Microsoft Lync Server 2010 Standard Edition using Direct SIP to Cisco Unified Communications Manager 8.5(1)
Nov 3rd, 2011 – Initial Version

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Introduction

- Microsoft Lync embeds the functions of instant messaging, voice and video on a software platform integrating tightly with the Microsoft Active Directory (AD) and other Microsoft products. This document explains integration steps, configurations, feature support, and limitations for direct SIP interoperability between Cisco Unified Communications Manager (Cisco UCM) Release 8.5(1) and Microsoft Lync 2010 Server Standard Edition.

- Testing has been done according to Microsoft’s Open Interoperability Program (OIP) test plan, which includes testing with Microsoft’s Open Interoperability Test Tool (LyncIT) and end-to-end manual testing for the Microsoft Lync 2010 OIP IP-PBX program. This test plan focuses on interaction between the Mediation Server and the Partner-supported system with 3 possibilities of tests, which are using LyncIT, PSTN supported devices and manual testing using Lync Server 2010.

- Cisco UCM 8.5.1 supports Early Offer (EO) without the use of MTP resources and as described in RFC 3261 (Offer SDP in INVITE (early offer), Answer SDP in 200 OK (possibly in 18x as well)). Traditionally, without the MTP resource assigned, Cisco UCM used Delayed Offer (DO) per RFC 3261 (No SDP in INVITE (delayed offer), Offer SDP in 200 OK and Answer SDP in ACK). The document explains the configuration steps to use EO feature on Cisco UCM 8.5(1). The two primary limitations of using traditional EO with MTP are that Cisco UCM or DSP-based MTP resources need to be used and the number of codec in the offer is restricted to one (G711 or G729).

- The Microsoft test plan covers scenarios to test Gateways/PBX’s or IP-PBX’s, per Microsoft’s Direct SIP specifications. This application note aims to provide a good understanding of what works and what does not work in terms of the interaction between a Cisco UCM device and Microsoft Lync. It also provides guidance to deployment participants regarding the limitations, expected behaviors, and known issues. The document is not meant to address any performance and scalability, which are part of broader criteria for a deployment-ready solution (for more details refer to 8.x SRND)


- Endpoints registered to Lync and Cisco UCM use E.164 dial plans and extensions. Cisco UCM 8.x and later support E164 extensions and E164 dialing.

- An alternative to Direct SIP, which provides only basic SIP trunk interoperability and requires dual call control systems, is Cisco UC Integration for Microsoft Lync. This tight integration with Cisco UC for Microsoft Lync offers the benefits of investment protection and reduced complexity delivered by a single, proven call control solution. It takes advantage of a common unified client services framework, providing the following benefits:

  1. Increased productivity, to instantly connect with colleagues, partners, and customers from anywhere and have a wide-band audio and high-definition video communications experience with the integrated Cisco IP soft phone
  2. Streamlined communications, with telephony presence, visual voicemail, communications history, and Cisco Unified IP Phone control from the Windows desktop
  3. Enhanced collaboration, allowing users to initiate or escalate into integrated voice, video, and web Cisco Unified MeetingPlace and Cisco WebEx sessions with multiple parties
  4. Reduced complexity, through an easy-to-deploy integration and a single-call control architecture
  5. Investment protection, providing an immediate business impact with Cisco Unified Communications, while protecting investments in existing desktop applications
Network Topology

Figure 1. Basic Call Setup
Limitations

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

- Microsoft Lync does not support sending or receiving calling and connected name in certain scenarios (like call forward and call-transfer). Cisco UCM sends its calling name and number to Microsoft Lync Mediation Server, but Lync only displays the calling number and no name is displayed.

- Lync 2010 cannot be configured for Call Forward Busy. As an option Lync allows the user to receive a notification of an incoming call during an active call and “redirect” the incoming call to a destination of the user’s choice.

- When Lync initiates a conference call, the conference call does not end until all parties hang up including the initiator. Additionally for the Lync users on the conference call, the session is not dropped and they can rejoin the conference, unless they close the window pop-up for the call, referred to as the toast.

- The Media Bypass capability introduced in Microsoft Lync 2010 requires all media through a SIP trunk integration to originate on a single IP address. For this Microsoft requirement in a Lync to Cisco UCM inter-working, media termination point is required on the SIP trunk for Media Bypass to work.

- Media Bypass is a mandatory requirement if customers want to collocate the Mediation Server role with the Front-End server role. **Important**: According to the Microsoft Lync Planning Tool collocation should only be used if the IP-PBX and/or the voice gateways support Media Bypass and Enterprise voice is not considered mission critical to the organization.

- Media Bypass works as long as an MTP is required on the SIP trunk. The need for an MTP is due to the following:
  1. Lync supports Media Bypass with early offer, but does not support re-invite without SDP
  2. Lync doesn’t support Media Bypass with delayed offer
  3. Because of #1, hold/resume, transfer, forward, etc won’t work without an MTP.

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

- On Lyne Mediation Server, RTCP is set to disabled, as RTCP is an end point specific configuration and not all Cisco phones support RTCP. Phones like 7970, 7962, 9971 have this support and this can be enabled under Phone Configuration – Product Specific Configuration Layout.
• Calling and connected number updates during call-forward and call-transfer scenarios are not fully supported due to SIP UPDATE messages not being interoperable between the systems. For example when a Lync user transfers a UCM call to another UCM user, the local UCM phone will still show connected to the first Lync user.

• Cisco SIP Phones do not send out Comfort Noise payload 13 in the m line of the SDP to indicate support for Comfort Noise as per RFC 3389 and as a result Microsoft doesn’t send out SID frames when call is muted on Microsoft side.

• On Cisco UCM the DND feature can be enabled with two options: call reject or ringer off. Ringer Off can flash or beep based on the DND Incoming Call Alert configuration on the IP Phone. However, Lync client will reject the call only when DND is enabled on the Lync Client.

• When multiple ptimes in SDP are sent from CUCM, Lync rejects the call as it does not support multiple ptimes in SDP. Issue is resolved with delayed offer or MTP checked with early offer.

• Basic simultaneous ring feature does not work when a Lync initiates a call as CUCM introduces an additional line “TIAS” when it forwards the call to another Lync. So a TIAS script on CUCM is used to remove this line in SDP.

• Microsoft Lync support for Encryption TLS-SRTP on SIP trunk to CUCM has not been tested.

• Microsoft Lync might not be able to handle port change for mid-call reINVITEs.

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

• Mediation server on Microsoft Lync supports G711ulaw and G711alaw only. If Cisco UCM needs to send out any other codec (example G729 to a SP SIP trunk, and then out to the PSTN ) then a transcoder needs to be used, e.g. with an ISR.

• REFER test cases work when CUCM uses only one SIP trunk assigned to a SIP Route Pattern

• CUCM phones have the ability to divert calls to voicemail, versus providing a way to decline a call.

• CUCM does not support using a FQDN in the connection field; this can, however, be achieved using a lua script on CUCM.

• CUCM does not support trying a different Lync cluster on the INVITE timeout during failover testing when
  (a) CUCM cannot find a Mediation server in the cluster that can accept connection and (b) CUCM’s configuration allows interfacing to multiple Lync clusters,
  In this case CUCM tries to find a server from the second cluster.

• When SIP Refer feature is enabled on Lync server, two issues are seen with CUCM 8.5.1.1.0000-26
  1. When UCM receives SIP REFER, it sends the INVITE back out to the other server which resets the TCP connection. This causes the SIP Stack to resend the INVITE as a new transaction with a new CSEQ number, and new branch.
  2. When UCM receives SIP REFER with Refer To URI that has no user part, it sends the INVITE back out to the other server with an @ symbol prepended and this causes the mediation server parser to fail the call, thus disconnecting the TCP session.

  These issues have been addressed in CUCM 8.5.1.1.12900-7

• The Microsoft Mediation Server does not support video transcoding, thus video is not possible between Lync and UCM endpoints using Direct SIP.
System Components

Hardware Requirements

The following hardware was tested:

- Cisco MCS 7825H servers
- Cisco Unified IOS gateway 3825 (with VIC 4FXS/DID, VIC2-4FXO, VWIC2-2MFT-T1/E1 modules)
- Cisco Catalyst 3560 V2 series with 48-port PoE
- Two Cisco 7961 IP Phone (SCCP)
- One Cisco 9971 IP Phone (SIP)
- One Cisco 7971 IP Phone (SCCP)
- One Cisco 7975 IP Phone (SIP)
- Two Third Party PBX Phones
- DELL notebook computers running Windows 7 Ultimate N (for Lync Client)

Software Requirements

The following software was tested:

- Cisco Unified Communications Manager Release 8.5(1)
- Microsoft Lync 2010 (build 4.0.7577.4)
- Microsoft Lync Interoperability Tool (build 4.0.7577.4)
- Microsoft Attendant Console

IMPORTANT:

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

This section lists supported and unsupported features.

Features Supported

- Inbound/Outbound Basic calls (G711 u-law and a-law) (RFC 3261)
- Call Forward (RFC 3261)
- Conference calls
- DTMF (RFC 2833)
- Failover
- Hairpin
- Options
- Refer feature
- Reliable 1XX Provisional Response or PRACK (RFC 3262)
- Early Media (RFC 3261)
- Call hold and resume using Offer/Answer Model (RFC 3264)
- ISDN Mapping
- Early Offer for Media Bypass (MTP required, see note in the Limitations section above)

Features Not Supported and Not Tested

- Encryption (TLS-SRTP) – Not tested.
- Comfort noise is not supported
- Decline a call (CUCM phones divert calls to voicemail, instead of declining the call)
- CUCM does not support using a FQDN in the connection field, a lua script needs to be used on CUCM to achieve this
- For the unplanned outage with 2 Lync clusters with each 2 Mediation servers, CUCM does not support trying a different cluster on the INVITE timeout during Failover testing (See note in the Limitations section above)
**Configuration**

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

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

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


*For Microsoft deployment and installation guides, go to:*

Configuring the Microsoft Lync 2010 Server Standard Edition

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

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

Domain Name System Configuration

Start > Administrative Tools > DNS

Forward Lookup Zone

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

PTR Records added for the Lync server and Active Directory.
Pointer records added for Cisco UCM.

<table>
<thead>
<tr>
<th>Name</th>
<th>Type</th>
<th>Data</th>
<th>Timestamp</th>
</tr>
</thead>
<tbody>
<tr>
<td>(same as parent folder)</td>
<td>Start of Authority (SOA)</td>
<td>mnc2010ded-rms.lync2010rms.com</td>
<td>state</td>
</tr>
<tr>
<td>172.20.91.10</td>
<td>Pointer (PTR)</td>
<td>cmc2.lync2010rms.com</td>
<td>state</td>
</tr>
<tr>
<td>172.20.91.110</td>
<td>Pointer (PTR)</td>
<td>cmc1.lync2010rms.com</td>
<td>state</td>
</tr>
<tr>
<td>(same as parent folder)</td>
<td>Pointer (PTR)</td>
<td>cmc1.lync2010rms.com</td>
<td>state</td>
</tr>
<tr>
<td>(same as parent folder)</td>
<td>Name Server (NS)</td>
<td>lync2010ded-rms.lync2010rms.com</td>
<td>state</td>
</tr>
</tbody>
</table>
SRV Record added for SIP domain service offered by the Lync pool (for Automatic Client Sign-in).
Additionally for failover testing CUCM was provisioned with two trunks. Each trunk used SRV records. The first SRV record returns one record and the second SRV record returns two records. For the Lync server configuration a total of 3 mediation servers were used with 2 servers provisioned in one mediation pool and the remaining server in another pool. Medserver34 pool consists of 2 mediation servers.
### Table: SRV Properties

<table>
<thead>
<tr>
<th>Name</th>
<th>Type</th>
<th>Data</th>
<th>Ttl</th>
</tr>
</thead>
<tbody>
<tr>
<td>_sip</td>
<td>Service Location (SRV)</td>
<td>11 11 11 12 11 13</td>
<td>static</td>
</tr>
</tbody>
</table>

### SRV Properties

- **Domain:** MedExpress14.Cisco.com
- **Service:** sip
- **Protocol:** tcp
- **Priority:** 1
- **Weight:** 10
- **Port Number:** 5060

Host offering this service:
MedExpress14.Cisco.com
Lync Server 2010 Site Topology Overview (Page 1 of 3)
CUCM must be added in Topology Builder as a PSTN gateway. PSTN gateway is the generic term to refer to IP-PBX and other gateway devices. The topology must be published to Central Management store. After the PSTN gateway (that is, CUCM) is added to the topology, it appears in the Lync Server Control Panel. The first thing to do is to add a trunk to the IP-PBX (that is, CUCM).
Lynd Server 2010 Site Topology Overview (Page 3 of 3)
Mediation Server overview (Page 2 of 6)

Lync Front End Server FQDN
Mediation Server overview (Page 3 of 6)

Mediation Server pool consisting of 2 mediation servers used for Failover testing.
The first cluster for failover MedServer-LYNC2010FE1-RTM.lync2010rtm.com is associated with CUCM-ExUM10.lync2010rtm.com, which is the PSTN gateway for the cluster.
Cluster two which is MedServer34 Cluster consists of two mediation servers, which are MedServer3 and Lync2010MEDPRTM.
CUCM1.lync2010rtm.com is the PSTN gateway selected in the mediation server cluster MedServer34.
Create users from Front End Server by accessing the Active Directory Users and Computers window.
Active Directory User Configuration (Page 2 of 3)

Create users from Front End Server by accessing the Active Directory Users and Computers window
Active Directory User Configuration (Page 3 of 3)

Create users from Front End Server by accessing the Active Directory Users and Computers window
Lync Server 2010 Configuration

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

User Configuration from Control Panel (Page 1 of 3)

Start > All Programs > Microsoft Lync Server 2010 > Lync Server Control Panel
To add users got to Home > Enable users for Lync Server and then go to Users > Add and then do a find as shown below.
User Configuration from Control Panel (Page 2 of 3)
In Lync Server, best practice is to use a dial plan that is based on the E.164 standard. This allows easier routing and troubleshooting as well as a scalable model for growth. It is best practices to represent numbers in an E.164 format (External DID’s as well as internal extensions).
Lync Server normalizes all outbound calls as E.164. This allows uniform routing that scales globally across the Lync Server deployment. Phone numbers are normalized to E.164 by normalization rules in this case the rule 4dig-ext. Normalization rules are added in the Dial Plan tab.
Calling features are selected in Voice policy tab. New user policy is created to dial to the CUCM. PSTN usage should be associated here.
Voice Routing Configuration (Page 3 of 5)

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

After the trunk is configured, routes can be configured to route PBX extensions to the trunk to CUCM. If PSTN connectivity is configured through CUCM, calls to PSTN from Lync 2010 can be routed to CUCM through the trunk.
Voice Routing Configuration (Page 4 of 5)

On the Trunk Configurations tabbed page, configure the following parameters:

- **Encryption Level**: Refers to whether media encryption should be required, optional, or not supported on the trunk. This setting depends on whether the CUCM is configured for SRTP.

- **Media Bypass**: Enable it if you want to support Media Bypass for media traffic on this trunk. If Media Bypass is enabled on this trunk, there are several other settings that must be configured for this to work:
  
  o **Enable refer support** must be turned on if needed.
  o The **RTP Callson Hold** and **RTP ActiveCalls** parameters must be turned off. Real-time transport control protocol (RTCP) is a control channel that uses the same RTP channel to monitor the network-specific conditions of the real-time transport protocol (RTP) channel. CUCM doesn’t support RTCP. If these variables are not disabled, the Mediation Server expects keep alives for the RTP channel from CUCM, which may cause unexpected behavior.

  o The RTCP parameters are not displayed through Lync Server Control Panel and have to be configured by using Windows PowerShell:
    
    ```powershell
    Set-CsTrunkConfiguration -Identity site:{site name} -EnableBypass $True -RTPActiveCalls $False -RTPCallsonHold $False –EnableSessionTimer $True
    ```

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Page 35 of 102
The Session Timer parameter must be enabled. Because the RTCP channel is disabled, session timers must be enabled so that calls don’t stay up indefinitely in case the call doesn’t get properly torn down.

Global Configuration for Media Bypass must be configured to Always Bypass under Network Management in Lync Server. This setting cannot be used in conjunction with call admission control (CAC).

Lync Server/Client Address Book Updating

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

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

```
Update-CsAddressBook
```

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

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

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

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

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

**Lyric 2010 Server draining mode**
Stop-CsWindowsService -Graceful rtemdsrv

**Media Bypass**
Media Bypass is disabled by default. Please note that every time you change media bypass settings, you have to logout from your Lyric client and login back again.

In order to always enable Media Bypass on proxy side and trunk side you have to run the following commands from Lyric Management PowerShell:

```powershell
$t=Get-CSNetworkConfiguration –Identity Global
$t.MediaBypassSettings.Enabled=$true
$t.MediaBypassSettings.AlwaysBypass=$true
$t.MediaBypassSettings.InternalBypassMode='any'
Set-CSNetworkConfiguration –Identity Global –MediaBypassSettings $t.MediaBypassSettings
Set-CSTrunkConfiguration –EnableBypass $true
```

Also, if EnableReferSupport is false then do these:

```powershell
Set-CSTrunkConfiguration -RTCPActiveCalls $false
Set-CSTrunkConfiguration -RTCPCallsOnHold $false
Set-CSTrunkConfiguration -EnableSessionTimer $true
```

In order to enable bypass on proxy side and trunk side for siteID you have to run the following commands from Lyric Management Shell:

```powershell
$t=Get-CSNetworkConfiguration –Identity Site:<siteID>
$t.MediaBypassSettings.Enabled=$true
Set-CSNetworkConfiguration –Identity Site:<siteID> –MediaBypassSettings $t.MediaBypassSettings
Set-CSTrunkConfiguration –EnableBypass $true
```

(Note* If needed for CSNetworkConfiguration bypass ID and site ID can be retrieved from topology document.)

In order to disable Media Bypass on proxy side and trunk side you have to run the following commands from Lyric Management PowerShell:

```powershell
$t=Get-CSNetworkConfiguration –Identity Global
$t.MediaBypassSettings.Enabled=$false
$t.MediaBypassSettings.AlwaysBypass=$false
$t.MediaBypassSettings.InternalBypassMode='off'
Set-CSNetworkConfiguration –Identity Global –MediaBypassSettings $t.MediaBypassSettings
Set-CSTrunkConfiguration –EnableBypass $false
```
**Some Lync2010 useful commands**

Get-CsTrunkConfiguration
Set-CsTrunkConfiguration
Get-CsMediaConfiguration
Set-CsMediaConfiguration

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

Identity : Global
OutboundTranslationRulesList : {}
SipResponseCodeTranslationRulesList : {}
Description :
ConcentratedTopology : True
EnableBypass : False
EnableMobileTrunkSupport : False
EnableReferSupport : True
EnableSessionTimer : False
EnableSignalBoost : False
MaxEarlyDialogs : 20
RemovePlusFromUri : False
RTCPActiveCalls : True
RTCP CallsOnHold : True
SRTPMode : Required
EnablePIDFLOSupport : False

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

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

**Lync2010 - enable music on hold:**
set-esclientpolicy -EnableClientMusicOnHold $TRUE
Microsoft Lync 2010 Configuration (Page 1 of 7)

Navigation: Choose Tools → Options and enter the sign-in information.
Click Advanced button to select the Advanced Connection Settings.
If there are DNS entries for this Microsoft Lync, automatic configuration can be used if not select manual configuration.
Sign in to Microsoft Lync.
Add contacts.
Modify user options as needed.
Adding Lync client as the work phone in the Phones tab.
Call-Forwarding Settings:

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

1. Cisco Unified Communications Manager Version
2. SIP Trunk Configuration
3. Media Resources Group Configuration
4. Cisco IP Phones Configuration
5. Route Pattern Configuration
6. Translation Pattern Configuration
7. SIP Gateway Configuration
Cisco Unified Communications Manager Version

Cisco Unified CM Administration
System version: 8.5.1.10000-26

Last Successful Logon: Aug 22, 2011 10:12:15 AM
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This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

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

For information about Cisco Unified Communications Manager please visit our Unified Communications System Documentation web site.

For Cisco Technical Support please visit our Technical Support web site.
SIP Trunk Configuration

SIP Trunk Configuration to Microsoft Lync 2010 Standard Edition Server

From the Cisco UCM Administration page, go to Device then click Trunk. Add a SIP Trunk connecting to the Microsoft Mediation Server. For Media Bypass ensure that the MTP Required box is checked with Early Offer.

Configure and assign the Media Resource Group List to the SIP trunk to ensure Media Termination Point and Conferencing resources are available if needed.
SIP Trunk Configuration to Microsoft Lync 2010 Server

For **Inbound Calls** select **All** as the significant digits to accept all number patterns incoming into the Cisco UCM from Microsoft Mediation Server.

Under SIP Information make sure **Destination Address** contains the IP Address of the Microsoft Mediation Server.
remove-tias is the normalization script to remove TIAS field from the SDP is used. The destination address is the IP address of the Mediation server. Also the SIP profile needs to be changed per the configuration, it is always default for delayed offer and changed according to the configuration needed.

**SIP Profile Configuration**

Below is the configuration when Early Offer and PRACK required feature of Cisco UCM 8.5 is enabled on the SIP Profile that is associated with the trunk. PRACK is required for early media test cases and Early Offer is required for Media Bypass testing. Here the **Early Offer Support For Voice and Video Calls** parameter needs to be enabled.
### Status
- Status: Ready
- All SIP devices using this profile must be restarted before any changes will take effect.

### SIP Profile Information
<table>
<thead>
<tr>
<th>Option</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default HTPP Event Record Type</td>
<td>RO SIP Profile for HTPP required</td>
</tr>
<tr>
<td>Route Priority NameSpace Set</td>
<td>None</td>
</tr>
<tr>
<td>Early Offer for O-Clear Calls</td>
<td>Disabled</td>
</tr>
<tr>
<td>* * *</td>
<td></td>
</tr>
</tbody>
</table>

### Parameters used in Phone
- **Timer Override Express (seconds)**: 180
- **Timer Register Delta (seconds)**: 6
- **Timer Register Express (seconds)**: 6000
- **Timer T1 (ms**): 5000
- **Timer T2 (ms**): 4000
- **Retry IN/OUT**: 6
- **Retry In/Out Time**: 10
- **Start Media Time**: 10000
- **Stop Media Time**: 5000
- **Call Pickup URI**: `tel:service-2-digit`
- **Call Pickup Group Other URI**: `tel:service-2-digit`
- **Call Pickup Group URI**: `tel:service-2-digit`
- **Meet Me Service URI**: `tel:service-2-digit`
- **User Info**: `None`
- **DTPP DS Level**: General
- **Call Hold Ring Back**: Off
- **Anonymous Call Block**: Off
- **Caller ID Blocking**: Off
- **Do Not Disturb**: User
- **Telnet Level for 7941 and 7960**: Disabled
- **Timer Keep Alive Express (seconds)**: 120
- **Timer Subscribe Express (seconds)**: 120
- **Timer Subscribe Delta (seconds)**: 8
- **Maximum Redirects**: 30
- **Off Hook To First Digit Timer (milliseconds)**: 8000
- **Call Forward URI**: `tel:service-2-digit`
- **Speed Dial (Abbreviated Dial URI)**: `tel:service-2-digit`
- **Conference Join Enable**: Yes
- **Emergency Hold**: Off
- **SIP Media Transfer**: Disabled
- **Contact Message Waiting**: Off

### Trunk Specific Configuration
- Remote Incoming Request to new Trunk based on **None**
- **RTP Over SIP**: Level RSVP
- **Send back to local RSVP**: Yes

### SIP OPTIONS
- **Enable Options**: Ring to monitor destination status for Trunk with Service Type "None (Default)"
- **Ring Interval for In-service and Partially In-service Trunks (seconds)**: 300
- **Ring Entry Time (milliseconds)**: 1000
- **Ring Entry Count**: 0

### Additional Options
- **Save**
- **Delete**
- **Copy**
- **Reset**
- **Apply Config**
- **Add New**
Below is the configuration when Early Offer is enabled and PRACK feature of Cisco UCM 8.5 is disabled on the SIP Profile that is associated with the trunk.
# Cisco Unified CM Administration

**SIP Profile Configuration**

### Status
- Status: Ready

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

### SIP Profile Information
- **Name**: Early Offer SIP Profile
- **Description**: Default SIP Profile
- **Default</b> SIP Telephone Event Pick up Type**: 101
- **Ringing Priority NameSpace Set**: None
- **Early Offer for 0-Char Calls**
- **Do Not Play by Application**
- **Disable Early Media on 120**
- **Enable AMF**
- **Disable SIP Trunk Exchange for Rich Call Change**

### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Local Timer Value (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>4000</td>
</tr>
<tr>
<td>Timer T1 (millisecs)</td>
<td>5000</td>
</tr>
<tr>
<td>Timer T2 (millisecs)</td>
<td>4000</td>
</tr>
<tr>
<td>Early INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Early NO/INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Time</td>
<td>0</td>
</tr>
<tr>
<td>Stop Media Time</td>
<td>0</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>0</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>0</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>0</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>0</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DND DS Level</td>
<td>Normal</td>
</tr>
<tr>
<td>Call Hold Ring Busy</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Callback 3D Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Answer Call</td>
<td>User</td>
</tr>
<tr>
<td>TAF Level for 7941 and 7960</td>
<td>Disabled</td>
</tr>
<tr>
<td>Time Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections</td>
<td>0</td>
</tr>
<tr>
<td>OFF Hook To First Digit Timer (milliseconds)</td>
<td>0</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>0</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial)</td>
<td>0</td>
</tr>
<tr>
<td>Conference Join Embedded</td>
<td>Yes</td>
</tr>
<tr>
<td>DND 304 Held</td>
<td>Yes</td>
</tr>
<tr>
<td>Semi-Attached Transfer</td>
<td>Yes</td>
</tr>
<tr>
<td>Enable VAD</td>
<td>Yes</td>
</tr>
<tr>
<td>Cluster Feature Waiting</td>
<td>Yes</td>
</tr>
</tbody>
</table>

### Trunk Specific Configuration
- **Enable Inbound T1 for trunk based on**: N/A
- **Enable SIP over SIP**: Level 5 SIP
- **Fallback to local SIP**: Disabled
- **Enable Conference Bridge Identifier**: Enabled
- **Early Offer Support for voice and video calls (insert HTTP if needed)**
- **Send SendReceive SCP in mid-call invite**

### SIP OPTIONS
- **Enable OPTIONS**: Ring to monitor destination status for Trunk with Service Type "None (Default)"
- **Ring Interval for In-service and Partially In-service Trunks (seconds)**
- **Ring Interval for Out-of-service Trunks (seconds)**
- **Ring Entry Time (milliseconds)**
- **Ring Entry Count**

---

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Below is the configuration when Early Offer and PRACK supported feature of Cisco UCM 8.5 is enabled on the SIP Profile that is associated with the trunk. PRACK is required for early media test cases and Early Offer is required for Media Bypass testing. Here the Early Offer Support For Voice and Video Calls parameter needs to be enabled.
### SIP Profile Configuration

**Status**
- Status: Ready
- All SIP devices using this profile must be retrained before any changes will take effect.

#### SIP Profile Information

- **Name**: [Insert Name]
- **Description**: [Insert Description]
- **Default HTP Telephony Event Handling Type**: SIP
- **Erase Priority Namespace Set**: [None]
- **Early Offer for O-Clear Calls**: [Disabled]
- **Do Not Retract by Application**: [Disabled]
- **Do Not Transmit on 180**: [Enabled]
- **Outgoing 10 Digit IN/OUT include audio mline**: [Enabled]
- **Enable AMT**: [Disabled]
- **Require SDP Inactive Exchange for Mid-Call Media Change**: [Enabled]

#### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Drive Expires (seconds)</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>0</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>30000</td>
</tr>
<tr>
<td>Timer Ti (milliseconds)</td>
<td>5000</td>
</tr>
<tr>
<td>Timer T2 (milliseconds)</td>
<td>4000</td>
</tr>
<tr>
<td>Early INTUE</td>
<td>6</td>
</tr>
<tr>
<td>Early NonINTUE</td>
<td>10</td>
</tr>
<tr>
<td>Short Media Port</td>
<td>32768</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>32768</td>
</tr>
<tr>
<td>Call Pickup URL</td>
<td>[Insert URL]</td>
</tr>
<tr>
<td>Call Pickup Group Other URL</td>
<td>[Insert URL]</td>
</tr>
<tr>
<td>Call Pickup Group URL</td>
<td>[Insert URL]</td>
</tr>
<tr>
<td>Meet Me Service URL</td>
<td>[Insert URL]</td>
</tr>
<tr>
<td>User Info</td>
<td>[None]</td>
</tr>
<tr>
<td>DTPP DS Level</td>
<td>[None]</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>[Off]</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>[Off]</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>[Off]</td>
</tr>
<tr>
<td>Do Not Default Contact</td>
<td>[Enabled]</td>
</tr>
<tr>
<td>Telnet Level for 7945 and 7960</td>
<td>[120]</td>
</tr>
<tr>
<td>Timer Keep Alive (seconds)</td>
<td>[120]</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>[5]</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>[5]</td>
</tr>
<tr>
<td>Maximum Redirections</td>
<td>[70]</td>
</tr>
<tr>
<td>Off-Hook To First Digit Timer (milliseconds)</td>
<td>[10000]</td>
</tr>
<tr>
<td>Call Forward URL</td>
<td>[Insert URL]</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated dial)</td>
<td>[Insert URL]</td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td>[Enabled]</td>
</tr>
<tr>
<td>RFC 2833 Held</td>
<td></td>
</tr>
<tr>
<td>Semi-Attended Transfer</td>
<td>[Enabled]</td>
</tr>
<tr>
<td>Cluster Members Waiting</td>
<td></td>
</tr>
</tbody>
</table>

#### Trunk Specific Configuration

- [Insert Trunk Specific Options]

#### SIP OPTIONS

- [Insert SIP Options]

---

**Note**: Indicates required items.
**Configuration for Lync IT Tool and Failover**

Lync-MedServer34-Cluster is the trunk for failover second mediation pool.

<table>
<thead>
<tr>
<th>Device Information</th>
<th>SIP Trunk</th>
<th>MediaServer/M-Cluster</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
<td>MedEL Cluster</td>
</tr>
<tr>
<td>Trunk Device Type</td>
<td>MedEL</td>
<td></td>
</tr>
<tr>
<td>Description</td>
<td>MedEL</td>
<td></td>
</tr>
<tr>
<td>Common Device Config</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Media Resource Group</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Audio Group</td>
<td>MedEL</td>
<td></td>
</tr>
<tr>
<td>Tunnelled Protocol</td>
<td>MedEL</td>
<td></td>
</tr>
<tr>
<td>Content-type</td>
<td>MedEL</td>
<td></td>
</tr>
<tr>
<td>Media Capture Mode</td>
<td>MedEL</td>
<td></td>
</tr>
<tr>
<td>Failover Capture</td>
<td>MedEL</td>
<td></td>
</tr>
</tbody>
</table>

- **SIP Trunk**: SIP
- **MediaServer/M-Cluster**: MedEL Cluster
- **Common Device Config**: MedEL
- **Media Resource Group**: MedEL
- **Audio Group**: MedEL
- **Tunnelled Protocol**: MedEL
- **Content-type**: MedEL
- **Media Capture Mode**: MedEL
- **Failover Capture**: MedEL

**Notes:**
- Video Call as Audio
- Media Replacement Support
- Transparent RTP for Calling Party Name
- Temporary RTP for CALLID PDU
- Unattended Port

**Consider Traffic on The Trunk:**
- Select NTP and TLS
- Use Trusted SIP comm
- PSTN Access

---

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LyncIT_SIP_Trunk is the sip trunk to the Lync IT tool.
Configuration of MGCP Gateway to PBX

Configuration of MGCP Gateway to PBX (Page 1 of 4)
Configuration of MGCP Gateway to PBX (Page 2 of 4)
Using the correct switch type on the MGCP gateway is critical for seamless operation of the trunk.
Configuration of MGCP Gateway to PBX (Page 4 of 4)

Media Resource Group Configuration

From the Cisco UCM Administration page, click Media Resources and choose Media Resource Group to create a new group with all needed resources for the trunk.
The below shows the MRG configuration with hardware MTP.
The below shows the MRG configuration with software MTP.
### Media Resource Group Configuration

**Status**

- Status: Ready

**Media Resource Group Status**

- Media Resource Group: HWG_SW_HIP (used by 7 devices)

**Media Resource Group Information**

- **Name**: HWG_SW_HIP
- **Description**: HWG_SW_HIP

**Devices for this Group**

- **Available Media Resources**: mp3937/3421100c

Selected Media Resources:

- AMP2_SW (SW)
- AMP2_SW (SW)
- AMP2_SW (SW)

- Use Multi-cast for HD Audio (if at least one multi-cast HDH resource is available)

---

* indicates required item.

** Includes Annunciations (ANX), Conference Bridges (CBPs), Media Termination Points (MTPs), Music On Hold Servers (MOHs) and Transcoders (HCCDE)
The above shows the MRG configuration with no MTP.

Media Resource Group List Configuration

Click Media Resources and choose Media Resource Group List to create a new list and assign the newly created group to it.
The above shows the MRGL configuration with media resource group that has no MTP.
The above shows the MRGL configuration with media resource group that has hardware MTP resource group.
The above shows the assignment of MRGL configuration with software MTP resource group. Assign corresponding MRG to the created MRGL.
Cisco IP Phone Configuration

From the “Cisco Unified CM Administration” page, go to “Device” then click “Phone”.

Add corresponding IP Phones. Shown below are the IP phones that were configured for this setup.

### Cisco 7971 SCCP Configuration

<table>
<thead>
<tr>
<th>Device Name/(Line)</th>
<th>Description</th>
<th>Device Pool</th>
<th>Device Protocol</th>
<th>Status</th>
<th>IP Address</th>
<th>Copy</th>
<th>Super Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>REP0771G9SK2K5</td>
<td>7971 SCCP Phone - Jane Austen x4513</td>
<td>Default</td>
<td>SCCP</td>
<td>Registered with CUOM-EXUM10</td>
<td>172.30.131.10</td>
<td>☑️</td>
<td>☑️</td>
</tr>
<tr>
<td>REP0771G9SK2K5</td>
<td>7975 SIP Phone - John Smith x4511</td>
<td>Default</td>
<td>SIP</td>
<td>Registered with CUOM-EXUM10</td>
<td>172.30.131.11</td>
<td>☑️</td>
<td>☑️</td>
</tr>
<tr>
<td>REP0771G9SK2K5</td>
<td>7971 SCCP Phone - David lynch x4513</td>
<td>Default</td>
<td>SCCP</td>
<td>Registered with CUOM-EXUM10</td>
<td>172.30.131.11</td>
<td>☑️</td>
<td>☑️</td>
</tr>
<tr>
<td>REP0771G9SK2K5</td>
<td>7971 SIP Phone - Diana Wedley x4514</td>
<td>Default</td>
<td>SIP</td>
<td>Registered with CUOM-EXUM10</td>
<td>172.30.130.8</td>
<td>☑️</td>
<td>☑️</td>
</tr>
</tbody>
</table>
Select the MRGL per configuration.
Select the MRGL per configuration.
### Phone Configuration

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>None</td>
</tr>
<tr>
<td>User Locale</td>
<td>None</td>
</tr>
<tr>
<td>Network Locale</td>
<td>None</td>
</tr>
<tr>
<td>Built In Bridge</td>
<td>Default</td>
</tr>
<tr>
<td>Privacy</td>
<td>Default</td>
</tr>
<tr>
<td>Device Mobility Node</td>
<td>Default</td>
</tr>
<tr>
<td>Owner User ID</td>
<td>User04</td>
</tr>
<tr>
<td>Phone Personalization</td>
<td>Default</td>
</tr>
<tr>
<td>Services Provisioning</td>
<td>Default</td>
</tr>
<tr>
<td>Phone Load Name</td>
<td>Default</td>
</tr>
<tr>
<td>Single Button Suggest</td>
<td>Default</td>
</tr>
<tr>
<td>Join Across Lines</td>
<td>Default</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
</tr>
<tr>
<td>BFL audible Alert Setting (Phone Edit)</td>
<td>Default</td>
</tr>
<tr>
<td>BFL audible Alert Setting (Phone Busy)</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line for Voice Message</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>Default</td>
</tr>
<tr>
<td>Geolocation</td>
<td>Default</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>Default</td>
</tr>
<tr>
<td>Entry Video Call as Audio</td>
<td>Default</td>
</tr>
<tr>
<td>Ignore Presentation Indicators (internal calls only)</td>
<td>Default</td>
</tr>
<tr>
<td>Allow Control of Device from CTI</td>
<td>Default</td>
</tr>
<tr>
<td>Logged Into Hunt Group</td>
<td>Default</td>
</tr>
<tr>
<td>Remote Device</td>
<td>Default</td>
</tr>
<tr>
<td>Protected Device</td>
<td>Default</td>
</tr>
<tr>
<td>Hot line Device</td>
<td>Default</td>
</tr>
</tbody>
</table>

### Protocol Specific Information

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Role Name</td>
<td>Admin</td>
</tr>
</tbody>
</table>

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Page 74 of 102
On the “Phone Configuration” page under “Association Information”, click “Line [1] – Add a new DN” to enter the “Directory Number Configuration”.
### Directory Number Configuration

#### Line 1 ON DEVICE SEP001D705P9036

<table>
<thead>
<tr>
<th>Display (Internal Caller ID)</th>
<th>Value</th>
<th>Update Shared Device Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Line Text Label</td>
<td>Jane Austen</td>
<td></td>
</tr>
<tr>
<td>ASCII Display (Internal Caller ID)</td>
<td>Jane Austen</td>
<td></td>
</tr>
<tr>
<td>External Phone Number Plan</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Visual Message Waiting Indicator Policy</td>
<td>Use System Policy</td>
<td></td>
</tr>
<tr>
<td>Audible Message Waiting Indicator Policy</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Ring Setting (Phone Idle)</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Idle)</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Active)</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Recording Option*</td>
<td>Cell Recording Disabled</td>
<td></td>
</tr>
<tr>
<td>Recording Profile</td>
<td>&quot;Home&quot;</td>
<td></td>
</tr>
<tr>
<td>Monitoring Ceiling Speech</td>
<td>Log Mixed Calls</td>
<td></td>
</tr>
</tbody>
</table>

**Multiple Call/Call Waiting Settings on Device SEP001D705P9036**

- **Max Number of Calls**: 1-200
- **Max Trips**: Less than or equal to Max. Calls

**Forwarded Call Information Display on Device SEP001D705P9036** -
### Directory Number Configuration

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ring Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td></td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Recording Option</td>
<td>Call Recording Disabled</td>
</tr>
<tr>
<td>Recording Profile</td>
<td></td>
</tr>
<tr>
<td>Hunting Ring Search</td>
<td></td>
</tr>
<tr>
<td>Logging Housed Calls</td>
<td></td>
</tr>
</tbody>
</table>

### Multiple Call/Call Waiting Settings on Device 10IPD7105P9026

- **Note:** The range to select the Max Number of calls is 1-1000
- **Busy Trigger:** Input is Max Calls

### Forwarded Call Information Display on Device 10IPD7105P9036

- **Call from:**
- **Call Number:**
- **Redirected Number:**
- **Dialed Number:**

### Users Associated with Line

- **Associate End Users:**

---

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Phone 9971 SIP Configuration

Choose the corresponding MRGL, Phone button template.
Put in the secure shell information.
Remote destination and remote destination profile is created for simultaneous ring testing.
Route Pattern Configuration

From the Cisco UCM Administration page, choose Call Routing → Route Hunt → Route Pattern

Route pattern setup to route calls from Cisco UCM to the Microsoft Lync Mediation Server

14152210XX is the remote destination route pattern and Lync2010-RTM_SIP_Trunk is the SIP trunk to the Microsoft Mediation Server. Prefix with a plus sign (+) to match the directory numbers (DN’s) on the LYNC side, in case the normalization rules in the Microsoft LYNC environment do not already add the plus sign.
SIP Route Pattern For REFER testing
Make sure that the SIP trunk is the trunk that goes to the Mediation Server in our case Lync2010-RTM_SIP_Trunk.

Route Pattern setup to route certain numbers Cisco UCM out to the PBX.

The T1 port gateway to the PBX is S0/SU1/DSI-0@MS_GW1. 4XXX is the PSTN route pattern.
Route Pattern setup to route certain numbers from Cisco UCM out to Lync 2010 Client.
Route Pattern setup to route certain numbers from Cisco UCM out to Lync IT Client

Cisco Unified CM Administration

Route Pattern Configuration

Status

Pattern Definition
- Route Pattern: 222012X
- Description: Route Pattern to Lync IT Client
- Numbering Plan: Not Selected
- Route Filter: None
- H.323 Preference: Default
- Resource Priority Namespace Network Domain: < Name >
- Route Class: Default
- Gateway/Route List: LyncIT_SIP_Trunk
- Route Option: Route this pattern
- Block this pattern: No Error

Call Classification:
- DialPlan Override
- Provide Outside Dial Tone
- Allow Overlay Budding
- Allow Priority
- Require Forced Authorization Code
- Authorization Level:
- Require Client Caller Code

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

Connected Party Transformations
- Connected Line ID Presentation: Default
- Connected Name Presentation: Default

Called Party Transformations
- Discard Digits
- Called Party Transform Mask
- Prefix Digits (Outgoing Calls)
- Called Party Number Type: Cisco CallManager
- Called Party Numbering Plan: Cisco CallManager

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

Save | Delete | Copy | Add New
Translation Pattern Configuration

From the Cisco UCM Administration page, click Call Routing then choose Translation Pattern.

Add a Translation Pattern for internal dialing on the Cisco UCM to enable 5 digits dialing to an E164 extension inside the Cisco UCM dial plan.
This pattern translates 4-digit extensions to the E.164 DN's of the IP phones.
The following normalization script is used to strip off the TIAS field inside the SDP towards the Microsoft Lync 2010.

M = {}

local function remove_tias(msg)
  M = {}
  local function remove_tias(msg)
    local sdp = msg:split('SDP')
    if sdp then
      local n = 0
      sdp = sdp:sub(n, n)
      if n > 0
        -- the substitution occurred and therefore the DCP changed
        msg:split(sdp:sub(n, n))
    end
  end
  -- indicates required item.

  <<Cisco UCM Normalization Script>>
local sdp = msg:getSdp()

if sdp

then

local n = 0

sdp, n = sdp:gsub("b=TIAS\[^\r\n]+\r?\n", ",")

-- Did the substitution occur?

if n > 0

then

-- The substitution occurred and therefore the SDP changed.

msg:setSdp(sdp)

end

end
end

-- These are the messages sent by CUCM that include SDP.

M.outbound_INVITE = remove_tias
M.outbound_180_INVITE = remove_tias
M.outbound_183_INVITE = remove_tias
M.outbound_200_INVITE = remove_tias
M.outbound_ACK = remove_tias
M.outbound_UPDATE = remove_tias
M.outbound_200_UPDATE = remove_tias
M.outbound_PRACK = remove_tias
M.outbound_200_PRACK = remove_tias
return M

Service Parameters for Refer
### Clustered Parameters (Feature - Call Pickup)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Auto Call Pickup Enabled</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Call Pickup Location Times</td>
<td>3</td>
<td>1</td>
</tr>
<tr>
<td>Call Pickup No Answer Time</td>
<td>5</td>
<td>0</td>
</tr>
</tbody>
</table>

### Clustered Parameters (Feature - Flow)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Validate Flow in Call</td>
<td>Always</td>
<td>Never</td>
</tr>
</tbody>
</table>

### Clustered Parameters (Feature - Replaces)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Block Office-To-Office Features</td>
<td>False</td>
<td>True</td>
</tr>
</tbody>
</table>

### Clustered Parameters (Feature - Redirection [3xx])

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Redirection Count</td>
<td>24</td>
<td>24</td>
</tr>
</tbody>
</table>

### Clustered Parameters (Feature - Multilevel Precedence and Preemption)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Execute Override Call Preemptible</td>
<td>False</td>
<td>True</td>
</tr>
</tbody>
</table>

### Clustered Parameters (Feature - Path Replacement)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Path Replacement Enabled</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Path Replacement on Troubled Calls</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Start Path Replacement Minimum Delay Time</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Start Path Replacement Maximum Delay Time</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Path Replacement SS Time</td>
<td>30</td>
<td>30</td>
</tr>
<tr>
<td>Path Replacement On Time</td>
<td>15</td>
<td>15</td>
</tr>
</tbody>
</table>
Configuring the MGCP Gateway

MS_GW1#sh ver

Cisco IOS Software, 3800 Software (C3825-IPVOICEK9-M), Version 15.0(1)M3, RELEASE SOFTWARE (fc2)

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

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Compiled Sun 18-Jul-10 05:11 by prod_rel_team

ROM: System Bootstrap, Version 12.3(11r)T2, RELEASE SOFTWARE (fc1)

MS_GW1 uptime is 25 weeks, 6 days, 21 hours, 22 minutes

System returned to ROM by power-on

System image file is “flash:c3825-ipvoicek9-mz.150-1.M3.bin”

Last reload type: Normal Reload

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

A summary of U.S. laws governing Cisco cryptographic products may be found at:


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

Cisco 3825 (revision 1.1) with 227328K/34816K bytes of memory.

Processor board ID FTX1025A25B

2 Gigabit Ethernet interfaces

24 Serial interfaces

2 Channelized T1/PRI ports

4 Voice FXO interfaces

4 Voice FXS interfaces
DRAM configuration is 64 bits wide with parity enabled.

479K bytes of NVRAM.

62720K bytes of ATA System CompactFlash (Read/Write)

License Info:

License UDI:

--------------------------------------------------
Device#   PID       SN
--------------------------------------------------
*0   CISCO3825      FTX1025A25B

Configuration register is 0x2102

MS_GW1#sh run
Building configuration...

Current configuration : 8455 bytes

! Last configuration change at 04:18:13 UTC Thu Jan 13 2011
!
version 15.0
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname MS_GW1
!
boot-start-marker
boot system flash c3825-ipvoicek9-mz.150-1.M3.bin
boot-end-marker
!
card type t1 0 1
logging buffered 10000000
enable secret 5 $1$rqA7$1Y9cTQOjnATrchIPN40
enable password CSCO123
!
no aaa new-model
network-clock-participate wic 1
network-dock-select 1 T1 0/1/0
!
dot11 syslog
ip source-route
ip cef
!
!
! ip host CUCM-ExUM10 172.20.85.110
no ipv6 cef
multilink bundle-name authenticated
!
!
! isdn switch-type primary-4ess
!
voice-card 0
dspfarm
dsp services dspfarm
!
!
voice service pots
!
voice service voip
qsig decode
sip
session transport tcp
header-passing
asserted-id pai
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g711alaw
!
!
!
license udi pid CISCO3825 sn FTX1025A25B
archive
log config
hidekeys
username cisco privilege 15 secret 5 $1$g9MP$FSHytQTj7RcJPsHV5.tL/
!
!
controller T1 0/1/0
cablelength long 0db
pri-group timeslots 1-24 service mgep
!
controller T1 0/1/1
shutdown
cablelength long 0db
!
!
!
!
interface GigabitEthernet0/0
description SETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
ip address 172.20.150.200 255.255.255.0
duplex auto
speed auto
media-type rj45
no mop enabled
!
interface GigabitEthernet0/1
no ip address
shutdown
duplex full
speed 100
media-type rj45
service-policy output HQoS
!
interface Serial0/1/0:23
no ip address
encapsulation hdlc
isdn switch-type primary-qsig
isdn timer T310 120000
isdn protocol emulate network
isdn incoming-voice voice
isdn bind-13 ccm-manager
no cdp enable
!
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
no ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 172.20.150.1
!
!
ip prefix-list test seq 5 permit 10.1.1.1/32
access-list 23 permit 10.10.10.0 0.0.0.7
access-list 100 permit udp host 10.1.1.1 host 20.1.1.1 dscp af41 log
!
!
control-plane
!
!
v4ce-port 0/0/0
  station-id name Chinh
  station-id number 4155265001
caller-id enable
!
v4ce-port 0/0/1
  station-id number 7750
caller-id enable
!
v4ce-port 0/0/2
!
v4ce-port 0/0/3
!
v4ce-port 0/1/0:23
echo-cancel coverage 64
!
voice-port 0/2/0
timing hookflash-out 50
!
voice-port 0/2/1
!
voice-port 0/2/2
!
voice-port 0/2/3
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server CUCM-ExUM10
ccm-manager config
!
mgcp
mgcp call-agent CUCM-ExUM10 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode nte-c
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
mgcp package-capability fm-package
no mgcp package-capability res-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
mgcp sd p 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 profile default
!
!
dial-peer voice 999001 pots
  service mgcp app
  destination-pattern 4l..
  incoming called-number ....
  direct-inward-dial
  port 0/10:23
  forward-digits all
  *
  sip ua
  *
  telephony-service
  max-conferences 12 gain -6
  transfer-system full-consult
create cnf-files version stamp Jan 01 2002 00:00:00
!
ephone-hunt 1 longest-idle
!
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
exec-timeout 0 0
privilege level 15
password cisco
login local
transport input telnet
line vty 5 15
access-class 23 in
exec-timeout 0 0
privilege level 15
password cisco
login local
transport input telnet
!
exception data-corruption buffer truncate
scheduler allocate 20000 1000
end

MS_GW1#
## Acronyms

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