Microsoft Lync Server 2010 Standard Edition using Direct SIP to Cisco Unified Communications Manager 8.5(1) Simultaneous Ring Feature
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MS_GW1#sh run

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Introduction

- This document describes the simultaneous ring feature interoperability and documents the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) Release 8.5(1) to interoperate with Microsoft Lync 2010 Standard Edition Server. It aims to provide a good understanding of what works and what does not work in terms of the feature interaction between a Cisco UCM device and Microsoft Lync. It also provides guidance to deployment participants regarding the limitations, expected behaviors, and known issues. Please note that this document does not address performance and scalability, which are part of a broader criteria for a deployment-ready solution (for more details refer to 8.x SRND)

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

- The simultaneous ring feature allows a Cisco UCM endpoint (IP Phone, IP Communicator, etc.) to simultaneously ring its remote destination(s) when a call is placed to that endpoint. The remote destination(s) could be Microsoft Lync client and/or any other device(s) including mobile phones, PSTN phones or even PBX extensions. The remote destination feature uses the Cisco Unified Mobility functionality and can be configured to allow or block the receipt of certain numbers. For the simultaneous ring configuration, Cisco UCM has a Direct SIP connection to Microsoft Mediation Server.

- The endpoints used in this testing all have E.164 numbering which is supported by Cisco UCM Release 8.5(1) and later.

- Cisco UCM’s Delayed Offer feature is used to integrate Lync 2010 Client and Cisco UCM, instead of the traditional MTP feature.

- 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, Offer SDP in 200 OK and Answer SDP in ACK). 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).

- 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 Lync Mediation Server, RTCP is set to disabled, as RTCP is an end point specific configuration and not all Cisco phones support RTCP. Phones like 7970, 7962, 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.

• 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 fails the call, thus disconnecting the TCP session.

The issues has been addressed in CUCM 8.5.1.12900-7

• 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 in this set up.

• 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.

• Microsoft Lync only supports G711 ulaw or alaw on the outside interface. If G729 is required an ISR can perform the transcoding function.

• 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
- 1 Cisco 7960 IP Phone (SCCP)
- 1 Cisco 7970 IP Phone (SIP)
- 1 Cisco 7970 IP Phone (SCCP)
- 1 Cisco 7971 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:

- Call Processing on Cisco: Cisco Unified Communications Manager Release 8.5(1)
- Microsoft Lync 2010 (build 4.0.7577.4)
- Cisco Unity Connection
- 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

- Basic Call Placement / Clearing
- Hold / Retrieval of Call
- Call Transfer
  - Unannounced or Blind
  - Announced or Attended
- Call Forwarding (CFA, CFB, CFNR)
- Call Conference
- Do-Not-Disturb (DND)
- Switch call back to Microsoft Lync client
- Resume call from Microsoft Lync client
- Voice Mail

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)
**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 Server 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 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
Forw ard Lookup Zone

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

<table>
<thead>
<tr>
<th>Name</th>
<th>Type</th>
<th>Data</th>
<th>Ttl</th>
</tr>
</thead>
<tbody>
<tr>
<td>LYNC2010/AAA/IM</td>
<td>Start of Authority (SOA)</td>
<td>[3], lync2010/AAA/IM, lync2010/AAA/Lync2010IM.com., hostmaster.lync2010/AAA/IM.com</td>
<td>static</td>
</tr>
<tr>
<td>LYNC2010/AAA/IM</td>
<td>Name Server (NS)</td>
<td>[3], lync2010/AAA/IM, lync2010/AAA/Lync2010IM.com., hostmaster.lync2010/AAA/IM.com</td>
<td>static</td>
</tr>
<tr>
<td>172.20.117.10</td>
<td>Pointer (PTR)</td>
<td>lync2010/dem1.lync2010IM.com</td>
<td>2020/02/02 2:08:00</td>
</tr>
<tr>
<td>172.20.117.152</td>
<td>Pointer (PTR)</td>
<td>lync2010/dem1.lync2010IM.com</td>
<td>2020/02/02 2:08:00</td>
</tr>
<tr>
<td>172.20.117.131</td>
<td>Pointer (PTR)</td>
<td>lync2010/dem1.lync2010IM.com</td>
<td>2020/02/02 2:08:00</td>
</tr>
<tr>
<td>172.20.117.132</td>
<td>Pointer (PTR)</td>
<td>lync2010/dem1.lync2010IM.com</td>
<td>2020/02/02 2:08:00</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Name</th>
<th>Details</th>
<th>Data</th>
<th>TimeStamp</th>
</tr>
</thead>
<tbody>
<tr>
<td>sip</td>
<td>Service Location (SRV)</td>
<td>[10.0.0.0] lync2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn</td>
<td>09/10/2011 00:00:00</td>
</tr>
<tr>
<td>_sip</td>
<td>Service Location (SRV)</td>
<td>[10.0.0.0] lync2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn</td>
<td>09/10/2011 00:00:00</td>
</tr>
<tr>
<td>_lsip</td>
<td>Service Location (SRV)</td>
<td>[10.0.0.0] lync2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn</td>
<td>09/10/2011 00:00:00</td>
</tr>
<tr>
<td>_unified</td>
<td>Service Location (SRV)</td>
<td>[10.0.0.0] lync2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn</td>
<td>09/10/2011 00:00:00</td>
</tr>
<tr>
<td>_sip颁发</td>
<td>Service Location (SRV)</td>
<td>[10.0.0.0] lync2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn</td>
<td>09/10/2011 00:00:00</td>
</tr>
<tr>
<td>_sip颁发</td>
<td>Service Location (SRV)</td>
<td>[10.0.0.0] lync2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn Lynx2010dmn-mn</td>
<td>09/10/2011 00:00:00</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 with Direct SIP 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.
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)
Mediation Server Overview (Page 1 of 6)
Mediation Server overview (Page 2 of 6)

Lync Front End Server FQDN

![Lync Server 2010 Topology Builder](image)
Mediation Server overview (Page 3 of 6)

Mediation Server pool consisting of 2 mediation servers used for Failover testing.
SQL stores
PSTN gateway selected in the mediation server configuration of the first mediation pool which is the front end server.
CUCM1.lync2010rtm.com is the PSTN gateway selected in the mediation server cluster MedServer34.
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, go to Home > Enable users for Lync Server and then go to Users > Add and then do a find as shown below.
In Lync Server, best practice is to use a dial plan that is based on the E.164 standard. This allows easier routing and troubleshooting as well as a scalable model for growth. It is best practices to represent numbers in an E.164 format (External DIDs as well as internal extensions).
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. User policy is created to dial to the CUCM. PSTN usage should be associated here.
Voice Routing Configuration (Page 3 of 6)

Local route added in the Route tab.
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.
PSTN Usage route added.
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:
- **Enable refer support** must be turned on if needed.
- The **RTCP call on hold** and **RTCP active calls** 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 RTCP channel from CUCM, which may cause unexpected behavior.

- 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 -RTCPActiveCalls $False –RTCPCallsOnHold $False -EnableSessionTimer $True
  ```

  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 force the Lync 2010 Client to pull down the latest files to update its local GalContacts.db file.

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

`Update-CsAddressBook`

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

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

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

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

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

**Lync 2010 Server draining mode**
Stop-CsWindowsService -Graceful rtcmdev

**Media Bypass**
Media Bypass is disabled by default. Please note that every time you change media bypass settings, you have to logout from your Lync 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 Lync Management 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:
```
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 Lync Management Shell:

```
$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 Lync Management 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
RTCPCallsOnHold : True
SRTPMode : Required
EnablePIDFLOSupport : False

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

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

Lync2010 - enable music on hold:
set-csclientpolicy -EnableClientMusicOnHold $TRUE
Active Directory Configuration (Page 1 of 4)

Create users from Front End Server by accessing the Active Directory Users and Computers window
Active Directory Configuration (Page 3 of 4)
Active Directory Configuration (Page 4 of 4)

<table>
<thead>
<tr>
<th>Name</th>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Lync02 RMP Properties**

- **User login name**:>Action
- **Log off**:Action

**Account options**:
- **User must change password at next login**:Action
- **User cannot change password**:Action
- **Password never expires**:Action
- **Hidden password using reversible encryption**:Action

**Expires**:Action
Microsoft Lync 2010 Client Configuration (Page 1 of 7)

Navigation: Choose Tools ➔ Options and enter the sign-in information.
Microsoft Lync 2010 Client Configuration (Page 2 of 7)

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.
Microsoft Lync 2010 Configuration (Page 4 of 7)

Sign in to Microsoft Lync.
Microsoft Lync 2010 Configuration (Page 5 of 7)

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. SIP Trunk Configuration
2. Media resources Group Configuration
3. Cisco IP Phones Configuration
4. Simultaneous Ring Feature Configuration
   - End Users Configuration
   - Remote Destination Profile Configuration
   - Remote Destination Configuration
   - Mobility Softkey Template Configuration
5. Route Pattern Configuration
6. Translation Pattern Configuration
7. MGCP Gateway Configuration
8. Unity Connection Voice Mail Configuration
SIP Trunk Configuration

SIP Trunk Configuration to Microsoft Lync 2010 Standard Edition Server

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

Add a SIP Trunk connecting to the Microsoft Mediation Server. Configure and assign the Media Resource Group List to the SIP trunk to ensure MTP Software and Conferencing resources are available.
For “Inbound Calls” select “All” in 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.

All number patterns are accepted 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.

**Set SIP Trunk Security Profile to TCP on “Outgoing Transport Type” since Lync Mediation Server does not support UDP for SIP messages.**

**IP Address of Microsoft Mediation Server.**

**Keep SIP Profile set to Default for Delayed Offer and this needs to be changed to EO and PRACK supported only in case of PRACK testing.**
Media Resources Group Configuration

From the “Cisco Unified CM Administration” page, click “Media Resources” and choose “Media Resources Group” to create a new group.

The above shows the MRG configuration with MTP.

Choose all needed resources.
The above shows the MRG configuration with no MTP.

Click “Media Resources” and choose “Media Resources Group List” to create a new list and assign the new group to it.

Media Resources Group List
The above shows the MRGL configuration with media resource group that has MTP.

Assign corresponding Media Resource Group to the created Media Resource Group List.
The above shows the MRGL configuration with media resource group that has no MTP.

Assign corresponding Media Resource Group to the created Media Resource Group List.
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.

<table>
<thead>
<tr>
<th>Device Name/Line</th>
<th>Device Pool</th>
<th>Device Protocol</th>
<th>Status</th>
<th>IP Address 1/169</th>
</tr>
</thead>
<tbody>
<tr>
<td>+14085563001</td>
<td>Default</td>
<td>SCCP</td>
<td>Registered with 172.20.201.102</td>
<td></td>
</tr>
<tr>
<td>+14085563003</td>
<td>Default</td>
<td>SIP</td>
<td>Registered with 172.20.201.103</td>
<td></td>
</tr>
<tr>
<td>+14085563002</td>
<td>Default</td>
<td>SIP</td>
<td>Registered with 172.20.201.169</td>
<td></td>
</tr>
<tr>
<td>+14085563005</td>
<td>Default</td>
<td>SIP</td>
<td>Registered with 172.20.201.169</td>
<td></td>
</tr>
<tr>
<td>+14085563004</td>
<td>Default</td>
<td>SCCP</td>
<td>Registered with 172.20.201.169</td>
<td></td>
</tr>
</tbody>
</table>
7970 SIP Phone

Ensure fields in red rectangles below are configured.
<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>PC Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>Settings Access</td>
<td>Enabled</td>
</tr>
<tr>
<td>Gratiuous Alarms</td>
<td>Disabled</td>
</tr>
<tr>
<td>PC Voice VLAN Access</td>
<td>Enabled</td>
</tr>
<tr>
<td>Auto LBE Selection</td>
<td>Disabled</td>
</tr>
<tr>
<td>Web Access</td>
<td>Disabled</td>
</tr>
<tr>
<td>Days Display Not Active</td>
<td>Sunday</td>
</tr>
<tr>
<td>Display On Time</td>
<td>07:30</td>
</tr>
<tr>
<td>Display On Duration</td>
<td>10:30</td>
</tr>
<tr>
<td>Display Idle Timeout</td>
<td>01:00</td>
</tr>
<tr>
<td>Dial to PC Port</td>
<td></td>
</tr>
<tr>
<td>Lagging Display</td>
<td>PC Controlled</td>
</tr>
<tr>
<td>Load Server</td>
<td></td>
</tr>
<tr>
<td>Recording Type</td>
<td>Disabled</td>
</tr>
<tr>
<td>Recording Tone Local Volume</td>
<td>110</td>
</tr>
<tr>
<td>Recording Tone Remote Volume</td>
<td>90</td>
</tr>
<tr>
<td>Recording Tone Duration</td>
<td></td>
</tr>
<tr>
<td>Display On When Incoming Call</td>
<td>Disabled</td>
</tr>
<tr>
<td>Ringer</td>
<td>Disabled</td>
</tr>
<tr>
<td>&quot;more&quot; Soft Key Timer</td>
<td>$</td>
</tr>
<tr>
<td>Auto Call Select</td>
<td>Enabled</td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
</tr>
<tr>
<td>Advertise G.722 Codes</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Wideband Headset UI Control</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Handset UI Control</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Handset</td>
<td>Use Phone Default</td>
</tr>
<tr>
<td>Text Recognition Engine</td>
<td></td>
</tr>
</tbody>
</table>
On the “Phone Configuration” page under “Association Information”, click “Line [1] – Add a new DN” to enter the “Directory Number Configuration”. Make sure voice mail is set to system default with this option, otherwise choose the corresponding voice mail system used from the drop down menu.
### Directory Number Configuration

**No Answer Ring Duration (seconds):**
- Default: None

**Call Pickup Group:**
- Default: None

#### Park Monitoring

<table>
<thead>
<tr>
<th>Park Monitoring Feature</th>
<th>Value</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Forward No Retrieve</td>
<td>or</td>
<td>or</td>
<td>or</td>
</tr>
<tr>
<td>Destination Internal</td>
<td>or</td>
<td>or</td>
<td>or</td>
</tr>
<tr>
<td>Call Forward No Retrieve</td>
<td>or</td>
<td>or</td>
<td>or</td>
</tr>
<tr>
<td>Destination External</td>
<td>or</td>
<td>or</td>
<td>or</td>
</tr>
<tr>
<td>Park Monitoring Reversion Timer</td>
<td>or</td>
<td>or</td>
<td>or</td>
</tr>
</tbody>
</table>

A blank value means to call the parker's line.

#### H.323 Alternate Party Settings

**Target (Destination):**
- Default: None

**H.323 Calling Search Space:**
- Default: None

**H.323 No Answer Ring Duration (seconds):**
- Default: None

#### Line Settings for All Devices

- **Hold Reversion Ring Duration (seconds):** Default: 0
- **Hold Reversion Notification Interval (seconds):** Default: 0
- **Party Entrance Tone:** Default: None

#### Line 1 on Device SEP0000E830C1543

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display (Internal Caller ID)</td>
<td>Casio02</td>
</tr>
<tr>
<td>ASCII Display (Internal Caller ID)</td>
<td>Casio03</td>
</tr>
<tr>
<td>Line Text Label</td>
<td>Casio03</td>
</tr>
<tr>
<td>ASCII Line Text Label</td>
<td>Casio03</td>
</tr>
<tr>
<td>External Phone Number Text</td>
<td>Casio03</td>
</tr>
</tbody>
</table>

Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
Simultaneous Ring Feature (Unified Mobility) Configuration

**End Users Configuration**

From the “Cisco Unified CM Administration” page, go to “User Management” then click “End User”.

Add corresponding users. Shown below are the end users configured for this setup.
As part of the configuration for simultaneous ring, provision remote destination information for end user.

Make sure to enable both for simultaneous ring configuration.

When the remote destination profile for this end user is created (see next section), it shows up here.
Remote Destination Profile Configuration

As part of the configuration for simultaneous ring, provision remote destination profile (RDP).

From the “Cisco Unified CM Administration” page, choose “Device” → “Device Settings” → “Remote Destination Profile”.

For each simultaneous ring enabled end user, a profile needs to be created to be used for the remote destination. The red rectangles below highlight the fields to configure and/or check.

Add the corresponding line information that will be associated with this remote destination first (see next section).

Make sure to add the corresponding end user to the “User ID”.

Add the remote destination after adding the line information.
After the remote destination profile is filled out and before adding the new remote destination, click “Line [1] – Add a new DN” under “Association Information” and complete before proceeding. The red rectangles below highlight the fields to configure and/or check.

When the corresponding remote destination is created (see next section), it shows up here with the corresponding end user.
Remote Destination Configuration

As part of the configuration for simultaneous ring, provision remote destination (RD).

This can be done from the “Remote Destination Profile Configuration” page or from the “Cisco Unified CM Administration” page by clicking “Device” then choosing “Remote Destination” then “Add New”.

Make sure “Line Association” and “Enable Mobile Connect” boxes are checked for simultaneous ring configuration.

Make sure “Mobile Phone” box is checked for Desk Phone to be able to switch call back to Microsoft Lync (or “send call to mobile” feature).

When the remote destination is configured to forward the call, its “Answer too Late Timer” may need to be tweaked to allow the forwarded call more time to ring at the new user’s IP Phone and its remote destination so they have sufficient time to answer.
Note that the remote destination number has to be different than the line DN of the IP phone in the association information.

Make sure to set this to the corresponding remote destination profile.
Mobility Softkey Template Configuration

As part of the configuration for simultaneous ring, create and configure Mobility Softkey template for “send call to mobile” feature.

From the “Cisco Unified CM Administration” page, click “Device” → “Device Settings” → “Softkey Template”

Add New → Select Standard User from Menu → Copy → Change Name accordingly → Save.

In the upper right-hand side, click “Go” next to “Related Links: Configure Soft Key Layout” to complete configuration.
Select the “On Hook” and “Connected” states, then add “Mobility (Mobility)” to the “Selected Softkeys”. Save the template.
From the “Cisco Unified CM Administration” page, click “Device” then choose “Phone”. For the IP Phone associated with a remote destination make sure to add the softkey template that includes “Mobility” on the “Phone Configuration” page. Also to complete the simultaneous ring configuration, add the end user with the corresponding remote destination profile to the “Owner User ID” field.

On the phone configuration page, make sure to add the created softkey template that includes “Mobility”. 
When the configuration is done, the corresponding remote destination profile and remote destination with end user will show up on the line information page of the IP Phone.

On the phone configuration page, to complete the simultaneous ring configuration, add the end user with the corresponding remote destination profile to the “Owner User ID”.
Corresponding remote destination profile.

Corresponding remote destination with end user.
Route Pattern Configuration

From the “Cisco Unified CM Administration” page, choose “Call Routing” → “Route Hunt” → “Route Pattern”.

Route pattern setup to route calls from Cisco UCM to remote destinations on Microsoft Lync Server 2010.

Remote destination route pattern.

SIP trunk to Microsoft Mediation Server.
SIP Route Pattern for REFER testing

Prefix with a plus sign (+) to match the directory numbers (DN’s) on the LYNC R2 side, in case the normalization rules in the Microsoft LYNC environment do not already add the plus sign.
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 calls from Cisco UCM to Avaya PBX.
Translation Pattern Configuration

From the “Cisco Unified CM Administration” page, click “Call Routing” then choose “Translation Pattern”.

This pattern translates 4-digit extensions to the E.164 DN’s of the IP phones.
MGCP Gateway Configuration

From the “Cisco Unified CM Administration” page, click “Device” then choose “Gateway”.

![Image of Cisco Unified CM Administration interface with MGCP Gateway configuration details]
MGCP T1 Port used for calls to/from PBX.
Unity Connection Voice Mail Configuration

To configure voice mail, perform the initial configuration steps, then set up a voice mail hunt list:

From the “Cisco Unified CM Administration” page, click “Voice Mail” → run “Cisco Voice Mail Port Wizard” to setup voice mail.

From the “Cisco Unified CM Administration” page, click “Call Routing” → “Route/Hunt” → “Hunt List”

Hunt List for voice mail configuration
From the “Cisco Unified CM Administration” page, click “Call Routing” → “Route/Hunt” → Line Group Line Group for voice mail configuration.
From the “Cisco Unified CM Administration” page, click “Call Routing” → “Route/Hunt” → Hunt Pilot

Hunt Pilot Number for voice mail configuration.
From the “Cisco Unified CM Administration” page, go to “Voice Mail” → “Message Waiting” MWI ON and OFF DNs.
From the “Cisco Unified CM Administration” page, choose “Voice Mail” → “Voice Mail Pilot”. The voice mail Pilot Number should be the same as the voice mail Hunt Pilot Number.
In the “Voice Mail Profile”, assign the voice mail Pilot Number into Default Profile:

On the Phone’s “Directory Number Configuration” page, set the “Voice Mail Profile” accordingly (in this case None or Default)
Cisco UCM Normalization Script

The following normalization script is used to strip off the TIAS field inside the SDP towards the Microsoft Lync 2010.

```
N = 1
local function remove_tias(msg)
    local sidb = msg:getSIdB()
    if sidb
        local n = 0
        sidb = str:replace("b=TIAS\(""", "")
        if n > 0
            -- The substitution occurred and therefore the SDP changed
            msg:writeSidb(sidb)
        end
    end
end
```
M = {}

local function remove_tias(msg)

    local sdp = msg:getSdp()

    if sdp

        then

            local n = 0

            sdp, n = sdp:gsub("b=TIAS[^v\r\n]+v?\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
<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Call Pickup)</th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Hot Call Pickup Enabled *</td>
<td>False</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Hot Call Pickup Location Time *</td>
<td>12</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>Hot Call Pickup No Answer Time *</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Replaces)</th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Block Office To Office Features *</td>
<td>False</td>
<td>True</td>
<td>False</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Redirection [SIP])</th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Redirect From No Answer/Abandoned Time *</td>
<td>24</td>
<td>24</td>
<td></td>
</tr>
<tr>
<td>Maximum Rejection Count *</td>
<td>70</td>
<td>70</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Multilevel Precedence and Preemption)</th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Execute Override Call Preemptable *</td>
<td>False</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Execute Override Call Preemptable *</td>
<td>False</td>
<td>True</td>
<td>False</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Path Replacement)</th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Path Replacement Enabled *</td>
<td>False</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Path Replacement on Troubled Calls *</td>
<td>True</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Path Replacement Minimum Delay Time *</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Path Replacement Maximum Delay Time *</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Path Replacement P2 Time *</td>
<td>30</td>
<td>30</td>
<td>30</td>
</tr>
<tr>
<td>Path Replacement P2 Time *</td>
<td>15</td>
<td>15</td>
<td>15</td>
</tr>
</tbody>
</table>
Configuring the MGCP Gateway

Highlighted in Bold are the relevant configuration steps for this setup.

MS_GW1#sh run

Building configuration...

Current configuration : 8817 bytes

!

version 12.4

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname MS_GW1

!

boot-start-marker

boot system slot1: c3825-ipvoicek9-mz.124-22.T3.bin

boot-end-marker

!

card type t1 0 1

logging message-counter syslog

logging buffered 99999

no logging console

enable password cisco

!

no aaa new-model

clock timezone pacific -8

network-clock-participate wic 1
network-clock-select 1 T1 0/1/0

! dot11 syslog
ip source-route
ip cef
!
!
!
!
!
no ip domain lookup

ip host CM_Administrator 172.20.201.254
no ipv6 cef
multilink bundle-name authenticated
!
!
!
!
!
isdn switch-type primary-ni
!
voice-card 0
dspfarm
dsp services dspfarm
!
!
!
!
voice service voip
allow-connections sip to sip
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
sip
min-se 90

registrar server expires max 3600 min 120
!
archive
log config
hidekeys
!
!
controller T1 0/1/0

cablelength long 0db

pri-group timeslots 1-24 service mgcp
!
controller T1 0/1/1

cablelength long 0db
!
!
!
!
!
!
!
!
!
!
interface GigabitEthernet0/0

ip address 172.20.201.165 255.255.255.0
duplex auto
speed auto
media-type rj45
!
interface GigabitEthernet0/1

no ip address
shutdown
duplex auto
speed auto
media-type rj45
!
interface Serial0/1/0:23
no ip address
encapsulation hdlc
isdn switch-type primary-ni
isdn incoming-voice voice
isdn bind-l3 ccm-manager
no cdp enable
!
ip default-gateway 172.20.201.1
ip forward-protocol mld
ip route 0.0.0.0 0.0.0.0 172.20.5.1
!
nocm ip http server
no ip http secure-server
!!
control-plane
!
!
!
voice-port 0/1/0:23
!
ccm-manager mgcp
ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 172.20.201.254
ccm-manager config

mgcp
cmgp call-agent 172.20.201.254 2427 service-type mgcp version 0.1
cmgp dtmf-relay voip codec all mode nte-ca
cmgp rtp unreachable timeout 1000 action notify
cmgp modem passthrough voip mode nse
cmgp package-capability rtp-package
cmgp package-capability sst-package
cmgp package-capability pre-package
cmgp package-capability fm-package
no mgcp package-capability res-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
cmgp sdp simple
cmgp fax t38 ecm
cmgp fax t38 inhibit
cmgp rtp payload-type g726r16 static
cmgp bind control source-interface GigabitEthernet0/0
cmgp bind media source-interface GigabitEthernet0/0
cmgp behavior g729-variants static-pt
!
cmgp profile default
!
codec g711ulaw
!
dial-peer voice 999001 pots
  service mgcp
  port 0/1:0:23
!
!
!
sip-ua
!
!
line con 0
line aux 0
line vty 0 4
password cisco
login
!

exception data-corruption buffer truncate
scheduler associate 20000 1000
end
## Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>CFA</td>
<td>Call Forwarding Always</td>
</tr>
<tr>
<td>CFB</td>
<td>Call Forwarding Busy</td>
</tr>
<tr>
<td>CFNR</td>
<td>Call Forwarding No Reply</td>
</tr>
<tr>
<td>DN</td>
<td>Directory Number</td>
</tr>
<tr>
<td>DND</td>
<td>Do-Not-Disturb</td>
</tr>
<tr>
<td>DNS</td>
<td>Domain Name System</td>
</tr>
<tr>
<td>DTMF</td>
<td>Dual Tone Multi Frequency</td>
</tr>
<tr>
<td>DTP</td>
<td>Desk Top Phone</td>
</tr>
<tr>
<td>MCS</td>
<td>Multimedia Communication Server</td>
</tr>
<tr>
<td>MWI</td>
<td>Message Waiting Indicator</td>
</tr>
<tr>
<td>LYNC</td>
<td>Lync 2010 Server</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>TCP</td>
<td>Transmission Control Protocol</td>
</tr>
<tr>
<td>TLS</td>
<td>Transport Layer Security</td>
</tr>
<tr>
<td>XML</td>
<td>Extensible Markup Language</td>
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<tr>
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<td>Quality of Service</td>
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<td>GW</td>
<td>Gateway</td>
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<td>S/W</td>
<td>Software</td>
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<td>DB</td>
<td>Database</td>
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</tbody>
</table>
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