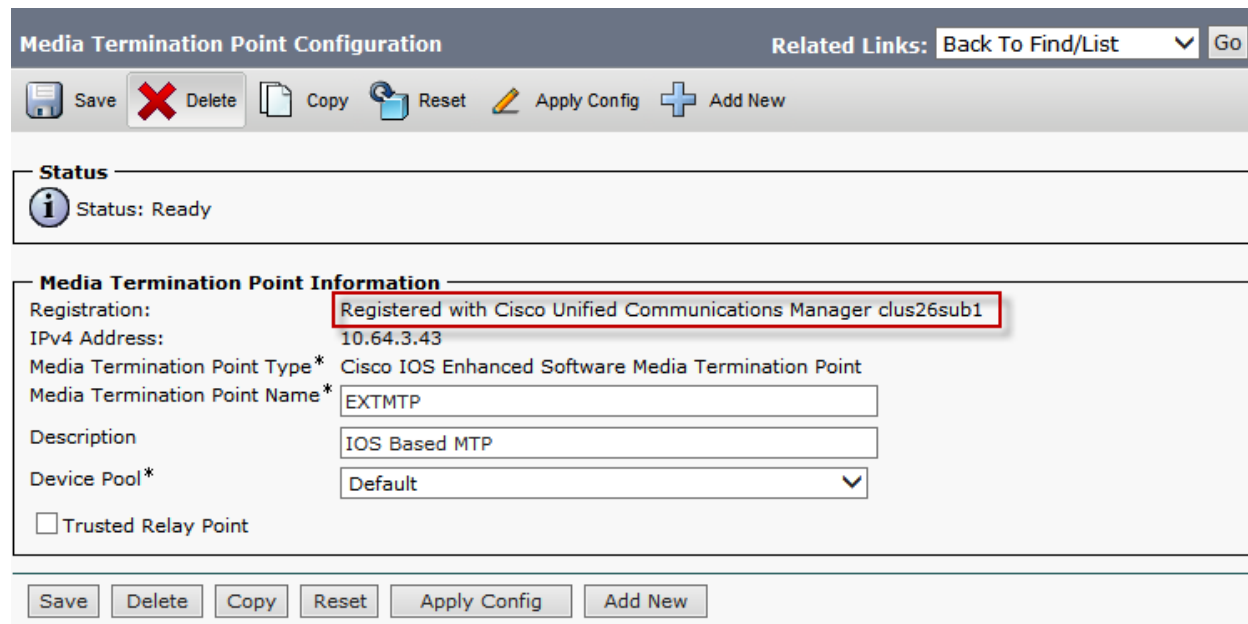
Device Pool* Default ▼

☐ Trusted Relay Point

[Save](#) [Reset](#) [Apply Config](#)

Figure 6: Media Termination Point-1

3. Confirm, the configured MTP is registered with the Unified Communications Manager



Media Termination Point Configuration Related Links: [Back To Find/List](#) [Go](#)

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

Status
Status: Ready

Media Termination Point Information

Registration: Registered with Cisco Unified Communications Manager clus26sub1

IPv4 Address: 10.64.3.43

Media Termination Point Type* Cisco IOS Enhanced Software Media Termination Point

Media Termination Point Name* EXTMTP

Description IOS Based MTP

Device Pool* Default ▼

☐ Trusted Relay Point

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





Figure 7: Media Termination Point-2


Media Resource Group

Navigation: Media Resources -> Media Resource Group

1. Add New Media Resource Group
2. Set Name: Enter a name for this group
3. Add Resources: Select the available resources as shown in the screen capture below.

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

 Save  Delete  Copy  Add New

Status
 Status: Ready

Media Resource Group Status
Media Resource Group: MRG_Lync (used by 3 devices)

Media Resource Group Information
Name*
Description

Devices for this Group
Available Media Resources**
ANN_2
ANN_4
CFB_2
CFB_4
MOH_2
Selected Media Resources*
ANN_3 (ANN)
CFB_3 (CFB)
EXTMTP (MTP)
MOH_3 (MOH)
☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Figure 8: Media Resource Group Configuration

Media Resource Group List Configuration

Navigation: Media Resources -> Media Resource Group List

1. Add New Media Resource Group List
2. Set **Name**: Enter name for this list
3. Select the media resource group you created under Media Resources -> Media Resource Group from the available groups

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

Status
Status: Ready

Media Resource Group List Status
Media Resource Group List: MRGL_Lync (used by 3 devices)

Media Resource Group List Information
Name*

Media Resource Groups for this List

Available Media Resource Groups
MRG_SW_MTP
MRG_SW_noMTP

Selected Media Resource Groups
MRG_Lync

Figure 9: Media Resource Group List Configuration

4. Repeat the same to create Media Resource Group Lists with MTP and without MTP. Select the relevant Media Resource Group for each list as shown in the screen captures below.

SIP Trunk Configuration to Lync

Navigation: Device -> Trunk

Trunks are created from Unified Communications Manager to each Lync Mediation Server for trunk failover and also to enable communications between Cisco UCM and Lync Mediation Servers. The FQDNs are used for configuring the trunks to the Lync Mediation Servers. However, due to the current limitation on Cisco UCM, if a SIP trunk is associated to a SIP-Route Pattern, the same trunk is not available to be included in a Route-List. This creates a need for a duplicate set of trunks to each Lync Mediation Server using IPv4 address. This makes the total number of trunks required to be four (two trunks using FQDN and two trunks using IP) to enable the provisioning of Route List and SIP Route Patterns to the Lync Mediation Servers.

Device Information

1. Set **Trunk Type**: SIP Trunk
2. Set **Device Protocol**: SIP
3. Set **Trunk Service Type**: None
4. Set **Device Name**: Enter a name for the trunk
5. Set **Description**: Enter a description relevant to your trunk
6. Set **Device Pool**: Default
For trunks, device pools specify a list of Cisco Unified Communications Managers that the trunk uses to distribute the call load dynamically
7. Set **Media Resource Group List**: MRGL_Lync, this is the list you created under Media Resources -> Media Resource Group List.
This list provides a prioritized grouping of media resource groups. An application chooses the required media resource, such as a Music on Hold server, from among the available media resources according to the priority order that a Media Resource Group List defines.
8. Confirm **Media Termination Point Required**: is checked
This check box is used to indicate whether a media termination point (MTP) is used to implement features that H.323 does not support (such as hold and transfer).
9. Confirm **Retry Video Calls as Audio**: is checked
10. Confirm **Run On All Active Unified CM Nodes**: is checked


SIP Information

11. Set the **Destination Address**: Enter the FQDN of the Mediation Server to which you are establishing a trunk.
12. Set **SIP trunk Security Profile**: Select the security profile you created under System -> Security -> SIP Security Profile
13. Set **SIP Profile**: Select the SIP Profile you created under Device -> Device Settings -> SIP Profile
14. Set **Normalization Script**: Select the normalization script to modify the bandwidth line $b=CT:64$ during call hold, ring back issue with PRACK enabled.


Trunk to Mediation Server 1 using FQDN

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

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


 Next

Status

 Status: Ready

Trunk Information

Trunk Type*	SIP Trunk
Device Protocol*	SIP
Trunk Service Type*	None(Default)

 Next


 *- indicates required item.

Figure 10: SIP Trunk to Lync Configuration -1_1

Trunk Configuration
Related Links: Back To Find/List

Save
Delete
Reset
Add New

Status
 Status: Ready

SIP Trunk Status
Service Status: Full Service
Duration: Time In Full Service: 0 day 4 hours 54 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Lync_Trunk
Description	
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_Lync
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 checked="" type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input checked="" type="checkbox"/> PSTN Access	
<input checked="" type="checkbox"/> Run On All Active Unified CM Nodes	

Figure 11: SIP Trunk to Lync Configuration -1_2

Intercompany Media Engine (IME)

E.164 Transformation Profile
< None >

MLPP and Confidential Access Level Information

MLPP Domain
< None >

Confidential Access Mode
< None >

Confidential Access Level
< None >

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type*
Default

SIP Privacy*
Default

Inbound Calls

Significant Digits*
All

Connected Line ID Presentation*
Default

Connected Name Presentation*
Default

Calling Search Space
< 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"/>

Figure 12: SIP Trunk to Lync Configuration -1_3

Incoming Called Party Settings

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

Clear Prefix Settings
Default Prefix Settings

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

Figure 13: SIP Trunk to Lync Configuration -1_4

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	med01.lyncclabsj.local		5060

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

Normalization Script

Normalization Script lync_interop

☒ Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

☒ None
☐ This trunk connects to a recording-enabled gateway
☐ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration





Geolocation < None >
Geolocation Filter < None >
☐ Send Geolocation Information


Save Delete Reset Add New

Figure 14: SIP Trunk to Lync Configuration -1_5

Trunk to Mediation Server 2 using FQDN

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

 Save  Delete  Reset  Add New

Status
 Status: Ready

SIP Trunk Status
Service Status: Full Service
Duration: Time In Full Service: 0 day 5 hours 0 minute

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Lync_trunk2
Description	
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
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 checked="" type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input checked="" type="checkbox"/> PSTN Access	
<input checked="" type="checkbox"/> Run On All Active Unified CM Nodes	

Figure 15: SIP Trunk to Lync Configuration -2_1

Intercompany Media Engine (IME)

E.164 Transformation Profile
< None >

MLPP and Confidential Access Level Information

MLPP Domain
< None >

Confidential Access Mode
< None >

Confidential Access Level
< None >

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type*
Default

SIP Privacy*
Default

Inbound Calls

Significant Digits*
All

Connected Line ID Presentation*
Default

Connected Name Presentation*
Default

Calling Search Space
< 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"/>

Figure 16: SIP Trunk to Lync Configuration -2_2

Incoming Called Party Settings

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

Clear Prefix Settings
Default Prefix Settings

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

Figure 17: SIP Trunk to Lync Configuration -2_3

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	med02.lyncclabsj.local		5060

MTP Preferred Originating Codec*
BLF Presence Group*
SIP Trunk Security Profile*
Rerouting Calling Search Space
Out-Of-Dialog Refer Calling Search Space
SUBSCRIBE Calling Search Space
SIP Profile*
DTMF Signaling Method*

711ulaw
Standard Presence group
Lync Security Profile
< None >
< None >
< None >
Lync SIP Profile
No Preference

View Details

Normalization Script

Normalization Script
☐ Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

☒ None
☐ This trunk connects to a recording-enabled gateway
☐ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation
Geolocation Filter
☐ Send Geolocation Information





< None >
< None >


Save Delete Reset Add New

Figure 18: SIP Trunk to Lync Configuration -2_4

Trunk to Mediation Server 1 using IP Address

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

 Save  Delete  Reset  Add New

Status
 Status: Ready

SIP Trunk Status
Service Status: Full Service
Duration: Time In Full Service: 0 day 5 hours 8 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Lync_SIP_Trunk_to_Med_1
Description	Lync SIP Trunk to Med 1
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_Lync
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 checked="" type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input checked="" type="checkbox"/> PSTN Access	
<input checked="" type="checkbox"/> Run On All Active Unified CM Nodes	

Figure 19: SIP Trunk to Lync Configuration -3_1

Intercompany Media Engine (IME)

E.164 Transformation Profile
< None >

MLPP and Confidential Access Level Information

MLPP Domain
< None >

Confidential Access Mode
< None >

Confidential Access Level
< None >

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type*
Default

SIP Privacy*
Default

Inbound Calls

Significant Digits*
All

Connected Line ID Presentation*
Default

Connected Name Presentation*
Default

Calling Search Space
< 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"/>

Figure 20: SIP Trunk to Lync Configuration -3_2

Incoming Called Party Settings

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

Clear Prefix Settings
Default Prefix Settings

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

Figure 21: SIP Trunk to Lync Configuration -3_3

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	10.64.3.202		5060

MTP Preferred Originating Codec* 711ulaw ▼
BLF Presence Group* Standard Presence group ▼
SIP Trunk Security Profile* Lync Security Profile ▼
Rerouting Calling Search Space < None > ▼
Out-Of-Dialog Refer Calling Search Space < None > ▼
SUBSCRIBE Calling Search Space < None > ▼
SIP Profile* Lync SIP Profile ▼ [View Details](#)
DTMF Signaling Method* RFC 2833 ▼

Normalization Script

Normalization Script lync_interop ▼
☒ Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

☒ None
☐ This trunk connects to a recording-enabled gateway
☐ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration





Geolocation < None > ▼
Geolocation Filter < None > ▼
☐ Send Geolocation Information


Save Delete Reset Add New

Figure 22: SIP Trunk to Lync Configuration -3_4

Trunk to Mediation Server 2 using IP Address

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

 Save  Delete  Reset  Add New

Status
 Status: Ready

SIP Trunk Status
Service Status: Full Service
Duration: Time In Full Service: 0 day 5 hours 14 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Lync_SIP_Trunk_to_Med_2
Description	Lync_SIP_Trunk_to_Med_2
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_Lync
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 checked="" type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input checked="" type="checkbox"/> PSTN Access	
<input checked="" type="checkbox"/> Run On All Active Unified CM Nodes	

Figure 23: SIP Trunk to Lync Configuration -4_1

Intercompany Media Engine (IME)														
E.164 Transformation Profile < None > ▼														
MLPP and Confidential Access Level Information														
MLPP Domain < None > ▼														
Confidential Access Mode < None > ▼														
Confidential Access Level < None > ▼														
Call Routing Information														
<input checked="" type="checkbox"/> Remote-Party-Id														
<input checked="" type="checkbox"/> Asserted-Identity														
Asserted-Type* Default ▼														
SIP Privacy* Default ▼														
Inbound Calls														
Significant Digits* All ▼														
Connected Line ID Presentation* Default ▼														
Connected Name Presentation* Default ▼														
Calling Search Space < None > ▼														
AAR Calling Search Space < None > ▼														
Prefix DN <input style="width: 150px;" type="text"/>														
<input type="checkbox"/> Redirecting Diversion Header Delivery - Inbound														
Incoming Calling Party Settings														
<p>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.</p> <div style="text-align: center; margin-bottom: 10px;"> Clear Prefix Settings Default Prefix Settings </div> <table border="1" style="width: 100%; border-collapse: collapse; background-color: #f2f2f2;"> <thead> <tr> <th style="width: 15%;">Number Type</th> <th style="width: 20%;">Prefix</th> <th style="width: 10%;">Strip Digits</th> <th style="width: 40%;">Calling Search Space</th> <th style="width: 15%;">Use Device Pool CSS</th> </tr> </thead> <tbody> <tr> <td>Incoming Number</td> <td><input style="width: 100%;" type="text" value="Default"/></td> <td><input style="width: 50px;" type="text" value="0"/></td> <td>< None > ▼</td> <td style="text-align: center;"><input checked="" type="checkbox"/></td> </tr> </tbody> </table>					Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS	Incoming Number	<input style="width: 100%;" type="text" value="Default"/>	<input style="width: 50px;" type="text" value="0"/>	< None > ▼	<input checked="" type="checkbox"/>
Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS										
Incoming Number	<input style="width: 100%;" type="text" value="Default"/>	<input style="width: 50px;" type="text" value="0"/>	< None > ▼	<input checked="" type="checkbox"/>										

Figure 24: SIP Trunk to Lync Configuration -4_2

Incoming Called Party Settings

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

Clear Prefix Settings
Default Prefix Settings

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

Figure 25: SIP Trunk to Lync Configuration -4_3

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	10.64.3.203		5060

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

[View Details](#)

Normalization Script

Normalization Script lync_interop

☒ Enable Trace

	Parameter Name	Parameter Value	
1			+ -

Recording Information

☒ None
☐ This trunk connects to a recording-enabled gateway
☐ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation < None >
Geolocation Filter < None >
☐ Send Geolocation Information

Save Delete Reset Add New

Figure 26: SIP Trunk to Lync Configuration -4_4

Trunk to Cisco Unity Voice Mail Server

Navigation: Device -> Trunk

The following procedure describes the trunk configuration from Unified Communications Manager to Cisco Unity Voice Mail Server

Device Information

1. Set **Trunk Type**: SIP Trunk
2. Set **Device Protocol**: SIP
3. Set **Trunk Service Type**: None
4. Set **Device Name**: Enter a name for the trunk
5. Set **Description**: Enter a description relevant to your trunk
6. Set **Device Pool**: Default
For trunks, device pools specify a list of Cisco Unified Communications Managers that the trunk uses to distribute the call load dynamically
7. Confirm **Media Termination Point Required**: is checked
This check box is used to indicate whether a media termination point (MTP) is used to implement features that H.323 does not support (such as hold and transfer).
8. Confirm **Retry Video Calls as Audio**: is checked

SIP Information

9. Set the **Destination Address**: Enter the FQDN/IP Address of the Unity Server to which you are establishing a trunk.
10. Set **SIP trunk Security Profile**: Select the security profile you created under System -> Security -> SIP Security Profile

Trunk Configuration

Related Links:
Back To Find/List

Save

Delete

Reset

Add New

Status

Status: Ready

SIP Trunk Status

Service Status: Unknown

Duration:
Time In Full Service: 0 day 22 hours 18 minutes

Device Information

Product:

SIP Trunk

Device Protocol:

SIP

Trunk Service Type

None(Default)

Device Name*

Unity_Connection

Description

Device Pool*

Default

Common Device Configuration

< None >

Call Classification*

Use System Default

Media Resource Group List

< None >

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

Figure 27: SIP Trunk to Cisco Unity-1

<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 > ▼

MLPP and Confidential Access Level Information	
MLPP Domain	< None > ▼
Confidential Access Mode	< None > ▼
Confidential Access Level	< None > ▼

Call Routing Information	
<input checked="" type="checkbox"/> Remote-Party-Id	
<input checked="" type="checkbox"/> Asserted-Identity	
Asserted-Type*	Default ▼
SIP Privacy*	Default ▼

Inbound Calls	
Significant Digits*	All ▼
Connected Line ID Presentation*	Default ▼
Connected Name Presentation*	Default ▼
Calling Search Space	< None > ▼
AAR Calling Search Space	< None > ▼
Prefix DN	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Inbound	

Figure 28: SIP Trunk to Cisco Unity-2

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"/>

Incoming Called Party Settings

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

Clear Prefix Settings
Default Prefix Settings

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

Figure 29: SIP Trunk to Cisco Unity-3

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 *	10.80.16.5		5060

MTP Preferred Originating Codec* 711ulaw

BLF Presence Group* Standard Presence group

SIP Trunk Security Profile* Lync to UC Security Profile

Rerouting Calling Search Space < None >

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

SUBSCRIBE Calling Search Space < None >

SIP Profile* Standard SIP Profile [View Details](#)

DTMF Signaling Method* No Preference

Normalization Script

Normalization Script < None >

☐ Enable Trace

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

Recording Information

☒ None

☐ This trunk connects to a recording-enabled gateway

☐ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation < None >

Geolocation Filter < None >

☐ Send Geolocation Information

Figure 30: SIP Trunk to Cisco Unity-4

Create Normalization Script

Navigation: Device->Device Settings->Normalization Script

This normalization script is used to modify the bandwidth line during call hold, to manipulate the user=phone parameter in SIP URIs, to modify History-Info on inbound INVITEs to Diversion header, and Referred-by to Diversion header and to provide ring back at the call originator when PRACK is enabled on trunk. Below is snap shot of Script configuration.

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

Status
 Status: Ready

SIP Normalization Script Info

Name*

Description

Content*

```
--[[
Description:
  Provides interoperability for Micro Soft Lync

Handle Below Scenarios

1. Add user=phone for all outbound Invite messages because it
is
   mandatory for Lync

2. Change the CT Line values to 1000 , Moderate bandwidth in all
   outgoing messages from CUCM to Lync
--]]
```

Script Execution Error Recovery Action*

System Resource Error Recovery Action*

Memory Threshold* kilobytes

Lua Instruction Threshold* instructions

Figure 31: Normalization Script

CISCO UCM Normalization Script

Download the script “SIP normalization script (version 1.2) for audio interoperability between Microsoft Lync 2013 and Cisco Unified Communications Manager” (lync_interop_lua.zip) at Downloads Home > Products > Unified Communications > Call Control > Cisco Unified Communications Manager (CallManager) > Cisco Unified Communications Manager Version 10.0 > SIP Normalization and Transparency Scripts-Scripts:

<http://software.cisco.com/download/release.html?i=ly&mdfid=284603137&softwareid=284695022&release=Scripts&os=>

Translation Pattern

Navigation: Call Routing-> Translation Pattern Configuration

1. Set **Translation Pattern**: Enter the ten digit number pattern to be translated
2. Set **Called Party Transform Mask**: Enter the four digit number pattern to be translated to, these will be the Cisco Phone extension pattern.

Translation Pattern Configuration
Related Links: Back To Find/List Go

Save Delete Copy Add New

Status
Update successful

Pattern Definition

Translation Pattern	9728522618
Partition	< None >
Description	
Numbering Plan	< None >
Route Filter	< None >
MLPP Precedence*	Default
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Calling Search Space	< None >
External Call Control Profile	< None >
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
<input checked="" type="checkbox"/> Provide Outside Dial Tone <input checked="" type="checkbox"/> Urgent Priority <input type="checkbox"/> Route Next Hop By Calling Party Number	

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	2618
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

Save Delete Copy Add New

Figure 32: Translation Pattern

Route Group

Navigation: Call Routing -> Route/Hunt -> Route Group

1. Set **Route Group Name**: Enter a name for the route group
2. Set **Distribution Algorithm**: Select the preferred distribution algorithm from the available list

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3. Select the available devices and them to the route group, once selected they should be available in the **Current Route Group Members**

Route Group Configuration
Related Links: Back To Find/List
Go

Save
Delete
Add New

Status
i Status: Ready

Route Group Information
Route Group Name * Lync Route Group
Distribution Algorithm * Circular

Route Group Member Information

Find Devices to Add to Route Group
Device Name contains Find
Available Devices ** Lync_SIP_Trunk_to_Med_1
Lync_SIP_Trunk_to_Med_2
Port(s) None Available
Add to Route Group

Current Route Group Members

Selected Devices (ordered by priority) * Lync_SIP_Trunk_to_Med_2 (All Ports)
Lync_SIP_Trunk_to_Med_1 (All Ports)

Removed Devices ***

Reverse Order of Selected Devices

Route Group Members

Lync SIP Trunk to Med 2
Lync SIP Trunk to Med 1

Save Delete Add New

Figure 33: Route Group

Route List

Navigation: Call Routing -> Route/Hunt -> Route List

1. Set **Name**: Enter a name for the route list

2. Set **Cisco Unified Communications Manager Group**: Default
3. Add Route Group: Under Route List Member Information, click the **Add Route Group** and add the route group configured in the previous section.

Route List Configuration

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

Save

Delete

Copy

Reset

Apply Config

Add New

Status

Status: Ready

Route List Information

Registration

Registered with Cisco Unified Communications Manager clus26pub

IP Address

10.80.16.2

Device is trusted

Name*

Lync Route List

Description

Lync Route List

Cisco Unified Communications Manager Group*

Default

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

Run On All Active Unified CM Nodes

Route List Member Information

Selected Groups**

Lync Route Group

Add Route Group

Removed Groups***

Route List Details

Lync Route Group

Save

Delete

Copy

Reset

Apply Config

Add New

Figure 34: Route List

Route Pattern Configuration

Route Pattern to Lync

Navigation: Call Routing -> Route/Hunt -> Route Pattern

Pattern Definition

1. Set **Route Pattern**: Enter the routing pattern
2. Set **Gateway/Route List**: Select the Route List you have created under Call Routing -> Route/Hunt -> Route List

Calling Party Transformations

3. Set **Prefix Digits (Outgoing Calls)**: +1

Called Party Transformations

4. Set **Prefix Digits(Outgoing Calls)**: +1

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

Save
Delete
Copy
Add New

Status
i Status: Ready

Pattern Definition
Route Pattern* 97285226XX
Route Partition < None >
Description Route Pattern from CUCM to Lync 2013
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* Lync Route List (Edit)
Route Option
☒ Route this pattern
☐ Block this pattern No Error
Call Classification* OnNet
External Call Control Profile < None >
☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level* 0
☐ Require Client Matter Code

Calling Party Transformations
☐ Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls) +1
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling Party Number Type* Cisco CallManager
Calling Party Numbering Plan* Cisco CallManager

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

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

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

Save Delete Copy Add New

Figure 35: Route Pattern -1

Route Pattern to Cisco Unity

Pattern Definition

1. Set **Route Pattern**: Enter the routing pattern
2. Set **Gateway/Route List**: Select the Trunk you have created under Device ->Trunk

Route Pattern ConfigurationRelated Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Add New

Status
 Status: Ready


Pattern Definition


Route Pattern*	1000
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*	Unity_Connection (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	

Calling Party Transformations


<input type="checkbox"/> 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* 


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 Val
<input data-bbox="211 724 682 756" type="text" value="-- Not Selected --"/> 	<input data-bbox="690 724 1209 756" type="text" value=" < Not Exist > "/>	<input data-bbox="1218 724 1429 756" type="text"/>

Route Pattern to Gateway

Navigation: Call Routing -> Route/Hunt -> Route Pattern

1. Set **Route Pattern**: \+1XXXXXXXXXX
2. Set **Gateway/Route List**: Select the End-Point you have created in the gateway configuration under Device -> Gateway
3. Set Called Party Transform Mask: XXXXXXXXXXXX

Route Pattern Configuration
Related Links: Back To Find/List

Save
Delete
Copy
Add New

Status
Status: Ready

Pattern Definition
Route Pattern* \+1XXXXXXXXXX
Route Partition < None >
Description Route Pattern to GW to PSTN
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* S0/SU0/DS1-0@cisco-lync-gw.lab.tekvizion.com (Edit)
Route Option
☒ Route this pattern
☐ Block this pattern No Error
Call Classification* OnNet
External Call Control Profile < None >
☐ 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 XXXXXXXXXXXX
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 XXXXXXXXXXXX
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

Figure 36: Route Pattern to Gateway

SIP Route Pattern

Navigation: Call Routing -> SIP Route Pattern

1. Set **IPv4 Pattern**: Enter the FQDN of the mediation server
2. Set **Description**: Enter the description of the SIP Route Pattern
3. Set **SIP Trunk**: From the drop-down list select your trunk to Lync Server [this will be the trunk, other than the one added in route group].

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

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

Status

Status: Ready

Pattern Definition

Pattern Usage: Domain Routing

IPv4 Pattern*: med01.lynclabsj.local

IPv6 Pattern:

Description:

Route Partition: < None >

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

☐ 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](#)

Figure 37: SIP Route Pattern-1

Create SIP Route Patterns similar to the above configuration to all mediation servers in the mediation pool as shown below

SIP Route Pattern Configuration

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

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Pattern Usage

Domain Routing

IPv4 Pattern*

med02.lynclabsj.local

IPv6 Pattern

Description

Route Partition

< None >

SIP Trunk/Route List*

Lync_trunk2

[\(Edit\)](#)

☐ 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

Figure 38: SIP Route Pattern-2

Cisco End Point Configuration

Navigation: Device->Phone

SIP Phone Configuration

Phone Type	
Product Type:	Cisco 9951
Device Protocol:	SIP
Real-time Device Status	
Registration:	Registered with Cisco Unified Communications Manager clus26sub1
IPv4 Address:	10.80.16.20
Active Load ID:	sip9951.9-4-1-9
Inactive Load ID:	sip9951.9-3-2-10
Download Status:	Successful
Device Information	
<input checked="" type="checkbox"/> Device is Active	
<input checked="" type="checkbox"/> Device is trusted	
MAC Address*	1C17D3407EE3
Description	SEP1C17D3407EE3
Device Pool*	Default View Details
Common Device Configuration	< None > View Details
Phone Button Template*	Standard 9951 SIP
Softkey Template	< None >
Common Phone Profile*	Standard Common Phone Profile View Details
Calling Search Space	< None >
AAR Calling Search Space	< None >
Media Resource Group List	< None >
User Hold MOH Audio Source	< None >
Network Hold MOH Audio Source	< None >
Location*	Hub_None
AAR Group	< None >
User Locale	< None >
Network Locale	< None >
Built In Bridge*	Default
Privacy*	Default

Figure 39: SIP Phone Configuration-1

Device Mobility Mode*	Default	View Current Device Mobility Settings
Owner	<input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)	
Owner User ID		
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 >	
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only) <input checked="" type="checkbox"/> Allow Control of Device from CTI <input checked="" type="checkbox"/> Logged Into Hunt Group <input type="checkbox"/> Remote Device <input type="checkbox"/> Protected Device**** <input type="checkbox"/> Require off-premise location		

- Number Presentation Transformation -

Caller ID For Calls From This Phone

Calling Party Transformation CSS < None >

☒ Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

Remote Number

Calling Party Transformation CSS < None >

☒ Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

- Protocol Specific Information -

Packet Capture Mode*	None
----------------------	------

Figure 40: SIP Phone Configuration-2

Packet Capture Duration	<input type="text" value="0"/>
BLF Presence Group*	<input type="text" value="Standard Presence group"/> ▼
SIP Dial Rules	<input type="text" value=" < None >"/> ▼
MTP Preferred Originating Codec*	<input type="text" value="711ulaw"/> ▼
Device Security Profile*	<input type="text" value="Cisco 9951 - Standard SIP Non-Secure Profile"/> ▼
Rerouting Calling Search Space	<input type="text" value=" < None >"/> ▼
SUBSCRIBE Calling Search Space	<input type="text" value=" < None >"/> ▼
SIP Profile*	<input type="text" value="Standard SIP Profile"/> ▼ View Details
Digest User	<input type="text" value=" < None >"/> ▼

☐ Media Termination Point Required
☐ Unattended Port
☐ Require DTMF Reception

- Certification Authority Proxy Function (CAPF) Information -

Certificate Operation*	<input type="text" value="No Pending Operation"/> ▼
Authentication Mode*	<input type="text" value="By Null String"/> ▼
Authentication String	<input type="text"/>
<input type="button" value="Generate String"/>	
Key Size (Bits)*	<input type="text" value="1024"/> ▼
Operation Completes By	<input type="text" value="2014"/> <input type="text" value="6"/> <input type="text" value="8"/> <input type="text" value="12"/> (YYYY:MM:DD:HH)
Certificate Operation Status: None	
Note: Security Profile Contains Addition CAPF Settings.	

- Expansion Module Information -

Module 1	<input type="text" value=" < None >"/> ▼
Module 1 Load Name	<input type="text"/>
Module 2	<input type="text" value=" < None >"/> ▼
Module 2 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"/>

Figure 41: SIP Phone Configuration-3

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

☐ Enable Extension Mobility

Log Out Profile ▼

Log in Time < None >

Log out Time < None >

MLPP and Confidential Access Level Information

MLPP Domain	<input type="text" value="< None >"/> ▼
MLPP Indication*	<input type="text" value="Default"/> ▼
MLPP Preemption*	<input type="text" value="Default"/> ▼
Confidential Access Mode	<input type="text" value="< None >"/> ▼
Confidential Access Level	<input type="text" value="< None >"/> ▼

Do Not Disturb

☐ Do Not Disturb

DND Option* ▼

DND Incoming Call Alert ▼

Secure Shell Information

Secure Shell User	<input type="text"/>
Secure Shell Password	<input type="password"/>

Figure 42: SIP Phone Configuration-4

Product Specific Configuration Layout

	Param	Override Common 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 *	Disabled	<input type="checkbox"/>
Console Access *	Disabled	<input type="checkbox"/>
Video Capabilities *	Disabled	<input type="checkbox"/>
Enable/Disable USB Classes	Mass Storage Human Interface Device Audio Class	<input type="checkbox"/>
Bluetooth *	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 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"/>

Figure 43: SIP Phone Configuration-5


<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>
EnergyWise Domain	<input type="text"/>	<input type="checkbox"/>
EnergyWise Endpoint Security Secret	<input type="text"/>	<input type="checkbox"/>
<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>
Span to PC Port*	Disabled ▾	
Logging Display*	Disabled ▾	
Load Server	<input type="text"/>	<input type="checkbox"/>
IPv6 Load Server	<input type="text"/>	<input type="checkbox"/>
Recording Tone*	Disabled ▾	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration	<input type="text"/>	
Display On When Incoming Call*	Enabled ▾	<input type="checkbox"/>
RTCP*	Enabled ▾	<input checked="" type="checkbox"/>
Log Server	<input type="text"/>	<input type="checkbox"/>
IPv6 Log Server	<input type="text"/>	<input type="checkbox"/>
Remote Log*	Disabled ▾	<input type="checkbox"/>
Log Profile	Default Preset Telephony	<input type="checkbox"/>
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	<input type="text"/>	
LLDP Power Priority*	Unknown ▾	
802.1x Authentication*	User Controlled ▾	<input type="checkbox"/>

Figure 44: SIP Phone Configuration-6

FIPS Mode*	Disabled	<input type="checkbox"/>
Detect Unified CM Connection Failure*	Normal	<input 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"/>
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 SRTCP*	Disabled	<input type="checkbox"/>
RTCP for Video*	Enabled	<input type="checkbox"/>
Record Call Log from Shared Line*	Disabled	<input type="checkbox"/>
Show Call History for Selected Line Only.*	Disabled	<input type="checkbox"/>
Actionable Incoming Call Alert*	Disabled	<input type="checkbox"/>
DF bit*	0	<input type="checkbox"/>
Default Line Filter		
Separate Audio and Video Mute*	Disabled	<input type="checkbox"/>
Softkey Control*	Feature Control Policy	<input type="checkbox"/>
Start Video Port		<input type="checkbox"/>
Stop Video Port		<input type="checkbox"/>
Lowest Alerting Line State Priority*	Disabled	<input type="checkbox"/>
TLS Resumption Timer*	3600	<input type="checkbox"/>

Figure 45: SIP Phone Configuration-7

Status

 Add successful

Directory Number Information

Directory Number*
2616

Route Partition
< None >

Description
SIP Phone Extension

Alerting Name
SIP Phone

ASCII Alerting Name
SIP Phone

External Call Control Profile
< None >

☒ Allow Control of Device from CTI

Associated Devices

SEP1C17D3407EE3

Edit Device
Edit Line Appearance

Dissociate Devices

Directory Number Settings

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

Calling Search Space
< None >

BLF Presence Group*
Standard Presence group

User Hold MOH Audio Source
< None >

Network Hold MOH Audio Source
< None >

Auto Answer*
Auto Answer Off

☐ Reject Anonymous Calls


Enterprise Alternate Number

Add Enterprise Alternate Number

+E.164 Alternate Number

Add +E.164 Alternate Number

Figure 46: SIP Phone Configuration-8

Directory URIs				
Primary	URI	Partition	Advertise Globally via ILS	Remove
<input checked="" type="radio"/>	<input type="text"/>	< None >	<input checked="" type="checkbox"/>	
<input type="button" value="Add Row"/>				

PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing

Advertised Failover Number

AAR Settings

	Voice Mail	AAR Destination Mask	AAR Group
AAR	<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			Use System Default
Forward All	<input checked="" 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 checked="" type="checkbox"/> or	<input type="text"/>	< None >
Forward No Answer External	<input checked="" 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	<input checked="" type="checkbox"/> or	<input type="text"/>	< None >

Figure 47: SIP Phone Configuration-9

Forward Unregistered External	<input checked="" 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 > ▼ A blank value means to call the parker's line.
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"/> service parameter	A blank value will use value set in Park Monitoring Reversion Timer

- MLPP Alternate Party And Confidential Access Level Settings

Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None > ▼
MLPP No Answer Ring Duration (seconds)	<input type="text"/>
Confidential Access Mode	< None > ▼
Confidential Access Level	< None > ▼
Call Control Agent Profile	< None > ▼

- 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 ▼	

Figure 48: SIP Phone Configuration-10

Line 1 on Device SEP1C17D3407EE3

Display (Caller ID)
SIP Phone
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.

ASCII Display (Caller ID)
SIP Phone

Line Text Label
SIP Phone

External Phone Number Mask

Visual Message Waiting Indicator Policy*
Use System Policy

Audible Message Waiting Indicator Policy*
Default

Ring Setting (Phone Idle)*
Use System Default

Ring Setting (Phone Active)
Use System Default
Applies to this line when any line on the phone has a call in progress.

Call Pickup Group Audio Alert Setting (Phone Idle)
Use System Default

Call Pickup Group Audio Alert Setting (Phone Active)
Use System Default

Recording Option*
Call Recording Disabled

Recording Profile
< None >

Recording Media Source*
Gateway Preferred

Monitoring Calling Search Space
< None >

☒ Log Missed Calls

Multiple Call/Call Waiting Settings on Device SEP1C17D3407EE3

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 SEP1C17D3407EE3

☒ Caller Name
☐ Caller Number
☐ Redirected Number
☒ Dialed Number

Users Associated with Line

Associate End Users

Save
Delete
Reset
Apply Config
Add New

Figure 49: SIP Phone Configuration-11

SCCP Phone Configuration

- Phone Type	
Product Type:	Cisco 7960
Device Protocol:	SCCP
- Real-time Device Status	
Registration:	Registered with Cisco Unified Communications Manager clus26sub1
IPv4 Address:	10.80.16.17
Active Load ID:	None
Download Status:	None
- Device Information	
<input checked="" type="checkbox"/> Device is Active	
<input checked="" type="checkbox"/> Device is trusted	
MAC Address*	0014A998C3FC
Description	SEP0014A998C3FC
Device Pool*	Default View Details
Common Device Configuration	< None > View Details
Phone Button Template*	Standard 7960 SCCP
Softkey Template	< None >
Common Phone Profile*	Standard Common Phone Profile View Details
Calling Search Space	< None >
AAR Calling Search Space	< None >
Media Resource Group List	< None >
User Hold MOH Audio Source	< None >
Network Hold MOH Audio Source	< None >
Location*	Hub_None
AAR Group	< None >
User Locale	< None >
Network Locale	< None >
Built In Bridge*	Default
Privacy*	Default
Device Mobility Mode*	Default View Current Device Mobility Settings

Figure 50: SCCP Phone Configuration-1

Owner	<input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)
Owner User ID	<input type="text"/>
Phone Load Name	<input type="text"/>
Join Across Lines	<input type="text" value="Default"/>
Use Trusted Relay Point*	<input type="text" value="Default"/>
BLF Audible Alert Setting (Phone Idle)*	<input type="text" value="Default"/>
BLF Audible Alert Setting (Phone Busy)*	<input type="text" value="Default"/>
Always Use Prime Line*	<input type="text" value="Default"/>
Always Use Prime Line for Voice Message*	<input type="text" value="Default"/>
Geolocation	<input type="text" value=" < None >"/>
<input checked="" type="checkbox"/> Retry Video Call as Audio <input type="checkbox"/> Ignore Presentation Indicators (internal calls only) <input checked="" type="checkbox"/> Allow Control of Device from CTI <input checked="" type="checkbox"/> Logged Into Hunt Group <input type="checkbox"/> Remote Device <input type="checkbox"/> Hot line Device*****	

- Number Presentation Transformation -

Caller ID For Calls From This Phone

Calling Party Transformation CSS

☒ Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

Remote Number

Calling Party Transformation CSS

☒ Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

- Protocol Specific Information -

Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>
BLF Presence Group*	<input type="text" value="Standard Presence group"/>
Device Security Profile*	<input type="text" value="Cisco 7960 - Standard SCCP Non-Secure Profile"/>
SUBSCRIBE Calling Search Space	<input type="text" value=" < None >"/>
<input type="checkbox"/> Unattended Port	

Figure 51: SCCP Phone Configuration-2

☐ Require DTMF Reception

☐ RFC2833 Disabled

Certification Authority Proxy Function (CAPF) Information

Certificate Operation* No Pending Operation ▼

Authentication Mode* By Null String ▼

Authentication String

Key Size (Bits)* 1024 ▼

Operation Completes By 2014 6 8 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

Figure 52: SCCP Phone Configuration-3

- Extension Information -

☐ Enable Extension Mobility

Log Out Profile -- Use Current Device Settings -- ▼

Log in Time < None >

Log out Time < None >

- MLPP and Confidential Access Level Information -

MLPP Domain < None > ▼

MLPP Indication* Default ▼

MLPP Preemption* Default ▼

Confidential Access Mode < None > ▼

Confidential Access Level < None > ▼


- Do Not Disturb -

☐ Do Not Disturb

DND Option* Ringer Off ▼

DND Incoming Call Alert < None > ▼

- Product Specific Configuration Layout -



☐ Disable Speakerphone

☐ Disable Speakerphone and Headset

PC Port * Enabled ▼

Settings Access* Enabled ▼

Gratuitous ARP* Enabled ▼

PC Voice VLAN Access* Enabled ▼

Video Capabilities* Disabled ▼

Auto Line Select* Disabled ▼

Web Access* Enabled ▼

Figure 53: SCCP Phone Configuration-4

Status

 Status: Ready

Directory Number Information

Directory Number*
☐ Urgent Priority

Route Partition

Description

Alerting Name

ASCII Alerting Name

External Call Control Profile

☒ 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

Enterprise Alternate Number

Add Enterprise Alternate Number

+E.164 Alternate Number

Add +E.164 Alternate Number

Figure 54: SCCP Phone Configuration-7

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Directory URIs

Primary	URI	Partition	Advertise Globally via ILS	Remove
<input checked="" type="radio"/>	<input type="text"/>	< None >	<input checked="" type="checkbox"/>	
<input type="button" value="Add Row"/>				

PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing

Advertised Failover Number

AAR Settings

	Voice Mail	AAR Destination Mask	AAR Group
AAR	<input type="checkbox"/> or <input type="text"/>	<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			Use System Default
Forward All	<input type="checkbox"/> or <input type="text"/>	<input type="text"/>	< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or <input type="text"/>	<input type="text"/>	< None >
Forward Busy External	<input type="checkbox"/> or <input type="text"/>	<input type="text"/>	< None >
Forward No Answer Internal	<input type="checkbox"/> or <input type="text"/>	<input type="text"/>	< None >
Forward No Answer External	<input type="checkbox"/> or <input type="text"/>	<input type="text"/>	< None >
Forward No Coverage Internal	<input type="checkbox"/> or <input type="text"/>	<input type="text"/>	< None >
Forward No Coverage External	<input type="checkbox"/> or <input type="text"/>	<input type="text"/>	< None >
Forward on CTI Failure	<input type="checkbox"/> or <input type="text"/>	<input type="text"/>	< None >
Forward Unregistered Internal	<input type="checkbox"/> or <input type="text"/>	<input type="text"/>	< None >

Figure 55: SCCP Phone Configuration-8

Forward Unregistered External	or	<input type="text"/>	<input type="text" value=" < None >"/>
No Answer Ring Duration (seconds)	<input type="text"/>		
Call Pickup Group	<input type="text" value=" < None >"/>		

Park Monitoring

	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value=" < None >"/> A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value=" < 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 And Confidential Access Level Settings

Target (Destination)	<input type="text"/>
MLPP Calling Search Space	<input type="text" value=" < None >"/>
MLPP No Answer Ring Duration (seconds)	<input type="text"/>
Confidential Access Mode	<input type="text" value=" < None >"/>
Confidential Access Level	<input type="text" value=" < None >"/>
Call Control Agent Profile	<input type="text" value=" < None >"/>

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*	<input type="text" value=" Default"/>	

Figure 56: SCCP Phone Configuration-9

Line 1 on Device SEP0014A998C3FC

Display (Caller ID)

Skinny Phone

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

ASCII Display (Caller ID)

Skinny Phone

Line Text Label

Skinny Phone

External Phone Number Mask

Visual Message Waiting Indicator Policy*

Use System Policy

Ring Setting (Phone Idle)*

Use System Default

Ring Setting (Phone Active)

Use System Default

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

Call Pickup Group Audio Alert Setting (Phone Idle)

Use System Default

Call Pickup Group Audio Alert Setting (Phone Active)

Use System Default

Monitoring Calling Search Space

< None >

Multiple Call/Call Waiting Settings on Device SEP0014A998C3FC

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 SEP0014A998C3FC

☒ Caller Name

☐ Caller Number

☐ Redirected Number

☒ Dialed Number

Users Associated with Line

Associate End Users

Save

Delete

Reset

Apply Config

Add New


Figure 57: SCCP Phone Configuration-10

MGCP Gateway Configuration

Navigation: Device -> Gateway

1. Set **Gateway Type**: Select the gateway model you are using, from the available list of devices.
2. Set **Domain Name**: Enter the FQDN of the gateway device.
3. Set **Cisco Unified Communications Manager Group**: Default
4. Set **Module in Slot 0**: Select 'NM-4VWIC-MBRD'
Upon saving the configuration, 'Subunit' configuration fields will appear
 - a. Set **Subunit 0**: Select the model of T1 card (hardware), which is on your gateway.
5. Set Global ISDN Switch Type: primary-qsig

Add a new GatewayRelated Links: [Back To Find/List](#) [Go](#)

 [Next](#)

Select the type of gateway you would like to add:

Gateway Type Cisco 3925

Change Gateway type

Protocol* MGCP

[Next](#)

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

Save
 Delete
 Reset
 Apply Config
 Add New

Status
 Status: Ready

Gateway Details

Product	Cisco 3925
Gateway	cisco-lync-gw.lab.tekvizion.com
Protocol	MGCP

Device is not trusted

Domain Name*

Description

Cisco Unified Communications Manager Group* ▼

Configured Slots, VICs and Endpoints

Module in Slot 0

Subunit 0

Subunit 1

Subunit 2

Subunit 3

Module in Slot 1

Module in Slot 2

0/0/ 0
0/0/ 1
0/0/ 2
0/0/ 3

Product Specific Configuration Layout

Global ISDN Switch Type

Switchback Timing*

Switchback uptime-delay (min)

Switchback schedule (hh:mm)

Type Of DTMF Relay*

Modem Passthrough*

Cisco Fax Relay*

T38 Fax Relay*

RTP Package Capability*

MT Package Capability*

RES Package Capability*

PRE Package Capability*

SST Package Capability*

RTP Unreachable OnOff*

RTP Unreachable timeout (ms)*

RTCP Report Interval (secs)*

Simple SDP*

Save
Delete
Reset
Apply Config
Add New

Figure 58: Trunk to Gateway-1

After saving the configuration page, beside the subunit (subunit 0 here) you have configured, click on the interface to which you have plugged the cable. A new configuration page appears, proceed with the configuration steps described below.

6. Set **Device Pool**: Default
7. Confirm **send Extra Leading Character in Display IE*****: is unchecked

Gateway Configuration
Related Links: [Back to MGCP Configuration](#) ▼ G

Save
 Delete
 Reset
 Apply Config
 Add New

Status
 Status: Ready

Device Information

Product	Cisco MGCP T1 Port
Gateway	cisco-lync-gw.lab.tekvizion.com
Device Protocol	Digital Access PRI
Registration:	Registered with Cisco Unified Communications Manager clus26sub1
IPv4 Address:	10.64.3.69
Device is not trusted	
End-Point Name *	S0/SU0/DS1-0@cisco-lync-gw.lab.tekvizion.com
Description	S0/SU0/DS1-0@cisco-lync-gw.lab.tekvizion.com
Device Pool*	Default ▼
Common Device Configuration	< None > ▼
Call Classification*	Use System Default ▼
NetworkLocale	< None > ▼
Packet Capture Mode*	None ▼
Packet Capture Duration	0
Media Resource Group List	< None > ▼
Location*	Hub_None ▼
AAR Group	< None > ▼
Load Information	
Use Trusted Relay Point*	Default ▼
Route Class Signaling Enabled*	Off ▼
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> V150 (subset)	
<input checked="" type="checkbox"/> PSTN Access	

MLPP and Confidential Access Level Information

MLPP Domain	< None > ▼
MLPP Indication	Default ▼
MLPP Preemption	Default ▼
Confidential Access Mode	< None > ▼
Confidential Access Level	< None > ▼

Figure 59: Trunk to Gateway-2

Interface Information	
PRI Protocol Type*	PRI NI2
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Protocol Side*	User
Channel Selection Order*	Bottom Up
Channel IE Type*	Use Number when 1B
PCM Type*	μ-law
Delay for first restart (1/8 sec ticks)*	32
Delay between restarts (1/8 sec ticks)*	4
<input checked="" type="checkbox"/> Inhibit restarts at PRI initialization <input type="checkbox"/> Enable status poll <input type="checkbox"/> Unattended Port <input type="checkbox"/> Enable G.Clear	

Call Routing Information - Inbound Calls	
Significant Digits*	All
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	

Call Routing Information - Outbound Calls	
Calling Party Presentation*	Default
Calling Party Selection*	Originator
Called party IE number type unknown*	Cisco CallManager
Calling party IE number type unknown*	Cisco CallManager
Called Numbering Plan*	Cisco CallManager
Calling Numbering Plan*	Cisco CallManager
Number of digits to strip*	0
Caller ID DN	
SMDI Base Port*	0
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	

Figure 60: Trunk to Gateway-3

PRI Protocol Type Specific Information

☐ Display IE Delivery

☒ Redirecting Number IE Delivery - Outbound
 Redirecting Party Transformation CSS < None >

☐ Use Device Pool Redirecting Party Transformation CSS
 ☐ Redirecting Number IE Delivery - Inbound
 ☐ Send Extra Leading Character in Display IE***
 ☐ Setup non-ISDN Progress Indicator IE Enable****
 ☐ MCDN Channel Number Extension Bit Set to Zero**
 ☐ Send Calling Name In Facility IE
 ☐ Interface Identifier Present**
 Interface Identifier Value** 0
 Connected Line ID Presentation (QSIG Inbound Call)* Default

Connected Party Settings

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

UUIE Configuration

☐ Passing Precedence Level Through UUIE
 Security Access Level* 2

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

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
National Number	Default	0	< None >	<input checked="" type="checkbox"/>
International Number	Default	0	< None >	<input checked="" type="checkbox"/>
Unknown Number	Default	0	< None >	<input checked="" type="checkbox"/>
Subscriber Number	Default	0	< None >	<input checked="" type="checkbox"/>

Figure 61: Trunk to Gateway-4

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
National Number	Default	0	< None >	<input checked="" type="checkbox"/>
International Number	Default	0	< None >	<input checked="" type="checkbox"/>
Unknown Number	Default	0	< None >	<input checked="" type="checkbox"/>
Subscriber Number	Default	0	< None >	<input checked="" type="checkbox"/>

Incoming Called Party Settings

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

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
National Number	Default	0	< None >	<input checked="" type="checkbox"/>
International Number	Default	0	< None >	<input checked="" type="checkbox"/>
Unknown Number	Default	0	< None >	<input checked="" type="checkbox"/>
Subscriber Number	Default	0	< None >	<input checked="" type="checkbox"/>

Product Specific Configuration Layout



Line Coding *	B8ZS
Framing *	ESF
Clock *	External
Input Gain (-6..14 db) *	0
Output Attenuation (-6..14 db) *	0
Echo Cancellation Enable *	Enable
Echo Cancellation Coverage (ms) *	128

Geolocation Configuration

Geolocation	< None >
Geolocation Filter	< None >

Save

Delete

Reset

Apply Config

Add New




Figure 62: Trunk to Gateway-5

Voice Mail Pilot Configuration


Navigation: Advanced Features->Voice Mail->Voice Mail Profile

Voice Mail Pilot Configuration

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

 Save  Delete  Add New

Status

 Status: Ready

Voice Mail Pilot Information

Voice Mail Pilot Number1000

Calling Search Space< None > ▼

Description

☒ Make this the default Voice Mail Pilot for the system

Save

Delete

Add New

Figure 63: Voice Mail Port Configuration

Cisco Unity Voice Mail Server Configuration

Phone System Configuration

Navigation: Telephony Integrations -> Phone System

1. Set Phone System Name: Enter a name for the phone system
2. To navigate to the next step of configuration, move to related links 'Add Port Group' and click Go.

Phone System Basics (Cluster26)

Search Phone Systems ▶ Phone System Basics (Cluster26)

Related Links Add Port Group ▼ Go

Phone System Edit Refresh Help

Save Delete Previous Next

Phone System

Phone System Name* Cluster26

☐ Default TRAP Phone System

Message Waiting Indicators

☐ Send Message Counts

☐ Use Same Port for Enabling and Disabling MWIs

☐ Force All MWIs Off for this Phone System

Run Synchronize All MWIs on This Phone System

Call Loop Detection by Using DTMF

☐ Enable for Supervised Transfers

☐ Enable for Forwarded Message Notification Calls (by Using DTMF)

DTMF Tone To Use A ▼

Guard Time 2500 milliseconds

Call Loop Detection by Using Extension

☒ Enable for Forwarded Message Notification Calls (by Using Extension)

Phone View Settings

☐ Enable Phone View

CTI Phone Access Username

CTI Phone Access Password

Outgoing Call Restrictions

☒ Enable outgoing calls

☐ Disable all outgoing calls immediately

☐ Disable all outgoing calls between

Beginning Time: 12 ▼ 00 ▼ AM ▼

Ending Time: 12 ▼ 00 ▼ AM ▼

Save Delete Previous Next

Fields marked with an asterisk (*) are required.

Figure 64: Unity Phone System Configuration

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Port Group Configuration

Navigation: Telephony Integrations -> Port Group

1. Set Phone System: Select the phone system configured in the above section
2. Set Create From Port Group Type: SIP
3. Set SIP Security Profile: 5060
4. Set SIP Transport Protocol: TCP

Search Port Groups ▶ New Port Group

New Port Group

Related Links [Check Telephony Configuration](#) [Go](#)

Port Group Reset Help

Save

New Port Group

Phone System Cluster26

Create From ☒ Port Group Type SIP

☐ Port Group Cluster26-1

Port Group Description

Display Name* Cluster26-1

☐ Authenticate with SIP Server

Authentication Username

Authentication Password

Contact Line Name

SIP Security Profile 5060

SIP Transport Protocol TCP

Primary Server Settings

IPv4 Address or Host Name

IPv6 Address or Host Name

Port 5060

Save

Fields marked with an asterisk (*) are required.

Figure 65: Port Group Configuration

Port Configuration

Navigation: Telephony Integrations -> Port

1. Set Port Name: Enter Name for the port
2. Set Phone System: Select the phone system which you are creating the ports to
3. Set Port Group: Select the port group configured to the phone system selected in the above step
4. Set Server: Select the unity server

Search Ports ▶ Port Basics (Cluster26-1-001)

Port Basics (Cluster26-1-001)Related LinksCheck Telephony Configuration ▼Go

Port Refresh Help

SaveDeletePreviousNext

Phone System Port

☒ Enabled

Port NameCluster26-1-001Restart

Phone SystemCluster26

Port GroupCluster26-1

Serverclus26unity.lab.tekvizion.com ▼

Port Behavior

☒ Answer Calls

☒ Perform Message Notification

☒ Send MWI Requests (may also be disabled by the port group)

☒ Allow TRAP Connections

SaveDeletePreviousNext

Figure 66: Port Configuration

Cisco Unity User Configuration

Navigation: Users -> Users

1. Set Alias: Enter a name for the user
2. Set Extension: Enter the extension you are configuring this user for
3. Set Phone System: Select the phone system you configured under *Telephony Integrations*->
Phone System

Cisco User Configuration

Edit User Basics (Cisco 2000)

Search Users ▶ Edit User Basics (Cisco 2000)

Related Links Bulk Edit By CSV ▼ Go

User Edit Refresh Help

Save Delete Previous Next

Name

Alias* Cisco 2000

First Name Cisco

Last Name 2000

Display Name Cisco 2000

SMTP Address cisco_2000 @clus26unity.lab.tekvizion.com

Initials

Title

Employee ID

LDAP Integration Status

☐ Integrate with LDAP Directory

☒ Do Not Integrate with LDAP Directory

Phone

Extension* 2000

Cross-Server Transfer Extension

Outgoing Fax Number

Outgoing Fax Server --- Not Selected --- ▼

Partition clus26unity Partition ▼

Search Scope clus26unity Search Space ▼

Phone System Cluster26 ▼

Class of Service Voice Mail User COS ▼

Active Schedule Weekdays ▼ View

☐ Set for Self-enrollment at Next Sign-In

☒ List in Directory

☒ Send Non-Delivery Receipts on Failed Message Delivery

☒ Skip PIN When Calling From a Known Extension

Caution! Security risk. See Help for This Page for details.

☐ Use Short Calendar Caching Poll Interval

Recorded Name Play/Record

Figure 67: Unity Cisco User Configuration-1

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Location

Address

Building

City

State

Postal Code

Country

☒ Use System Default Time Zone

Time Zone

Language ☒ Use System Default Language

☐

Department

Manager

Billing ID

Corporate Email Address

☐ Generate SMTP Proxy Address From Corporate Email Address

Corporate Phone Number

Figure 68: Unity Cisco User Configuration-2

Lync User Configuration

Search Users ▶ Edit User Basics (Lync User 2619)

Related Links Bulk Edit By CSV ▼ Go

User Edit Refresh Help

Save Delete Previous Next

Name

Alias* Lync User 2619

First Name Lync User

Last Name 2619

Display Name Lync User 2619

SMTP Address lync_user_2619 @clus26unity.lab.tekvizion.com

Initials

Title

Employee ID

LDAP Integration Status

☐ Integrate with LDAP Directory

☒ Do Not Integrate with LDAP Directory

Phone

Extension* +19728522619

Cross-Server Transfer Extension +19728522619

Outgoing Fax Number

Outgoing Fax Server --- Not Selected --- ▼

Partition clus26unity Partition ▼

Search Scope clus26unity Search Space ▼

Phone System Cluster26 ▼

Class of Service Voice Mail User COS ▼

Active Schedule Weekdays ▼ View

☐ Set for Self-enrollment at Next Sign-In

☒ List in Directory

☒ Send Non-Delivery Receipts on Failed Message Delivery

☒ Skip PIN When Calling From a Known Extension

Caution! Security risk. See Help for This Page for details.

☐ Use Short Calendar Caching Poll Interval

Recorded Name Play/Record

Figure 69: Unity Lync User Configuration-1

Location

Address

Building

City

State

Postal Code

Country ▼

☒ Use System Default Time Zone

Time Zone ▼

Language ☒ Use System Default Language

☐ ▼

Department

Manager

Billing ID

Corporate Email Address

☐ Generate SMTP Proxy Address From Corporate Email Address

Corporate Phone Number

Figure 70: Unity Lync User Configuration-2

Gateway Configuration

```
version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname cisco-lync-gw
!
boot-start-marker
boot system flash c3900e-universalk9-mz.SPA.154-1.T.bin
boot-end-marker
!
aqm-register-fnf
!
card type t1 0 0
logging buffered 999999999
no logging rate-limit
no logging console
enable secret 4 sKpgCY/XPea3wk8xoeSWo7UGFaNVwzXDEyXWhuDJeLk
!
no aaa new-model
!
network-clock-participate wic 0
network-clock-select 1 T1 0/0/0
!
ip domain name lab.tekvizion.com
ip name-server <name-server IP address>
ip cef
no ipv6 cef
!
multilink bundle-name authenticated
!
isdn switch-type primary-qsig
!
voice-card 0
dsp services dspfarm
!
voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
sip
session transport tcp
header-passing
asserted-id pai
!
```



```

voice class uri 1 sip
  host cisco-lync-gw.lab.tekvizion.com
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g711alaw
!
controller T1 0/0/0
  cablelength long 0db
  pri-group timeslots 1-24 service mgcp
!
controller T1 0/0/1
  cablelength long 0db
!
controller T1 0/0/2
  cablelength long 0db
!
controller T1 0/0/3
  cablelength long 0db
!
interface GigabitEthernet0/0
  description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
  ip address 10.64.3.69 255.255.0.0
  duplex auto
  speed auto
!
interface GigabitEthernet0/1
  no ip address
  shutdown
  duplex auto
  speed auto
!
interface GigabitEthernet0/2
  no ip address
  shutdown
  duplex auto
  speed auto
!
interface GigabitEthernet0/3
  no ip address
  shutdown
  duplex auto
  speed auto
!
interface Serial0/0/0:23
  no ip address
  encapsulation hdlc
  isdn switch-type primary-qsig

```

```

isdn timer T310 120000
isdn protocol-emulate network
isdn incoming-voice voice
isdn bind-l3 ccm-manager
no cdp enable
!
ip default-gateway <default-gateway IP Address>
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 10.64.1.1
!
nls resp-timeout 1
cpd cr-id 1
!
control-plane
!
voice-port 0/0/0:23
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 0/1/2
!
voice-port 0/1/3
!
mgcp
mgcp call-agent clus26sub1 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode nte-ca
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp voice-quality-stats
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
mgcp package-capability fm-package
no mgcp package-capability res-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 ecm

```

```
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
mgcp bind control source-interface GigabitEthernet0/0
mgcp bind media source-interface GigabitEthernet0/0
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
mgcp rtp payload-type nte 101
!
mgcp profile default
!
ccm-manager music-on-hold
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager config server clus26sub1
ccm-manager config
!
sip-ua
set pstn-cause 31 sip-status 480
timers expires 1800000
!
gatekeeper
shutdown
!
line con 0
login local
line aux 0
line vty 0 4
access-class 23 in
exec-timeout 0 0
privilege level 15
login local
transport input telnet ssh
line vty 5 15
access-class 23 in
exec-timeout 0 0
privilege level 15
login local
transport input telnet ssh
!
scheduler allocate 20000 1000
!
end
```

Cisco Router[External IOS MTP] Configuration

```
sccp local GigabitEthernet0/1
sccp ccm 10.80.16.3 identifier 1 priority 1 version 7.0
sccp
!
sccp ccm group 40
description EXT MTP
bind interface GigabitEthernet0/1
associate ccm 1 priority 1
associate profile 40 register EXTMTP
!
dspfarm profile 40 mtp
codec g711ulaw
maximum sessions software 20
associate application SCCP
!
```

Lync Server Configuration

Add CISCO UCM to Lync Topology

Lync recognizes CISCO UCM as a PSTN gateway connected by SIP trunk. So we need to add CISCO UCM to the Lync topology by adding it as a PSTN gateway.

1. To add a PSTN gateway to the Lync topology, run Lync Server Topology Builder as a user in the CSAdministrator group. Then add the CISCO UCM to the PSTN gateway topology

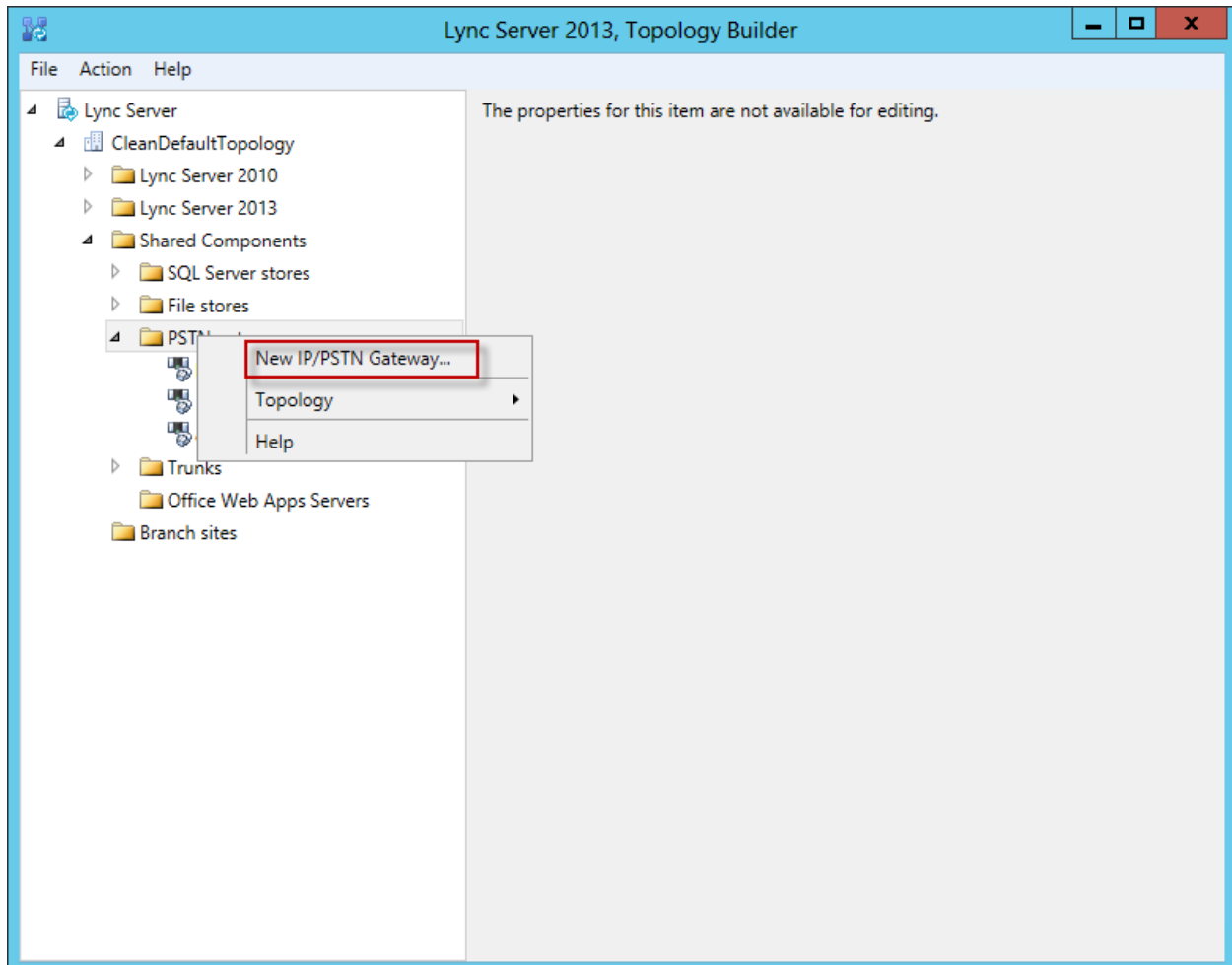
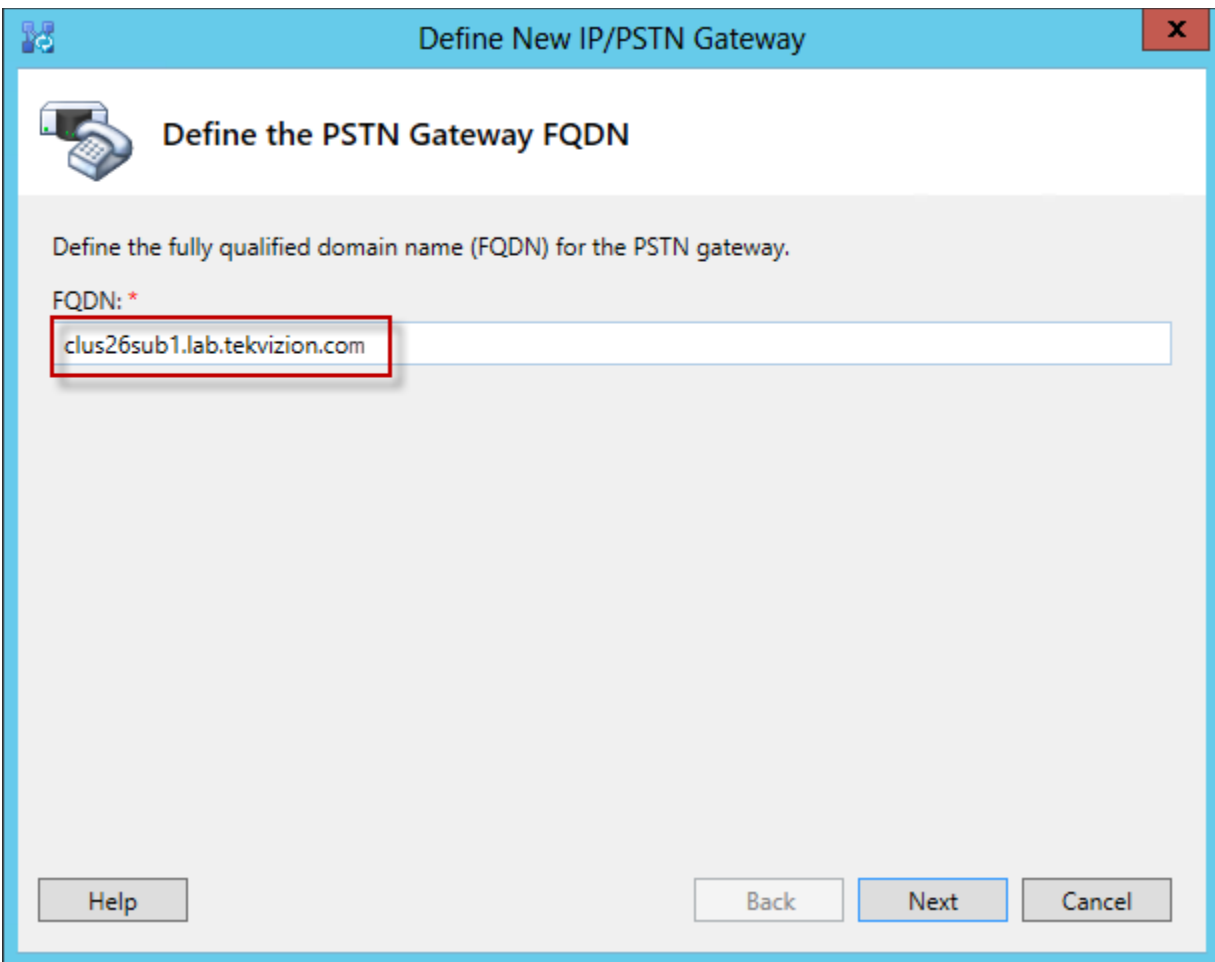



Figure 71: Configure PSTN Gateway -1


2. Set **FQDN**: This is the IP Address or FQDN of the CISCO UCM.



The screenshot shows a Windows-style dialog box titled "Define New IP/PSTN Gateway" with a close button (X) in the top right corner. Inside the dialog, there is a sub-header "Define the PSTN Gateway FQDN" accompanied by a telephone icon. Below this, a text label reads "Define the fully qualified domain name (FQDN) for the PSTN gateway." followed by "FQDN: *". A text input field contains the value "clus26sub1.lab.tekvizion.com", which is highlighted with a red rectangular box. At the bottom of the dialog, there are four buttons: "Help", "Back", "Next", and "Cancel". The "Next" button is highlighted with a blue border.

Figure 72: Configure PSTN Gateway -2

 Define New IP/PSTN Gateway ✕

 Define the IP address

☒ Enable IPv4

☒ Use all configured IP addresses.

☐ Limit service usage to selected IP addresses.

PSTN IP address:

☐ Enable IPv6

☒ Use all configured IP addresses.

☐ Limit service usage to selected IP addresses.

PSTN IP address:

Help

Back

Next

Cancel

3. Set **Trunk Name**: This is the FQDN of the CISCO UCM
4. Set **Listening port for IP/PSTN gateway**: This **Listening port** should match the **Incoming Port** setting in the CISCO UCM's **SIP Trunk Security Profile**.
5. Set **SIP Transport Protocol**: TCP
6. Set **Associate Mediation Server**: Assign this PSTN gateway to the Mediation Server. Medpool.lyncclabkm2013.local is used here for example.

Define New IP/PSTN Gateway

Define the root trunk

Trunk name: *

clus26sub1.lab.tekvizion.com

Listening port for IP/PSTN gateway: *

5060

SIP Transport Protocol:

TCP

Associated Mediation Server:

medpool.lyncclabkm2013.local CleanDefaultTopology

Associated Mediation Server port: *

5060

Help Back Finish Cancel

Figure 73: Configure PSTN Gateway -3

7. Publish topology to make the changes effective, refer to below screen capture for the process.

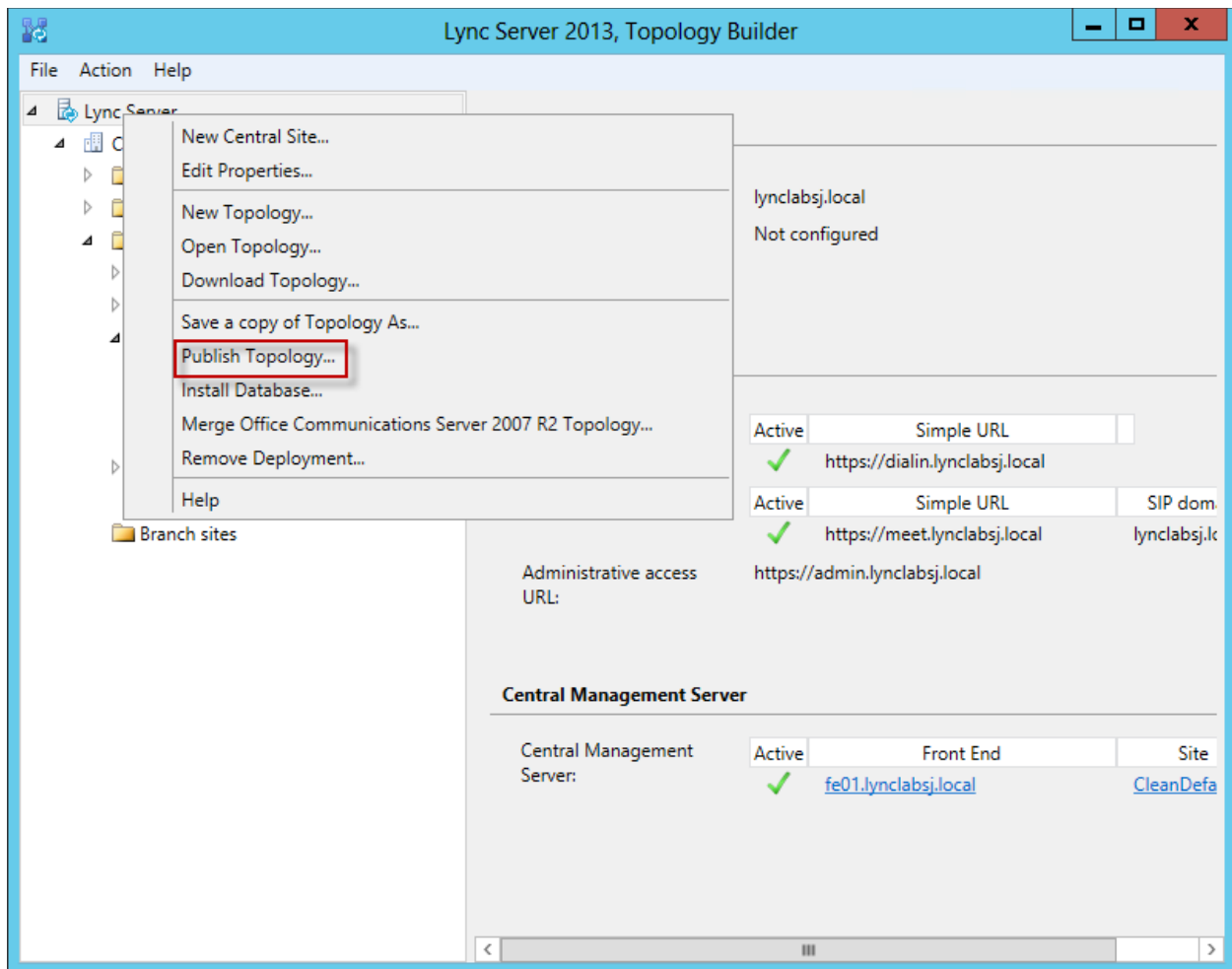


Figure 74: Publish Topology

Trunk Configuration

Navigation: Voice Routing -> Trunk Configuration

1. Create a **Pool Trunk** by selecting New
2. Select **Service**: Select the trunk to CISCO UCM you created in topology builder
3. Set **Maximum early dialogs supported**: 20
4. Set **Encryption support level**: Optional
5. Set **Refer Support**: Enable sending refer to the gateway
6. Confirm **Enable media bypass**: is checked
7. Confirm **Centralized media processing**: is checked
8. Confirm **Enable RTP latching**: is unchecked
9. Confirm **Enable forward call history**: is unchecked
10. Confirm **Enable forward P-Asserted-Identity data**: is unchecked
11. Confirm **Enable outbound routing failover timer**: is checked

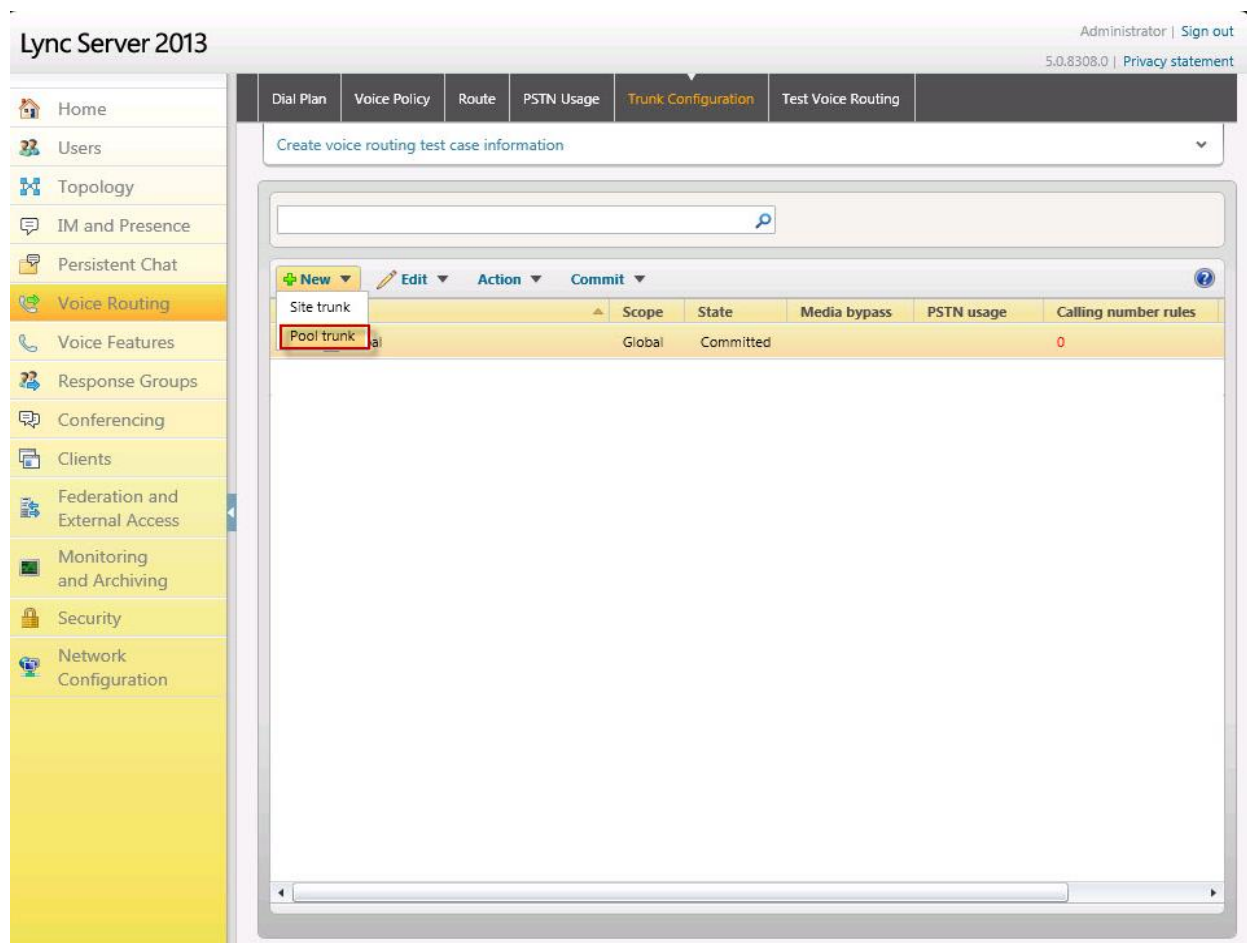


Figure 75: Trunk Configuration -1

Edit Trunk Configuration - PstnGateway:clus26sub1.lab.tekvizion....

✓ OK ✗ Cancel ?

Scope: Pool

Name: *

PstnGateway:clus26sub1.lab.tekvizion.com

Description:

Maximum early dialogs supported:

20

Encryption support level:

Optional

Refer support:

Enable sending refer to the gateway

☒ Enable media bypass

☒ Centralized media processing

☐ Enable RTP latching

Figure 76: Trunk Configuration -2

Edit Trunk Configuration - PstnGateway:clus26sub1.lab.tekvizion....

✓ OK ✗ Cancel ?

☐ Enable forward call history

☐ Enable forward P-Asserted-Identity data

☒ Enable outbound routing failover timer

^ Associated PSTN Usages

Select... Remove ↑ ↓

PSTN usage record	Associated routes

Translated number to test:

Go

Figure 77: Trunk Configuration -3

Edit Trunk Configuration - PstnGateway:clus26sub1.lab.tekvizion...

OK

Cancel

?

^ Associated translation rules

Calling number translation rules

New

Copy

Paste

Select...

Show details...

Remove

↑

↓

Translation rule	State	Pattern to match	Translation pattern

Called number translation rules

New

Copy

Paste

Select...

Show details...

Remove

↑

↓

Translation rule	State	Pattern to match	Translation pattern

Phone number to test:

Go

?

☒ Calling number
☐ Called number

Figure 78: Trunk Configuration -4

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Route

Navigation: Voice Routing -> Route

1. Set **Name**: Enter a name for this route
2. Add **Associated gateways**: Add the gateway (CISCO UCM here) to which this route should send all the calls.

Edit Voice Route - CUCM10_Route

✓ OK ✗ Cancel

Scope:

Name: *
CUCM10_Route

Description:

Build a Pattern to Match

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

Starting digits for numbers that you want to allow:

Type a valid number and then click Add.

Add

Exceptions

Remove

Match this pattern: *
.*

Edit Reset ?

☐ Suppress caller ID

Alternate caller ID:

Associated trunks:

PstnGateway:clus26sub1.lab.tekv...

Add...

Remove

Associated PSTN Usages

Select... Remove ↑ ↓

PSTN usage record	Associated voice policies
CUCM10_PSTN_Usage	CUCM10_Policy

Translated number to test:

Go

Figure 79: Route Configuration-1

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Voice Policy and PSTN Usage

Navigation: Voice Routing -> Voice Policy

1. Create a **User policy** by selecting New
2. Set **Name**: Enter a name for this Voice Policy
3. Set **Calling Features**:
 - a. Enable call forwarding : Checked
 - b. Enable delegation : Checked
 - c. Enable call transfer : Checked
 - d. Enable call park : Checked
 - e. Enable simultaneous ringing of phones : Checked
 - f. Enable team call : Checked
 - g. Enable PSTN reroute : Checked
 - h. Enable bandwidth policy override : Unchecked
 - i. Enable malicious call tracing : Unchecked
4. Set **Associated PSTN Usages**:
 - a. Select New to create a new PSTN Usage
 - b. Set **Name**: Enter a name for this PSTN Usage
 - c. Set **Associated Routes**: Select the route you created under Voice Routing -> Route

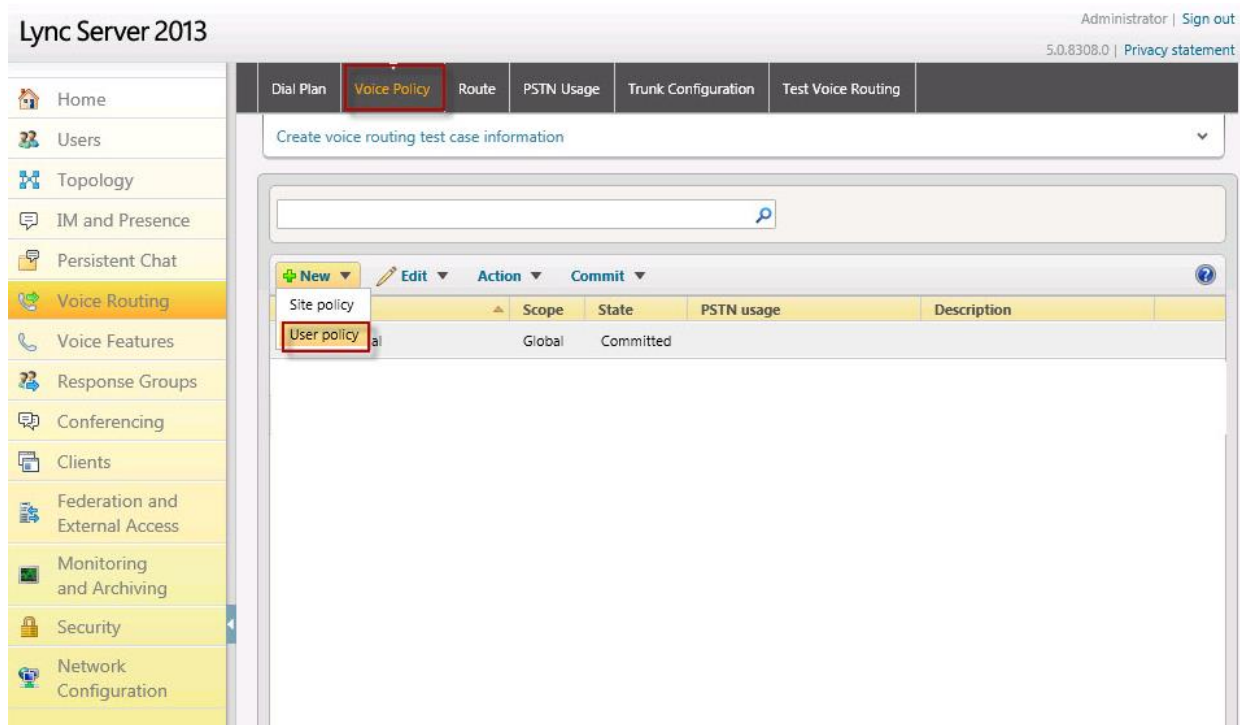


Figure 80: Voice Policy -1

Edit Voice Policy - CUCM10_Policy

OK

Cancel

?

Scope: User
Name: *

CUCM10_Policy

Description:

^ Calling Features

☒ Enable call forwarding
☒ Enable team call

☒ Enable delegation
☒ Enable PSTN reroute

☒ Enable call transfer
☐ Enable bandwidth policy override

☒ Enable call park
☐ Enable malicious call tracing

☒ Enable simultaneous ringing of phones

Associated PSTN Usages

New

Select...

Show details...

Remove

↑

↓

PSTN usage record	Associated routes
CUCM10_PSTN_Usage	CUCM10_Route

Call forwarding and simultaneous ringing PSTN usages:

Route using the call PSTN usages

Translated number to test:

Go

Figure 81: Voice Policy -2

View PSTN Usage Record - CUCM10_PSTN_Usage

Close

Name:

CUCM10_PSTN_Usage

Associated Routes

Route	Pattern to match
CUCM10_Route	.*

Associated Voice Policies


Voice policy	Description
 CUCM10_Policy	

Figure 82: PSTN Usage

Dial Plan

Navigation: Voice Routing-> Dial Plan

Create a dial plan with normalization rules for all the enterprise and local voice calls

1. Under Dial plan tab, select New (site/pool/user) dial plan or modify the existing Global dial plan
2. Configure the Normalization rules for 10-digit, 4-digit and 3-digit dialing as shown in the below screen shots.

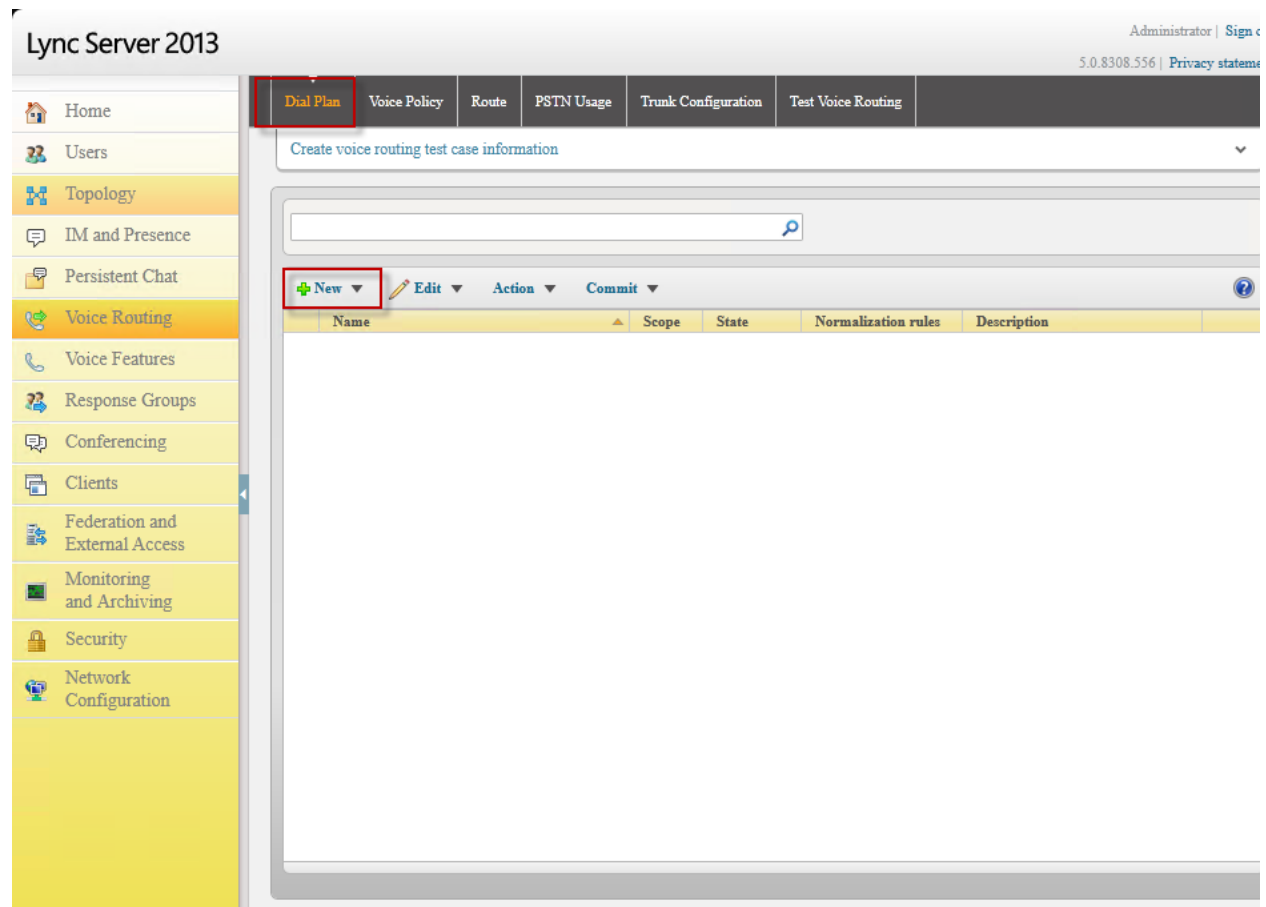


Figure 83: Dial Plan-1

Edit Dial Plan - Global

OK

Cancel

?

Scope: Global
Name: *
Simple name: *
Description:
Dial-in conferencing region:
External access prefix:

Associated Normalization Rules

New

Copy

Paste

Select...

Show details...

Remove

↑

↓

Normalization rule	State	Pattern to match	Translation pattern
Three digit	Committed	<code>^(d{3})\$</code>	\$1
Four Digit	Committed	<code>^(d{3}d+)\$</code>	\$1
Keep All	Committed	<code>^(d{9}d+)\$</code>	+1\$1
<input type="text"/>			

Dialed number to test:

Go
?

Figure 84: Dial Plan-2

Edit Dial Plan ▸ Edit Normalization Rule - Keep All

✓ OK ✗ Cancel ?

Name: *
Keep All

Description:

Build a Normalization Rule

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

Starting digits:

Length:
At least 10

Digits to remove:
0

Digits to add:
+1

Pattern to match: *
`^(\d{9}\d+)$`

Translation rule: *
`+1$1`

Edit Reset ?

☐ Internal extension ?

Dialed number to test:

Go

Figure 85: Normalization Rule for 10-digit dialing

Edit Dial Plan ▸ Edit Normalization Rule - Three digit

✓ OK ✗ Cancel ?

Name: *
Three digit

Description:

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 3

Digits to remove:
0

Digits to add:

Pattern to match: *
^(d{3})\$

Translation rule: *
\$1

Edit Reset ?

☐ Internal extension ?

Dialed number to test:
Go

Figure 86: Normalization Rule for 3-digit dialing

Edit Dial Plan ▶ Edit Normalization Rule - Four Digit

✓ OK ✗ Cancel ?

Name: *
Four Digit

Description:

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 ▼ 4

Digits to remove:
0

Digits to add:

Pattern to match: *
^(d{4})\$

Translation rule: *
\$1

Edit Reset ?

☐ Internal extension ?

Dialed number to test:
Go

Figure 87: Normalization Rule for 4-digit dialing

Configure Media Bypass

Navigation: Network Configuration -> Global

1. Enable '**Enable media bypass**' in Global setting.
2. Confirm you have also selected **Always bypass**

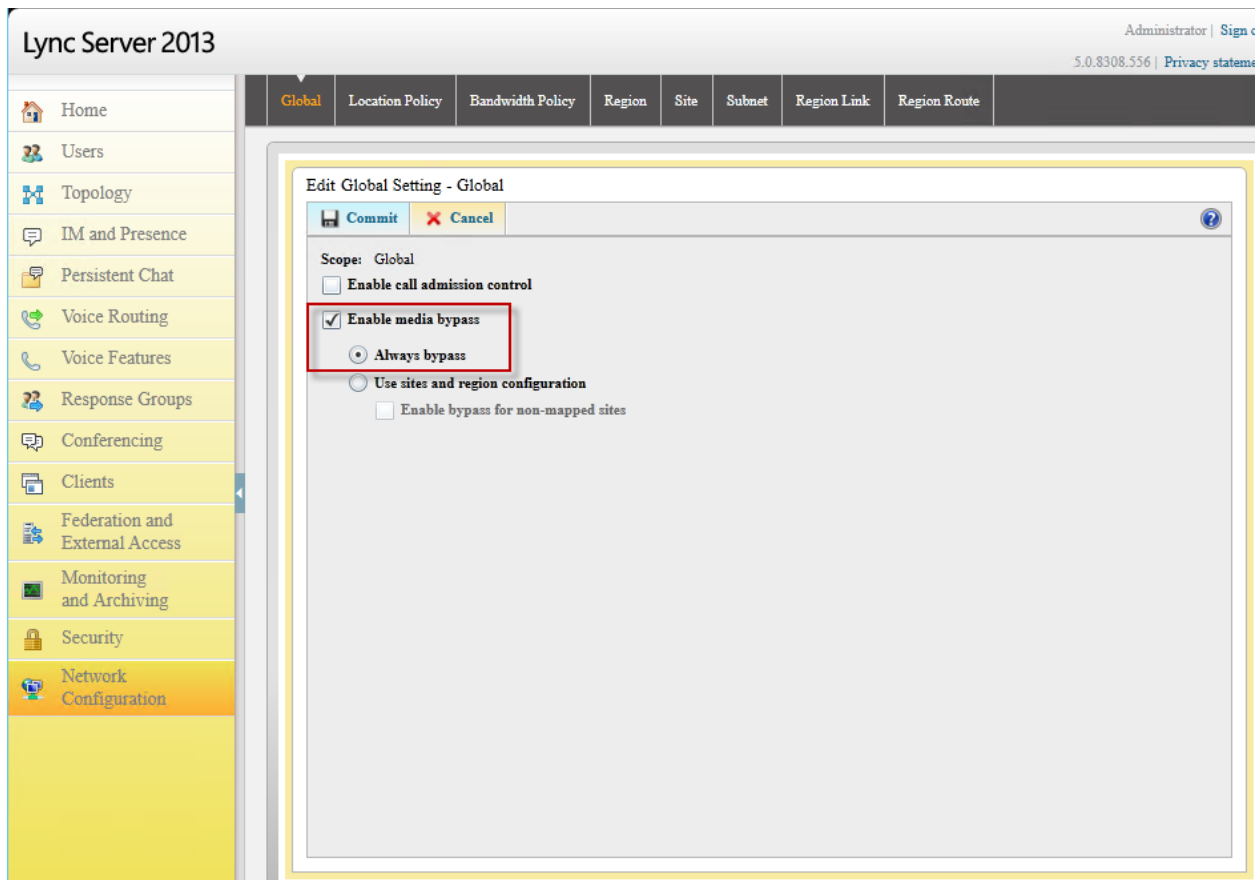


Figure 88: Media Bypass

Configure Encryption Level

Configure Encryption level parameters through the Windows PowerShell® command line interface because they are not configurable on Lync Server Control Panel

Media EncryptionLevel must be set to SupportEncryption. Since we do not support SRTP to Cisco through Direct SIP, we need to set the media configuration's EncryptionLevel to SupportEncryption so that SRTP will only be used if it can be negotiated. By default, this parameter is set to RequireEncryption, meaning SRTP must be used.

Set-CsMediaConfiguration –identity Global -EncryptionLevel SupportEncryption

```
PS C:\Users\administrator.LYNCLABKM2013> Get-CsMediaConfiguration

Identity                : Global
EnableQoS                : False
EncryptionLevel          : SupportEncryption
EnableSiren              : False
MaxVideoRateAllowed      : UGA600K
EnableG722StereoCodec    : True
EnableH264Codec          : True
EnableAdaptiveBandwidthEstimation : True
```