Microsoft Lync 2010 with Microsoft Mediation Server via Cisco Unified Border Element (Enterprise Edition) 8.7 using SIP to Cisco Unified Communications Manager 8.5.1
July 26, 2011 – Initial Version

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Introduction

- This application note shows basic call interoperability and documented steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) 8.5.1 with Cisco Unified Border Element (Cisco UBE) 8.7 to interoperate with Microsoft Lync Server using Microsoft Mediation Server. The integration uses SIP trunk between Cisco Unified Border Element and Microsoft Mediation Server. The basic call features include calls between Lync 2010 (formerly Office Communicator) and Cisco IP phone (SIP and Skinny) in both directions, attended call transfer, unattended call transfer, call forward (all, busy and no answer), three-way conference, DTMF, hold/resume, caller ID functionality and single-number reach using Cisco UCM mobile connect feature. This application note aims to provide a good understanding of what works and what does not work in terms of the feature interaction between various components of the Cisco UCM /Cisco UBE and Microsoft Lync /Mediation server. It also provides guidance to deployment participants of the limitations, expected behaviors as well as 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.

- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability between Cisco UCM/ Cisco UBE connected to Microsoft Lync /Microsoft Mediation Server via SIP protocol.

- This Application Note uses the c3845 IOS-voice-gateway for Cisco UBE functionality, however other Cisco voice gateways are an option to use since Cisco UBE implementation does not depend on the platform. Here is a list of Cisco Products capable of Cisco UBE functionality:
  - Cisco 3900 Series Integrated Services Routers
  - Cisco 2900 Series Integrated Services Routers
  - Cisco 2800 Series Integrated Services Routers
  - Cisco 3800 Series Integrated Services Routers
  - Cisco AS5350XM Universal Gateway
  - Cisco AS5400XM Universal Gateway

It is important to analyze and research your deployment needs to ensure you acquire the correct Cisco platform based on features required, number of users and future growth planning.
Network Topology

Figure 1. Basic Call Setup

Note: The Cisco UBE was configured for media “flow-through” mode.
Limitations

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

- Calling and connected name are now supported with Microsoft Lync however in many scenario’s like call-forward and call-transfer the calling and connected names are not updated.

- When a Cisco Unified IP phone is performing a call-forward toward a Lync 2010 client the INVITE message from Cisco UBE to Microsoft mediation server must not contain a SIP Diversion header. The SIP Diversion header is not supported by the Microsoft mediation server causing forwarded calls to fail. You must use Cisco UBE sip profiles feature to remove the Diversion header from the Cisco UCM INVITE before handing-off the INVITE to the Microsoft mediation server. See configuration section for details.

- 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.

- Lync 2010 client does not support call-forward on busy. As an option the Lync application 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 a Lync user initiates a conference (Lync is the conference leader) the call will remain up even after all Lync users have dropped from the conference and only one Cisco Unified IP phone remains connected. Microsoft never sends a BYE message toward the Cisco UBE even after all Lync users have left the conference, the Microsoft Lync relies on Cisco UBE/Cisco UCM to send the BYE for all media to be terminated.

- Microsoft mediation server only supports TCP transport to carry SIP messages. If UDP is required Cisco UBE can be used to perform the conversion.

- Microsoft mediation server only supports G711 ulaw or alaw on the outside interface. If G729 is required Cisco UBE can perform the conversion.

- When a call is ringing to a Lync user, the caller (either on an Avaya station or a PSTN line routed through the PBX) will not get a ring-back tone. Avaya has resolved this issue with the 5.2.1 SP1 software release.

- 15 minutes into the call, CUBE sends a RTCP BYE message to Lync, which causes loss of audio at the Lync client end for 20 seconds and then audio resumes. Cisco CDETS CSCtr54269 has been filed for the problem.

Comments

- Some Microsoft Lync Client endpoints require RTCP packets. If RTCP packets are not generated by all endpoints, additional features of the Unified Border Element can be used to resolve this issue. Contact your Cisco sales engineer for information.
System Components

Hardware Requirements

The following hardware is required:

Cisco equipment

- Cisco Unified Border Element is an integrated Cisco IOS Software application that runs on various IOS platforms, follow the link for more details: http://www.cisco.com/go/cube
- Packet Voice Data Module (PVDM). You will need to install DSP modules (PVDM) on Cisco UBE 8.7 if you require MTP, Transcoding or Conference Bridge resources for codecs other than G.711. DSP are not required for basic calls. Follow the link for system required DSP calculator. http://www.cisco.com/cgi-bin/Support/DSP/cisco_dsp_calc.pl
- Cisco Catalyst 3500 - switches
- Cisco Unified IP Phones (The topology diagram shows 7970, 7971 and 7960, but any Cisco IP phone model supporting RFC2833 can be used)
- Cisco MCS 7825H servers

Microsoft equipment

- Cisco MCS 7825 server (Lync front-end server with Co-located mediation server)
- Cisco MCS 7825 server (Active Directory server and Domain Controller)
- 2 x Laptops with Windows 7 Ultimate N versions (for Lync Client)

Software Requirements

The following software is required:

- Cisco IOS 151-4.M
- Microsoft Lync Client
- The documented Cisco UBE configuration can be supported with the following IOS feature sets: IP VOICE, SP SERVICES, ADVANCED IP SERVICES, ADVANCED ENTERPRISE SERVICES, INT VOICE/VIDEO, IPIP GW, TDMP GW, INT VOICE/VIDEO, IP1PGW, TDMP GW AES
- Cisco Unified Communications Manager Release 8.5(1)
Features

This section lists supported and unsupported features.

Features Supported

- Basic call G711 u-law and A-law
- Calling number
- Call Transfer attended and unattended
- Three-way Conference call
- Call hold and resume
- Call Forward All, Call Forward No Reply and Call Forward Busy (CFB is not supported by Microsoft Lync.)
- DTMF (RFC2833)
- Single-number reach (SNR) for Cisco Unified IP phone and Microsoft Lync (tested using Cisco UCM mobile connect feature).
- Transcoding of G711 to G729
- Conversion of transport protocol udp-to-tcp and tcp-to-udp

Features Not Supported

- Calling Name not supported by the Microsoft platform in various scenarios
- Connected Name not supported by the Microsoft platform in various scenarios
- Connected Number not supported by the Microsoft platform in various scenarios
Configuration

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

Configuring the Microsoft Lync Server Standard Edition

1. Configuring Domain Name Server.
   - Forward Lookup Zone
   - Reverse Lookup Zone
2. Configuration of Front End Server
3. Configuring User - General
4. Configuring User – Account
5. Active Directory User configuration
6. Configuring Users - Pool
7. Mediation Server configuration
8. Voice Routing Configuration
This screenshot shows the DNS entries. This highlights the Lync pool in lync2010rtm.com domain. Host A record added for Cisco UCM, CUBE, Lync Server and Active Directory.
Reverse Lookup zone

This screenshot shows PTR record entry for reverse lookup.
This screenshot shows Front End Server settings. Configuration details are found in the following pages.
This screenshot shows how to add/edit Front End server properties by right clicking on applicable pool. Here the CUBE is selected as default PSTN gateway.
This screenshots shows the Lync Topology from the Lync Server Control Panel Configuration of Front End Server (Page 4 of 5)
CUBE added as a PSTN gateway.
Mediation Pools are used for failover where the voice calls route through the second mediation server when the first one goes down.
Configuring Users - General
Configuring User – Account
This screen shot shows how to add or configure a new user, which is done in the Active Directory. This is done by right clicking on USERS. Users added here will be displayed in the pool.
These settings were selected for lab use. The administrator installing and managing should set these values as per his/her administrative requirements.
Configuring Users – Pool (Page 1 of 2)
The screen shot shows the list of Pool users when viewing from the Control Panel.
The Mediation Server has two Ethernet Interfaces, one listens to the Microsoft Lync Server (172.20.117.152) and the other to the Cisco UBE (172.20.85.101)
Mediation Server Configuration (Page 2 of 3)
The next hop PSTN Gateway connected to the Mediation Server is the Cisco UBE.
Number patterns to be routed through this Mediation Server.
Calling features are selected in Voice policy tab.
Local route added in the Route tab.
PSTN Usage route added.
Voice Routing Configuration (Page 5 of 5)

The phone pattern expression and translation expression allows the Lync user to dial a 4-digit number (3001) DN and/or a 10-digit 14085663001 E.164 number, either pattern will be passed to the mediation server which will then pass to CUBE. Using CUBE you can manipulate digits to reach the proper destination. Microsoft Lync has the capability of manipulating digits as well, but this document does not cover those configurations.
Microsoft Lync 2010 Configuration (Page 1 of 7)

Navigation: Choose Tools → Options and enter the sign-in information.
Note: Click Advanced button to select the Advanced Connection Settings.
Note: If there are DNS entries for this Microsoft Lync, automatic configuration can be used if not select manual configuration.
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.
Microsoft Lync 2010 Configuration (Page 7 of 7)

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

Cisco Unified Communications Manager System version

Cisco Unified CM Administration

System version: 8.5.1.10000-26

Last Successful Login: Mar 2, 2011 11:19:34 PM

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

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

For Cisco Technical Support please visit our Technical Support web site.
## SIP trunk configuration

### SIP trunk main page

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Ceiling Search Space</th>
<th>Ceiling Pool</th>
<th>Route Pattern</th>
<th>Partition</th>
<th>Route Group</th>
<th>Priority</th>
<th>Trunk Type</th>
<th>SIP Trunk Security Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>1035_CUE</td>
<td>SIP trunk to cube</td>
<td>MIST_Lynx</td>
<td>Default</td>
<td>15XX</td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>1045_CUE</td>
<td>SIP trunk to cube</td>
<td>MIST_Lynx</td>
<td>Default</td>
<td>1413221106XX</td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>Lync2010-0005EV</td>
<td>SIP Trunk To Lync Mediation Server</td>
<td>Default</td>
<td></td>
<td><a href="mailto:lync2010@Lync.com">lync2010@Lync.com</a></td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>Lync2010-0005EV</td>
<td>SIP Trunk To Lync Mediation Server</td>
<td>Default</td>
<td></td>
<td><a href="mailto:lync2010@Lync.com">lync2010@Lync.com</a></td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>Lync2010-0005EV</td>
<td>SIP Trunk To Lync Mediation Server</td>
<td>Default</td>
<td></td>
<td><a href="mailto:lync2010@Lync.com">lync2010@Lync.com</a></td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>OCS_SIP Trunk</td>
<td>SIP trunk to OCS Mediation Server</td>
<td>Default</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
</tbody>
</table>
The SIP Trunk should be given a unique name and make sure the Device Pool settings are set as per your deployment requirements.
The Calling Search Space is assigned here for incoming calls. Also by enabling the Redirecting Diversion Header Cisco UBE will convert Diversion Header to History-Info header.
The Cisco UBE IP address and Cisco UBE signaling port number are assigned here.
### Region configuration

#### Region main page

<table>
<thead>
<tr>
<th>Name</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>G.728</td>
<td></td>
</tr>
<tr>
<td>phonea711</td>
<td></td>
</tr>
<tr>
<td>phonea729</td>
<td></td>
</tr>
</tbody>
</table>
Region 'Default' configuration

Region 'phonesg711' is assigned to Cisco phones using the device pool of the same name. Notice the codec relationship to Default is g711 which should not run on other g711 will be supported but not g711 or any codec with a lower bandwidth requirement will be supported (e.g. g729).

**Region Relationships**

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>phonosg711</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>phonosg729</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

**Modify Relationship to other Regions**

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td></td>
</tr>
<tr>
<td>phonosg711</td>
<td>G.711</td>
<td>Keep Current Setting</td>
<td></td>
</tr>
<tr>
<td>phonosg729</td>
<td>G.729</td>
<td>Keep Current Setting</td>
<td></td>
</tr>
</tbody>
</table>

- * indicates required item.
- **The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 kbps between regions and can be used interchangeably.**
### Region ‘phonesg711’ configuration

![Cisco Unified CM Administration](Image)

#### Region Configuration

![Region Configuration](Image)

---

**Find and List Regions Information**

**Name**

- **phonesg711**

---

**Region Relationships**

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>G.729</td>
<td>711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>phonesg711</td>
<td>711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>phonesg729</td>
<td>711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

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

---

**Modify Relationship to other Regions**

![Modify Relationship to other Regions](Image)

---

- **Save**
- **Delete**
- **Reset**
- **Apply Config**
- **Add New**

---

- **NOTE:** Region 'phonesg711' configuration does not display any relationships.

---

**The Audio Codec selection determines bandwidth only. The G.711 and G.712 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.**

---

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## Device Pool

### Device Pool – Main page

The image shows a page from Cisco Unified CM Administration software, displaying the device pool settings. The interface includes a table with columns for **Name**, **Cisco Unified CM Group**, **Region**, **Date/Time Group**, and **Copy**. The table lists four device pools: `Default`, `SIP trunk`, `phones/11`, and `phones/29`, each with different settings in the respective columns. The interface also includes options for adding, selecting, clearing, and deleting device pools.
**Device Pool – Default**

<table>
<thead>
<tr>
<th>Device Pool Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
</table>

---

### Device Pool Information

Device Pool: **Default** (12 members***)

---

### Device Pool Settings

- **Device Pool Name**: Default
- **Cisco Unified Communications Manager Group**: Default
- **Calling Search Space for Auto-registration**: `< None >`
- **Reverted Call Focus Priority**: Default
- **Local Route Group**: `< None >`

---

### Roaming Sensitive Settings

- **Date/Time Group**: CM/Local
- **Region**: Default
- **Media Resource Group List**: `< None >`
- **Location**: `< None >`
- **Network Locale**: `< None >`
- **SIP Extension**: Disable
- **Connection Monitor Duration**: ***
- **Single Button Barge**: Default
- **Join Across Lines**: Default
- **Physical Location**: `< None >`
- **Device Mobility Group**: `< None >`
### Device Mobility Related Information

- **Device Mobility Calling Search Space**: <None>
- **AAR Calling Search Space**: <None>
- **AAR Group**: <None>
- **Calling Party Transformation CSS**: <None>
- **Called Party Transformation CSS**: <None>

### Geolocation Configuration

- **Geolocation**: <None>
- **Geolocation Filter**: <None>

### Incoming Calling Party Settings

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

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digit</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td>&lt;None&gt;</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>&lt;None&gt;</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>&lt;None&gt;</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>&lt;None&gt;</td>
<td>&lt;None&gt;</td>
</tr>
</tbody>
</table>

- **Save**  - Delete  - Copy  - Reset  - Apply Config  - Add New

* indicates required item.

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

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

****These five parameters will overwrite device level settings when device is roaming and in the same device mobility group.
### Device Pool – phonesg711

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

#### Device Pool Configuration

- **Status**
  - Status: Ready

#### Device Pool Information
- Device Pool: phonesg711 (4 members***)

#### Device Pool Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Pool Name*</td>
<td>phonesg711</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager Group*</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Search Space for Auto-registration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Reverted Call Flow Priority*</td>
<td>Default</td>
</tr>
<tr>
<td>Local Route Group</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

#### Roaming Sensitive Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Region*</td>
<td>phonesg711</td>
</tr>
<tr>
<td>Media Resource Group List*</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location*</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Network Locale*</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SRST Reference*</td>
<td>Use Default Gateway</td>
</tr>
<tr>
<td>Connection Monitor Duration***</td>
<td></td>
</tr>
<tr>
<td>Single Button Range*</td>
<td>Default</td>
</tr>
<tr>
<td>Join Across Lines*</td>
<td>Default</td>
</tr>
<tr>
<td>Physical Location*</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Device Mobility Group*</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
### Device Mobility Related Information

<table>
<thead>
<tr>
<th>Device Mobility Calling Search Space</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Geolocation Configuration

<table>
<thead>
<tr>
<th>Geolocation</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geolocation Filter</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Incoming Calling Party Settings

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

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

* indicates required item.
**Number of devices that have to be reset when this device pool is updated. To see a detailed list of those devices and other dependencies, click on Dependency Records.
***Leave the field blank or enter -1 to use the configuration from the enterprise parameter.
****These five parameters will overwrite device level settings when device is roaming and in the same device mobility group.
Media resource group assigned in Cisco SIP trunk

<table>
<thead>
<tr>
<th>Media Resource Group Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
<tbody>
<tr>
<td>Status</td>
<td></td>
</tr>
<tr>
<td>Status: Ready</td>
<td></td>
</tr>
</tbody>
</table>

**Media Resource Group Status**
Media Resource Group: MRC_SN_HTP (used by 7 devices)

**Media Resource Group Information**

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

**Devices for this Group**

Available Media Resources

Selected Media Resources

- ANN_2
- MTP_2
- MTP_1
- MDH_2 (MGCP)
- MDH_2 (MGCP)

Use Multi-cost for MCH Audio (if at least one multi-cost MCH resource is available)

---

* indicates required field.

**Includes Announciators (ANN), Conference Bridges (CB), Media Termination Points (MTP), Music On Hold Servers (MCH) and Transcoders (XCODE)
Media resource list assigned in Cisco SIP trunk

<table>
<thead>
<tr>
<th>Status</th>
<th>Media Resource Group List Status</th>
<th>Media Resource Group List Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Status: Ready</td>
<td>Media Resource Group List: MRGL_SW_HTTP (used by 7 devices)</td>
<td>Name: MRGL_SW_HTTP</td>
</tr>
</tbody>
</table>

- **Available Media Resource Groups**
  - MRG for EXUM 2007 Fax
  - MRG for EXUM 2010
  - MRG_HTTP_MUP
  - MRG_SW_HTTP

- **Selected Media Resource Groups**
  - MRG_SW_HTTP

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Owner User ID is required if setting up SNR "mobile connect".
### Phone Configuration

<table>
<thead>
<tr>
<th>Geolocation</th>
<th>&lt;None&gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Reference</td>
<td>False</td>
</tr>
<tr>
<td>Ignore Presentation Indicators</td>
<td>Internal calls only</td>
</tr>
<tr>
<td>Allow Control of Device</td>
<td>From CTI</td>
</tr>
<tr>
<td>Logged into Hunt Group</td>
<td>False</td>
</tr>
<tr>
<td>Remote Device</td>
<td>False</td>
</tr>
<tr>
<td>Protected Device</td>
<td>False</td>
</tr>
</tbody>
</table>

#### Protocol Specific Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
<tr>
<td>Presence Group</td>
<td>Standard Presence group</td>
</tr>
<tr>
<td>SIP Call Rules</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>RTP Preferred Originating Codec</td>
<td>钛图</td>
</tr>
<tr>
<td>Device Security Profile</td>
<td>Cisco 7970 - Standard SIP Non-Secure Profile</td>
</tr>
<tr>
<td>Redirecting Calling Search Space</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Standard SIP Profile</td>
</tr>
<tr>
<td>Digest User</td>
<td>&lt;None&gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Media Termination Point</td>
<td>Required</td>
</tr>
<tr>
<td>Unattended Port</td>
<td>False</td>
</tr>
<tr>
<td>Require OTMF Reception</td>
<td>True</td>
</tr>
</tbody>
</table>

#### Certification Authority Proxy Function (CAPF) Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Certificate Operation</td>
<td>No Pending Operation</td>
</tr>
<tr>
<td>Authentication Mode</td>
<td>By Name</td>
</tr>
<tr>
<td>Authentication String</td>
<td>None</td>
</tr>
<tr>
<td>Key Size (Bits)</td>
<td>1024</td>
</tr>
<tr>
<td>Certificate Operation Status</td>
<td>None</td>
</tr>
</tbody>
</table>

Note: Security Profile Contains Additional CAPF Settings.

#### Expansion Module Information

<table>
<thead>
<tr>
<th>Module</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Module 1</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Module 1 Load Name</td>
<td>None</td>
</tr>
<tr>
<td>Module 2</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Module 2 Load Name</td>
<td>None</td>
</tr>
<tr>
<td>Internal Data Locations Information (Leave blank to use default)</td>
<td></td>
</tr>
<tr>
<td>---</td>
<td></td>
</tr>
<tr>
<td>Information</td>
<td></td>
</tr>
<tr>
<td>Directory</td>
<td></td>
</tr>
<tr>
<td>Messages</td>
<td></td>
</tr>
<tr>
<td>Services</td>
<td></td>
</tr>
<tr>
<td>Authentication Server</td>
<td></td>
</tr>
<tr>
<td>Proxy Server</td>
<td></td>
</tr>
<tr>
<td>Lda</td>
<td></td>
</tr>
<tr>
<td>Lda Timer (seconds)</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Extension Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Extension Mobility</td>
</tr>
<tr>
<td>Log Out Profile</td>
</tr>
<tr>
<td>Log In Time</td>
</tr>
<tr>
<td>Log Out Time</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>MLPP Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP Domain</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Do Not Disturb</th>
</tr>
</thead>
<tbody>
<tr>
<td>Do Not Disturb</td>
</tr>
<tr>
<td>DND Cannon</td>
</tr>
<tr>
<td>DND Incoming Call Alert</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Secure Shell Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Shell User</td>
</tr>
<tr>
<td>Secure Shell Password</td>
</tr>
<tr>
<td>Configuration Item</td>
</tr>
<tr>
<td>-----------------------------------</td>
</tr>
<tr>
<td>Enable Speakerphone</td>
</tr>
<tr>
<td>Enable Speakerphone and Headset</td>
</tr>
<tr>
<td>PC Port</td>
</tr>
<tr>
<td>Settings Access</td>
</tr>
<tr>
<td>Gratuitous ARP</td>
</tr>
<tr>
<td>PC Voice VLAN Access</td>
</tr>
<tr>
<td>Auto Line Select</td>
</tr>
<tr>
<td>WBD Access</td>
</tr>
<tr>
<td>Days Display Not Active</td>
</tr>
<tr>
<td>Display On Time</td>
</tr>
<tr>
<td>Display On Duration</td>
</tr>
<tr>
<td>Display Off Timeout</td>
</tr>
<tr>
<td>Span to PC Port</td>
</tr>
<tr>
<td>Logging Display</td>
</tr>
<tr>
<td>Load Server</td>
</tr>
<tr>
<td>Recording Tone</td>
</tr>
<tr>
<td>Recording Tone Local Volume</td>
</tr>
<tr>
<td>Recording Tone Remote Volume</td>
</tr>
<tr>
<td>Recording Tone Duration</td>
</tr>
<tr>
<td>Display on When Incoming Call</td>
</tr>
<tr>
<td>RTCP</td>
</tr>
<tr>
<td>&quot;Insert&quot; Soft Key Timer</td>
</tr>
<tr>
<td>Auto-Call Select</td>
</tr>
<tr>
<td>Log Server</td>
</tr>
<tr>
<td>Advertone G.722 Codec</td>
</tr>
<tr>
<td>Wideband Headset CC Control</td>
</tr>
<tr>
<td>Wideband Headset CAC</td>
</tr>
<tr>
<td>Wideband Headset</td>
</tr>
<tr>
<td>Wideband Handset</td>
</tr>
<tr>
<td>Peer Firmware Sharing</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port</td>
</tr>
<tr>
<td>LLDP Asset ID</td>
</tr>
<tr>
<td>LLDP Power Priority</td>
</tr>
<tr>
<td>IPv6 Log Server</td>
</tr>
<tr>
<td>IPv6 Authentication</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure</td>
</tr>
<tr>
<td>Minimum Ring Volume</td>
</tr>
<tr>
<td>Headset Sidetone Level</td>
</tr>
<tr>
<td>Whispering</td>
</tr>
</tbody>
</table>

---

* Indicates required item.

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

*** Note: Security Profile Contains Additional CDP Settings.

**** Note: A new Softkey template without supplementary service Softkeys must be used for a protected phone.
Cisco Unified SIP IP phone Directory Number configuration

Here Cisco03_RDP is the SNR device and 14152221013 is the remote destination number of SNR device for simultaneously ringing.

<table>
<thead>
<tr>
<th>Name</th>
<th>Destination Number</th>
<th>Owner</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco03_RDP</td>
<td>14152221013</td>
<td>User1</td>
</tr>
</tbody>
</table>
### Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Calling Search Space Activation Policy</th>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward All</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy Internal</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy External</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer Internal</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer External</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage External</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward on CTI Failure</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered Internal</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

No-Answer Ring Duration (seconds) <br>
Call Pickup Group < None >

### Park Monitoring

<table>
<thead>
<tr>
<th>Park Monitoring</th>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward No Retrieve Destination External</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Retrieve Destination Internal</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

Park Monitoring Inversion Timer <br>A blank value will use value set in Park Monitoring Inversion Timer service parameter

### MLPP Alternate Party Settings

<table>
<thead>
<tr>
<th>Target (Destination)</th>
<th>MLPP Calling Search Space</th>
<th>MLPP No Answer Ring Duration (seconds)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>
### Line 1 on Device SEP0000E839C1543

<table>
<thead>
<tr>
<th>Attribute</th>
<th>Value</th>
<th>Update Shared Device Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display (Internal Caller ID)</td>
<td>Cisco03</td>
<td></td>
</tr>
<tr>
<td>ASCII Display (Internal Caller ID)</td>
<td>Cisco03</td>
<td></td>
</tr>
<tr>
<td>Line Text Label</td>
<td>Cisco03</td>
<td></td>
</tr>
<tr>
<td>ASCII Line Text Label</td>
<td>Cisco03</td>
<td></td>
</tr>
<tr>
<td>External Phone Number Mask</td>
<td></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>Call Recording Disabled</td>
<td></td>
</tr>
<tr>
<td>Recording Profile</td>
<td>&lt;None&gt;</td>
<td></td>
</tr>
<tr>
<td>Monitoring Calling Search Spacing</td>
<td>&lt;None&gt;</td>
<td></td>
</tr>
<tr>
<td>Loop Missed Calls</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Multiple Call/Call Waiting Settings on Device SEP0000E839C1543

Note: The range to select the Max Number of calls is 1-30

<table>
<thead>
<tr>
<th>Attribute</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Number of Calls*</td>
<td></td>
</tr>
<tr>
<td>Busy Trigger*</td>
<td></td>
</tr>
</tbody>
</table>

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**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

**Directory Number Configuration**

- **Line Settings for All Devices**
  - Hold Reversion Ring Duration (seconds) 0 Setting the Hold Reversion Ring Duration to zero will disable the feature
  - Hold Reversion Notification Interval (seconds) 0 Setting the hold reversion notification interval to zero will disable the feature
  - Party Entrance Tone* Default

- **Line 1 on Device SEP000E839C1543**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Update Shared Device Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display (Internal Caller ID)</td>
<td>Cisco03</td>
<td></td>
</tr>
<tr>
<td>ASCII Display (Internal Caller ID)</td>
<td>Cisco03</td>
<td></td>
</tr>
<tr>
<td>Line Text Label</td>
<td>Cisco03</td>
<td></td>
</tr>
<tr>
<td>ASCII Line Text Label</td>
<td>Cisco03</td>
<td></td>
</tr>
<tr>
<td>External Phone Number</td>
<td>Cisco03</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>Call Recording Disabled</td>
<td></td>
</tr>
<tr>
<td>Recording Profile</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Monitor Calling Search Space</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Log Missed Calls</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- **Multiple Call/Call Waiting Settings on Device SEP000E839C1543**
  - Note: The range to select the max Number of calls is: 1-50
  - Maximum Number of Calls* 4
  - Busy Trigger* 2 (Less than or equal to Max. Calls)

- **Forwarded Call Information Display on Device SEP000E839C1543**
  - Caller Name
  - Caller Number
  - Redirected Number
  - Dialed Number

- **Users Associated with Line**

<table>
<thead>
<tr>
<th>User ID</th>
<th>Permission</th>
</tr>
</thead>
<tbody>
<tr>
<td>Lym02</td>
<td>1</td>
</tr>
</tbody>
</table>

- **Save | Delete | Reset | Apply Config | Add New**

* Indicates required item.
** Changes to Line or Directory Number settings require restart.

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End User configuration

SEP000E839C1543 is the SIP phone.
Make sure to enable mobility for SNR (mobile connect) to work.
Configuring Microsoft Lync 2010 number as remote destination for SNR (mobile connect)

Remote Destination

14152221013 is the telephone number to dial from outside to reach the Lync client.
Remote Destination Profile

Cisco03_RD is the remote destination created in the previous step.
Configuring Cisco Skinny IP phone

If configuring SNR you must assign the same Softkey template as in the SIP IP phone configuration example.
If configuring SNR you must configure end user (User04) and assign here.
**Extension Information**

<table>
<thead>
<tr>
<th>Enable Extension Mobility</th>
<th>-- Use Current Device Settings --</th>
</tr>
</thead>
<tbody>
<tr>
<td>Log In Time</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Log Out Time</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**MLPP Information**

<table>
<thead>
<tr>
<th>MLPP Domain</th>
<th>MLPP Indication</th>
<th>MLPP Preemption</th>
</tr>
</thead>
<tbody>
<tr>
<td>&lt; None &gt;</td>
<td>Default</td>
<td>Default</td>
</tr>
</tbody>
</table>

**Do Not Disturb**

<table>
<thead>
<tr>
<th>Do Not Disturb</th>
<th>DND Option</th>
<th>DND Incoming Call Alert</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Ringer Off</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**Secure Shell Information**

<table>
<thead>
<tr>
<th>Secure Shell User</th>
<th>Secure Shell Password</th>
</tr>
</thead>
</table>

**Product Specific Configuration Layout**

<table>
<thead>
<tr>
<th>Disable Speakerphone and Headset</th>
<th>Forwarding Delay</th>
<th>PC Port</th>
<th>Settings Access</th>
<th>Gratuitous ARP</th>
<th>PC Voice VLAN Access</th>
<th>Video Capabilities</th>
<th>Auto Line Select</th>
<th>Web Access</th>
<th>Days Display Not Active</th>
<th>Display On Time</th>
<th>Display On Duration</th>
<th>Display Idle Timeout</th>
<th>Span to PC Port</th>
<th>Logging Display</th>
<th>Load Server</th>
<th>Recording Tone</th>
<th>Recording Tone Local Volume</th>
<th>Recording Tone Remote Volume</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Disabled</td>
<td>Enabled</td>
<td>Enabled</td>
<td>Disabled</td>
<td>Enabled</td>
<td>Disabled</td>
<td>Disabled</td>
<td>Enabled</td>
<td>Sunday</td>
<td>02:30</td>
<td>10:30</td>
<td>01:00</td>
<td>Disabled</td>
<td>PC Controlled</td>
<td>Disabled</td>
<td>Disabled</td>
<td>100</td>
<td>50</td>
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</table>
## Recording Tone Duration

<table>
<thead>
<tr>
<th>Setting</th>
<th>Status</th>
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</thead>
<tbody>
<tr>
<td>Display On When Incoming Call</td>
<td>Disabled</td>
</tr>
<tr>
<td>RTCP</td>
<td>Disabled</td>
</tr>
<tr>
<td><em>more</em> Soft Key Timer</td>
<td>5</td>
</tr>
<tr>
<td>Auto Call Select</td>
<td>Enabled</td>
</tr>
<tr>
<td>Log Server</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Advertisement G.722 Codec</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset UI Control</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset Config</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset</td>
<td>Use Phone Default</td>
</tr>
<tr>
<td>Wideband Handset</td>
<td>Disabled</td>
</tr>
<tr>
<td>Peer Firmware Sharing</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP)</td>
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</tr>
<tr>
<td>Cisco Discovery Protocol (CDP)</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP-MED) Switch Port</td>
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<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port</td>
<td>Enabled</td>
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<tr>
<td>LLDP Asset ID</td>
<td>Unknown</td>
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<tr>
<td>IPv6 Load Server</td>
<td>User Controlled</td>
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<tr>
<td>IPv6 Log Server</td>
<td>Normal</td>
</tr>
<tr>
<td>802.1x Authentication</td>
<td>0-Silent</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure</td>
<td>Use Phone Default</td>
</tr>
<tr>
<td>Minimum Ring Volume</td>
<td>Enabled</td>
</tr>
<tr>
<td>Headset Sidetone Level</td>
<td>Enabled</td>
</tr>
<tr>
<td>Enable Dialing</td>
<td>Enabled</td>
</tr>
</tbody>
</table>

Add New

Packet Capture Mode and Packet Capture Duration.

Settings.

Inventory service Softkeys must be used for a protected phone.
Configuring Cisco Skinny IP phone DN

The DN example for skinny does not include SNR (remote destination) configuration, but configuration of a remote destination end device is configured the same as it is configured for a SIP IP phone.
Configuring IP phone softkey template for mobility (mobile connect)

Go to Device ➔ Device settings ➔ Softkey template, click on Standard User, then click copy. Rename then follow the steps below:

### Softkey Template Configuration

**Status**
- Status: Ready

**Softkey Template Information**
- **Name**: Mobility User
- **Description**: Mobility Softkey Template for CallManager only
- **Applications**:
  - Cisco CallManager

- **Default Softkey Template**: 

  * indicates required item.
Softkey Template Configuration

**Status**

Status: Ready

**Notes**

Use this window to specify the Softkeys and their relative order for any phone models that support downloadable Softkey templates. This window lists all the Softkeys in the system even though not all phones support a Softkey. For information about the supported Softkeys for a particular phone, refer to the administration guide for the phone. If you choose a Softkey that is not supported by the phone, the Softkey will not display on the phone even if you configure it in this list.

**Softkey Layout Configuration**

Softkey Template: Standard User-mobile

Select a call state to configure: **On Hook**

Unselected Softkeys:

- Conference List (ConfList)
- Direct Transfer (DrTfn)
- Group Pick Up (GrpPick)
- Hold (Hold)
- Immediate Divert (Divert)
- Join (Join)
- Meet Me (MeetMe)
- Other Pickup (OthPick)
- Pick Up (PickUp)
- Quality Report Tool (QRT)
- Remove Last Conference Party (RmLstCf)
- Select (Select)
- Toggle Do Not Disturb (DND)
- Undefined (Undefined)
- Hold/Incoming Call (WtMode)
- Hold (Hold)
- **New Call (NewCall)**
- Forward All (CfndAll)

Selected Softkeys (ordered by position):

- **New Call (NewCall)**
- Forward All (CfndAll)

Under call state select 'On Hook', then click on the Mobility feature to highlight blue and click the > (more than) button to move Mobility to the 'Selected Softkeys' box. Click Save.

* indicates a required item.
** indicates mandatory fields.
Softkey Template Configuration

Status

Status: Ready

Notes

Use this window to specify the Softkeys and their relative order for any phone models that support downloadable Softkey templates. This window lists all the Softkeys in the system, even though not all phones support a Softkey. For information about the supported Softkeys for a particular phone, refer to the administration guide for the phone. If you choose a Softkey that is not supported by the phone, the Softkey will not display on the phone even if you configured it in this list.

Softkey Layout Configuration

Softkey Template: Standard user-mobile

Select a call state to configure: Connected

Unselected Softkeys

Connected

Selected Softkeys (ordered by position)

Under call state selected 'Connected', then click on the Mobility feature to highlight blue and click the > (more than) button to move "Hold" to the "Selected Softkeys" box.

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Confguring Translation pattern to reach Cisco IP Phone using 4 Digit DN

300X is the pattern dialed to reach a Cisco Phone
Digits prefixed to the left of the 4-digit dialed number.
Configuring Outgoing Route Pattern

Outgoing route pattern to the Lync Client.
Cisco Unified Border Element Configuration

3845_CUBE#sh ver

Cisco IOS Software, 3800 Software (C3845-ADVENTERPRISEK9-M), Version 15.1(4)M, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2011 by Cisco Systems, Inc.
Compiled Thu 24-Mar-11 17:29 by pmd_rel_team

ROM: System Bootstrap, Version 12.3(11t)T1, RELEASE SOFTWARE (fc1)
3845_CUBE uptime is 6 weeks, 3 days, 2 hours, 38 minutes
System returned to ROM by reload at 09:30:02 pst Fri Jun 3 2011
System restarted at 09:33:05 pst Fri Jun 3 2011
System image file is "flash:c3845-adventerprisek9-mz.151-4.M.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: http://www.cisco.com/wwl/export/crypto/tool/stqrg.html
If you require further assistance please contact us by sending email to export@cisco.com.

Cisco 3845 (revision 1.0) with 483328K/40960K bytes of memory.
Processor board ID FHK0837F0KP

16 FastEthernet interfaces
2 Gigabit Ethernet interfaces
24 Serial interfaces
2 Channelized T1/PRI ports
1 Virtual Private Network (VPN) Module
2 Voice FXO interfaces
4 Voice FXS interfaces

DRAM configuration is 64 bits wide with parity enabled.
479K bytes of NVRAM.
125440K bytes of ATA System CompactFlash (Read/Write)

License Info:
License UDI:

<table>
<thead>
<tr>
<th>Device#</th>
<th>PID</th>
<th>SN</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>CISCO3845-MB</td>
<td>FOC083500LZ</td>
</tr>
</tbody>
</table>

Configuration register is 0x2102
Building configuration...

Current configuration : 9430 bytes
NVRAM config last updated at 11:02:25 pst Fri Jul 8 2011

version 15.1

service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!

hostname 3845_CUBE

!

boot-start-marker
boot system flash:c3845-adventerprisek9-mz.151-4.M.bin
boot-end-marker
!
logging buffered 99999
no logging console
enable password cisco
!
no aaa new-model
!
clock timezone pst -10 0
clock summer-time pst recurring

network-clock-participate wic 0
!
dot11 syslog

ip source-route
ip cef
!
no ip dhcp use vrf connected
ip dhcp excluded-address 172.20.151.220
ip dhcp excluded-address 172.20.151.1
!
ip dhcp pool voice
network 172.20.151.0 255.255.255.0
option 150 ip 172.20.151.220
default-router 172.20.151.220
voice service voip
allow-connections sip to sip
  signaling forward none
  sip
  session transport tcp
  header-passing
  asserted-id pa1
  history-info
  midcall-signaling passthru
  privacy-policy passthru

voice class sip-profiles 3
  request INVITE sip-header Diversion remove

crypto pki token default removal timeout 0

license udi pid CISCO3845-MB sn FOC083500LZ
archive
log config
hidekeys

redundancy

controller T1 0/0/0
  pri-group timeslots 1-24

controller T1 0/0/1

1 This command enables the basic IP-to-IP Cisco UBE feature for SIP calls
2 This command allows for SIP messages to pass-through end-to-end without modification through Cisco UBE
3 This command enables P-Asserted-ID feature on Cisco UBE
4 This command enables history-info header used to carry redirect information of redirected calls (callforward) when Diversion header is not supported.
5 This command must be enabled at a global level to maintain integrity of SIP signaling across SIP end-points.
6 This command allows privacy settings to be transparently passed across Cisco UBE SIP call legs
7 This command is used to strip the “Diversion” header from a INVITE message to MSFT Lync during a Cisco UCM redirect (callforward) call. See also dial-peer configuration
interface GigabitEthernet0/0

ip address 172.20.85.101 255.255.255.0
duplex auto
speed auto
media-type rj45

! interface GigabitEthernet0/1

no ip address

! interface Serial0/0/0:23

no ip address

! interface FastEthernet1/0

no ip address

! interface FastEthernet1/1

no ip address

! interface FastEthernet1/2

no ip address

! interface FastEthernet1/3

no ip address

! interface FastEthernet1/4

no ip address

! interface FastEthernet1/5

no ip address

! interface FastEthernet1/6

no ip address

! interface FastEthernet1/7

no ip address

! interface FastEthernet1/8

no ip address

! interface FastEthernet1/9

no ip address

! interface FastEthernet1/10
no ip address
!
interface FastEthernet1/11
no ip address
!
interface FastEthernet1/12
no ip address
!
interface FastEthernet1/13
no ip address
!
interface FastEthernet1/14
no ip address
!
interface FastEthernet1/15
no ip address
!
interface Vlan1
no ip address
!

ip forward-protocol nd
ip http server
ip http authentication local
no ip http secure-server
ip http path flash:
!
!
ip route 0.0.0.0 0.0.0.0 172.20.85.1
!
logging esm config
! control-plane
! voice-port 0/0/0:23
! voice-port 0/1/0
! voice-port 0/1/1
shutdown
! voice-port 0/1/2
!
voice-port 0/1/3
!
voice-port 0/2/0
!
voice-port 0/2/1
!

dial-peer voice 1000 voip
description from Cisco UCM to Lync outbound dial-peer
destination-pattern +1415222101.
    session protocol sipv2
session target ipv4:172.20.117.152:5068
session transport tcp
voice-class sip early-offer forced
voice-class sip profiles 3
voice-class sip block 183 sdp present
dtmf-relay rtp-npe
codec g711ulaw
!

dial-peer voice 1001 voip
description from Cisco UCM to Lync inbound dial-peer
    session protocol sipv2
incoming called-number +1415222101.
    dtmf-relay rtp-npe
codec g711ulaw
!

dial-peer voice 1002 voip
description from Lync to Cisco UCM outbound dial-peer
destination-pattern 1408566300.
    session protocol sipv2
session target ipv4:172.20.201.254
dtmf-relay rtp-npe
codec g711ulaw
!

dial-peer voice 1003 voip
description from Lync to Cisco UCM inbound dial-peer
    session protocol sipv2
    session transport tcp
incoming called-number 1408566300.
dtmf-relay rtp-npe
codec g711ulaw
!
!
sip-ua
    no remote-party-id
!
    line con 0

---

8 This command enables SIP protocol communication on the dial-peer
9 This command applies the ip address of the destination SIP call-agent (in this case Cisco UCM server)
10 This command enables TCP communication between Cisco UBE and Microsoft mediation server
11 This command enables delay-offer to early offer conversion of Cisco UCM delay-offer INVITE when calling to Microsoft Lync. Although Microsoft Lync does support delay-offer requests, providing early-offer request improves interoperability during the invocation of some supplementary services
12 This command applies the sip profile logic created in “voice-class sip-profiles 3” to this dial-peer
13 This command is used to circumvent the 183 w/SDP no early media issue.
14 This command enables DTMF transport using RFC2833
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco
no login
transport input all

! scheduler allocate 20000 1000
ntp server 172.20.2.181
end
3845_CUBE#
Configuring Transcoding g711 to g729 using CUBE

The following dial-peers are an identical match to the dial-peers in the above CUBE configuration, except for the codec settings. The codec change is made on the dial-peers facing the Cisco UCM for inbound and outbound calls. In this configuration Cisco UCM side is using G729 and Microsoft Lync is using G711. Match the configuration below to successfully transcode g711 to g729 during calls between the Microsoft Lync environment and any other environment on the opposite side of CUBE. Note that CUBE must be in flow-through mode for transcoding to work.

dial-peer voice 1000 voip
description from Cisco UCM to Lync outbound dial-peer
destination-pattern +1415222101.
session protocol sipv2
session target ipv4:172.20.117.152:5068
session transport tcp
voice-class sip early-offer forced
voice-class sip profiles 3
voice-class sip block 183 sdp present
dtmf-relay rtp-nte
codec g711ulaw
!
dial-peer voice 1001 voip
description from Cisco UCM to Lync inbound dial-peer
session protocol sipv2
incoming called-number +1415222101.
dtmf-relay rtp-nte
codec g729r8
!
dial-peer voice 1002 voip
description from Lync to Cisco UCM outbound dial-peer
destination-pattern 1408566300.
session protocol sipv2
session target ipv4:172.20.201.254
dtmf-relay rtp-nte
codec g729r8
!
dial-peer voice 1003 voip
description from Lync to Cisco UCM inbound dial-peer
session protocol sipv2
session transport tcp
incoming called-number 1408566300.
dtmf-relay rtp-nte
codec g711ulaw
!
sccp
sccp local GigabitEthernet0/0
sccp ccm 172.20.85.101 identifier 1 version 4.0
sccp
!
sccp ccm group 1
associate ccm 1 priority 1
associate profile 1 register mt9876543210ab
!
dsccfarm profile 1 transcoded
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
codec g729r8
codec g729br8
maximum sessions 5
associate application SCCP

! telephony-service
sdspfarm units 1
sdspfarm transcode sessions 10
sdspfarm tag 1 mtp9876543210ab
max-ephones 10
max-dn 20
ip source-address 172.20.85.101 port 2000
create cnf-files
<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>Lync Server</td>
<td>Microsoft Lync 2010 Server</td>
</tr>
<tr>
<td>Lync 2010</td>
<td>Microsoft Lync Client</td>
</tr>
<tr>
<td>SNR</td>
<td>Single number reach</td>
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<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>SDP</td>
<td>Session Description Protocol</td>
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</table>
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