# Microsoft OCS 2007 R2 with Microsoft Mediation Server via Cisco Unified Border Element (Enterprise Edition) 1.4 to Cisco Unified Communications Manager 7.1 using SIP

February 4, 2009

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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) 7.1 with Cisco Unified Border Element (Cisco UBE) 1.4 to interoperate with Microsoft Office Communications Server (OCS) 2007 R2 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 Microsoft Office Communicator (MOC) 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. Note for single-number reach testing a simulated SIP service provider was used to make inbound calls only. 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 OCS/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 OCS 2007 R2/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 (CUBE 1.4 will be available on IOS release 15.1.1T)
  - Cisco 2900 Series Integrated Services Routers (CUBE 1.4 will be available on IOS release 15.1.1T)
  - Cisco 2800 Series Integrated Services Routers
  - Cisco 3800 Series Integrated Services Routers
  - Cisco AS5350XM Universal Gateway
  - Cisco AS5400XM Universal Gateway

Note: 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.
Comments

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

Limitations

• Calling and connected name are not supported. Name is not sent by Microsoft Mediation Server.

• When a Cisco Unified IP phone is performing a call-forward toward a MOC 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.

• The Microsoft office communicator client (MOC) does not support call-forward on busy. As an option the MOC 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 users choice.

• During a three-way conference initiated by a MOC user the MOC user is unable to “mute” any participant that is connected to the conference using a Cisco Unified IP phone. The issue is caused by the incompatibility between Microsoft OCS media keeplive mechanism against Cisco Unified Communications solutions. Microsoft OCS utilizes a Re-INVITE session to establish single direction media toward the muted user (in this case MOC MoH is not played but media is still forced to “sendonly-reconly” attributes). During the time that the muted end-point remains in recvonly mode, Microsoft OCS system expects to receive RTCP packets from the muted end-point as a means to detect the muted user has not dropped from the conference. Cisco Unified Communications does not support RTCP in recvonly mode which causes the call to be dropped, after 30sec, from the Microsoft OCS side (Note: Cisco Unified Communications utilizes a ICMP solution to monitor livewhood of an end-point on hold). Performing a “hold” from the MOC side does not cause this issue, performing a hold forces a “inactive” media state and no RTCP packet is expected by the Microsoft mediation server.

• The issue described above (bullet 5) applies to the Microsoft Attendant client for hold and mute (Attendant client supports MoH during a mute or hold, which forces a single direction media negotiation) and all other Microsoft OCS applications that support MoH (single media direction) toward a Cisco Unified IP phone

• When a MOC user initiates a conference (Moc is the conference leader) the call will remain up even after all MOC users have dropped from the conference and only one Cisco Unified IP phone remains connected. The Microsoft never sends a BYE message toward the Cisco UBE even after all MOC users have left the conference, the Microsoft OCS 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 transcoding function.

Early-media negotiation may not be supported due to the new provisional response mechanism implemented on Microsoft OCS 2007 R2. Microsoft OCS 2007 R2 sends three provisional responses to a single INVITE (183 w/SDP, 180, 183) which causes incompatibility against Cisco UBE which expects ringback media to flow from the Microsoft OCS due to the 183 with SDP response message (early media indication), but the Microsoft OCS does not send ringback media and the calling user never hears ringback. The workaround is to use the command “voice-class sip block 183 sdp present” and apply it to the appropriate dial-peer, not globally, in Cisco UBE to block all 183 messages from mst OCS toward Cisco UCM allowing only 180 message to be sent to Cisco UCM and receiving local generated ringback on the calling side.
**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 MOC app.)
- DTMF (RFC2833)
- Single-number reach (SNR) for Cisco Unified IP phone and Microsoft MOC (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
- Connected Name not supported by the Microsoft platform
- Connected Number not supported by the Microsoft platform
System Components

Hardware Requirements

Cisco equipment

- Cisco IOS gateway running Cisco Unified Border Element (Enterprise Edition) release 1.4 (IOS version 15.0.1XA and future 15.T releases)
- 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 1.4 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 - switch
- Cisco Unified IP Phones (The topology diagram shows 7970 and 7961, but any Cisco IP phone model supporting RFC2833 can be used)
- Cisco MCS 7800 Series server (Cisco Unified Communications Manager)

Microsoft equipment

- Cisco MCS 7825 server (OCS 2007 front-end server)
- Cisco MCS 7825 server (Active Directory server)
- Cisco MCS 7825 server (Mediation Server)
- Cisco MCS 7825 server (Pool server)
- 2 x Laptops with Windows XP and Microsoft Office Communicator 2007 (Beta 3)

Software Requirements

- Microsoft OCS 2007 R2
- Microsoft Mediation Server for Microsoft OCS 2007 R2
- Cisco IOS 15.0.1.XA
- 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, TDMIP GW, INT VOICE/VIDEO, IPIP GW, TDMIP GW AES
- Cisco Unified Communications Manager version 7.1.3.21901-1
Configuration

Configuring Microsoft OCS 2007 R2 Enterprise edition

1. Configuring Domain Name Server.
   - Forward Lookup Zone
   - Reverse Lookup Zone
2. Configuration of FrontEnd Server
3. Configuring User - General
4. Configuring User - Account
5. Configuring User - Communications
6. Configuring User - Telephony options
7. Active Directory User configuration
8. Configuring Users - Pool
9. Configuring Users – Pool properties
10. Mediation Server configuration
11. Mediation Server overview

Microsoft Office Communicator Configuration.

Domain Name Server Configuration

Forward Lookup zone

This screenshot shows the DNS entries. This highlights OCS pool in MSPBX.com domain.
Reverse Lookup zone

This screenshot shows PTR record entry for pool1-fe for reverse lookup.
This screenshot shows OCS Pool FrontEnd Server settings. Configuration details are found in the following pages.
This screenshot shows how to add/edit FrontEnd server properties by right clicking on applicable pool.
– Click on Add to add and configure Front End IP address, port and transport protocol
This screen shot shows how to add/edit pool properties by right clicking on applicable pool.
Configuration of FrontEnd Server (Page 5 of 5)

Configuration using the Wizard during install.
Configuring User – Account

Active Directory Users and Computers

- User: Administrator
- User: CPE1
- User: CPE2
- User: CPE3
- User: CPE4
- User: cube01 pool1
- User: cube02 pool1
- User: Denied RODC Password Replication Group
- User: Domain Administrators
- User: Domain Admins
- User: Domain Computers
- User: Domain Controllers
- User: Domain Guests
- User: Domain Users
- User: Enterprise Admins
- User: Enterprise Readonly Domain Controllers
- User: Group Policy Creator Owners
- User: Guest
- User: IAS and JAS Servers
- User: MSRPC Universal Services
- User: Netlogon Universal Services
- User: RTCUniversalGlobalGroup
- User: RTCUniversalGlobalGroupRead
- User: RTCUniversalGlobalGroupWrite
- User: RTCUniversalGuest
- User: RTCUniversalReadOnlyGroup
- User: RTCUniversalServerAdmins
- User: RTCUniversalServerAdminsRead
- User: RTCUniversalServerAdminsWrite
- User: RTCUniversalUserAdmins
- User: RTCUniversalUserAdminsRead
- User: RTCUniversalUserAdminsWrite
- User: Schema Admins

User logon name: cube01
Use logon name (pre-Windows 2000): MSPbx
User cannot change password
Password never expires
Account expires: Never

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Configuring User – Communications

Active Directory Users and Computers

Name | Type
--- | ---
Administrator | User
Allowed RODC Password Replication | Security Group
Cert Publishers | Security Group
CME1 | User
CME11 MSPBX | User
CME12 MSPBX | User
CME2 | User
CME3 | User
CME4 | User
CME-0501 Pool1 | User
CME-0502 Pool1 | User
cube01 pool1 | User

Enable user for Office Communications Server
Sign-in name: sbocube01 @ MSPBX.COM
Server or pool: Pool1, MSPBX.COM

Meeting settings:  
Telephone settings:  
Other settings:  

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EDCS#846451 Rev # 3
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.

- This window pops up after clicking on NEXT > in the previous page.
### Configuring Users - Pool

The screenshot shows the list of Pool users when viewing from the Front-end server.

<table>
<thead>
<tr>
<th>Enabled</th>
<th>Display name</th>
<th>SIP URL</th>
<th>Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enabled</td>
<td>CME1</td>
<td>sip: <a href="mailto:CME1@MSPBX.COM">CME1@MSPBX.COM</a></td>
<td>User</td>
</tr>
<tr>
<td>Enabled</td>
<td>CME2</td>
<td>sip: <a href="mailto:CME2@MSPBX.COM">CME2@MSPBX.COM</a></td>
<td>User</td>
</tr>
<tr>
<td>Enabled</td>
<td>CME3</td>
<td>sip: <a href="mailto:CME3@MSPBX.COM">CME3@MSPBX.COM</a></td>
<td>User</td>
</tr>
<tr>
<td>Enabled</td>
<td>CME11:MSPBX</td>
<td>sip: <a href="mailto:CME11@MSPBX.COM">CME11@MSPBX.COM</a></td>
<td>User</td>
</tr>
<tr>
<td>Enabled</td>
<td>CME12:MSPBX</td>
<td>sip: <a href="mailto:CME12@MSPBX.COM">CME12@MSPBX.COM</a></td>
<td>User</td>
</tr>
<tr>
<td>Enabled</td>
<td>CME4</td>
<td>sip: <a href="mailto:CME4@MSPBX.COM">CME4@MSPBX.COM</a></td>
<td>User</td>
</tr>
<tr>
<td>Enabled</td>
<td>CME-0501 Pool1</td>
<td>sip: <a href="mailto:CME-0501@MSPBX.COM">CME-0501@MSPBX.COM</a></td>
<td>User</td>
</tr>
<tr>
<td>Enabled</td>
<td>CME-0502 Pool1</td>
<td>sip: <a href="mailto:CME-0502@MSPBX.COM">CME-0502@MSPBX.COM</a></td>
<td>User</td>
</tr>
</tbody>
</table>
Configuring Users – Pool Properties

Click to complete user configuration
The Mediation Server has two Ethernet Interfaces, one listens to the Microsoft Office Communications Server (172.20.241.101) and the other to the Cisco UBE (172.20.8.26)
Mediation Server Configuration (Page 2 of 7)
Note: The next hop PSTN Gateway connected to the Mediation Server is the Cisco UBE.
Mediation Server Configuration (Page 4 of 7)
Number patterns to be routed through this Mediation Server.
Mediation Server Configuration (Page 6 of 7)
Meditation Server Overview (Page 1 of 2)

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<th>General Settings</th>
</tr>
</thead>
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<td>Windows services</td>
</tr>
<tr>
<td>Mediation service: Running</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Certificate settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Notes: MedServ.MSPBL.COM</td>
</tr>
<tr>
<td>Issuer Name: MSPBL-FOC-CA</td>
</tr>
<tr>
<td>Expiration Date: 12/19/2011</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Location Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>123 CNE</td>
</tr>
<tr>
<td>Phone Pattern: ^+(+)?(\d{3})\d{3}-\d{4}$</td>
</tr>
<tr>
<td>Translation: $</td>
</tr>
<tr>
<td>54321 Phone Pattern: ^+(+)?(\d{3})\d{3}-\d{4}$</td>
</tr>
<tr>
<td>Translation: +1-800-555-1234</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Edge Server FQDN:</th>
</tr>
</thead>
<tbody>
<tr>
<td>&lt;Name&gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Listening Connections</th>
</tr>
</thead>
<tbody>
<tr>
<td>Listening address for Communications Server: 172.20.241.101</td>
</tr>
<tr>
<td>Communications Server listening port: 5061</td>
</tr>
<tr>
<td>Listening address for Gateway traffic: 172.20.127.101</td>
</tr>
<tr>
<td>PSTN Gateway Listening Port: 5000</td>
</tr>
<tr>
<td>Media port range: 600000 - 640000</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Next Hop Connections</th>
</tr>
</thead>
<tbody>
<tr>
<td>Communications Server Next Hop FQDN: MedServ.MSPBL.COM</td>
</tr>
<tr>
<td>Communications Server Next Hop Port: 5061</td>
</tr>
<tr>
<td>PSTN Gateway Address: 172.20.3.26</td>
</tr>
<tr>
<td>PSTN Gateway Port: 5000</td>
</tr>
<tr>
<td>Gateway Transport Type: TCP</td>
</tr>
<tr>
<td>Gateway Encryption Level: Do Not Support Encryption</td>
</tr>
</tbody>
</table>
### Route Information

The following routes are served by this Mediation Server. Please use the Route tab on the Voice property page to add, modify or delete a route. To access the Voice property page, right click the Forest node of the MMC tree-view pane.

#### route_to_cube

**Phone Number Pattern:**

```
^\(1510555\d\(3\)\d+\)$
```

**Phone Usage:**

- [ ] Default Usage

<table>
<thead>
<tr>
<th>Description</th>
<th>Sample phone usage</th>
</tr>
</thead>
</table>
Configuring Normalization rule for MOC user to be able to dial a 4-digit DN or a 10-digit number
Location profiles define how numbers are to be translated when dialed from a defined location. Each profile has a set of normalization rules.

Location Profiles:

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>123</td>
<td>CME_TEST</td>
</tr>
<tr>
<td>pass_al</td>
<td>no translation applied</td>
</tr>
</tbody>
</table>

Click “Edit”
Click "Add" if you don’t have an existing one
The phone pattern expression and translation expression allows the MOC user to dial a 4-digit number (4401) DN and/or a 10-digit 15105554401 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 OCS has the capability of manipulating digits as well, but this document does not cover those configurations.
Microsoft Office Communicator Configuration (Page 1 of 7)

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

– Click Advanced button to select the Advanced Connection Settings.
Because there is no DNS entry for this Microsoft Office Communicator, manual configuration is used.
Sign in to Microsoft Office Communicator.

```
Sign-in address:
cube01@mspbx.com
Change your sign-in address

User name:
cube01@mspbx.com

Examples:  domain\username
           someone@example.com

Password:
**********

Sign in as:  Available

Sign In
```
Add contacts.

### Recent Contacts
- 15105554401
- cube02 pool1 Offline
- 4401
- 4404
- 8010
- 15105554402
- 15105553350
- 15105554080
- 4402

### All Contacts
- cube02 pool1 Offline
Modify user options as needed.
Office Communicator - Options

My phone numbers

To enter or edit your phone numbers, click the related button. To share the number with others, select the adjacent check box.

- Work Phone: +14084445501
- Mobile Phone
- Home Phone
- Other Phone

Phone integration

- Enable integration with your phone system

Phone accessibility

To use text communication via a telephone line, turn on TTY mode. Note that a TTY device must be connected to interpret the modified audio.

- Turn on TTY mode

OK Cancel Help
Microsoft Office Communicator Configuration (Page 7 of 7)

Call-Forwarding Settings:

![Image of Microsoft Office Communicator window with call-forwarding settings option highlighted]
Office Communicator - Call-Forwarding Settings

Do the following when I get calls:

- **Ring me**

**Summary of Current Settings**

Incoming calls will ring the following: +1 (408) 444-5501 (you).

Choose an additional number to ring:

<table>
<thead>
<tr>
<th>Ring</th>
<th>Phone Numbers</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Work +1 (408) 444-5501 will always be rung</td>
</tr>
<tr>
<td></td>
<td>Do not ring an additional number</td>
</tr>
<tr>
<td></td>
<td>15105553350</td>
</tr>
</tbody>
</table>

Send unanswered calls to the following:

- **None**

Ring for this many seconds before redirecting:

- **59**

- **Only apply these settings during my working hours specified in Outlook**

[OK] [Cancel] [Help]
Cisco Unified Border Element configuration

Router#sh ver
Cisco IOS Software, 3800 Software (C3845-ADVENTERPRISEK9-M), Version 15.0(1)XA1, RELEASE SOFTWARE (fc2)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2009 by Cisco Systems, Inc.
Compiled Fri 18-Dec-09 20:06 by prod_rel_team

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

Router uptime is 2 hours, 33 minutes
System returned to ROM by reload at 20:17:40 UTC Thu Jan 14 2010
System image file is "flash:c3845-adventerprisek9-mz.150-1.XA1.bin"

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 480255K/44032K bytes of memory.
Processor board ID FTX0933A1JA
2 Gigabit Ethernet interfaces
77 Serial interfaces
4 Channelized T1/PRI ports
1 Virtual Private Network (VPN) Module
1 802.11 Radio
DRAM configuration is 64 bits wide with parity enabled.
479K bytes of NVRAM.
125184K bytes of ATA System CompactFlash (Read/Write)

License Info:

License UDI:

-------------------------------
Device#   PID                   SN
-------------------------------
*0        CISCO3845-MB          FOC09250PU2

Configuration register is 0x2102
Router#sh run
Building configuration...

Current configuration : 8780 bytes
!
! Last configuration change at 19:12:56 UTC Thu Jan 14 2010
!
version 15.0
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname Router
!
boot-start-marker
boot system flash:c3845-adventerprisek9-mz.150-1.XA.bin
boot-end-marker
!
logging buffered 3000000
no logging console
enable password cisco
!
no aaa new-model
!
!
!
!
ip source-route
ip cef
!
!
no ip domain lookup
no ipv6 cef
!
multilink bundle-name authenticated
!
!
!
!
voice-card 0
!
voice-card 3
!
!
!
voice service voip
allow-connections sip to sip
  
 signaling forward none

1 This command enables the basic IP-to-IP Cisco UBE feature for SIP calls
h323
sip
bind control source-interface GigabitEthernet0/0
bind media source-interface GigabitEthernet0/0
header-passing error-passthru2
asserted-id pai3
history-info4
midcall-signaling passthru5
privacy-policy passthru6
!
voice class sip-profiles 3
request INVITE sip-header Diversion remove7
!
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!
![2] This command allows for SIP error 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 OCS during a Cisco UCM redirect (callforward) call. See also dial-peer configuration
interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
media-type rj45
!
!
!
ip forward-protocol nd
no ip http server
no ip http secure-server
!
!
ip route 0.0.0.0 0.0.0.0 172.20.8.1
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!
control-plane
!
!
!
!

dial-peer voice 9999 voip
description from OCS to Cisco UCM outbound dial-peer
destination-pattern 1510555....
session protocol sipv2
session target ipv4:172.20.8.254
dtmf-relay rtp-nte
codec g711ulaw

dial-peer voice 5500 voip
description from Cisco UCM to OCS outbound dial-peer
destination-pattern +140844455..
session protocol sipv2
session target ipv4:172.20.127.101
session transport tcp
voice-class sip early-offer forced
voice-class sip profiles 3
voice-class sip block 183 sdp present

---

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 DTMF transport using RFC2833
11 This command enables TCP communication between Cisco UBE and Microsoft mediation server
12 This command enables early-offer to early offer conversion of Cisco UCM delay-offer INVITE when calling to Microsoft OCS. Although Microsoft OCS does support delay-offer requests, providing early-offer request improves interoperability during the invocation of some supplementary services
13 This command applies the sip profile logic created in “voice-class sip-profiles 3” to this dial-peer
This command is used to circumvent the 183 w/SDP no early media issue.
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 OCS is using G711. Match the configuration below to successfully transcode g711 to g729 during calls between the Microsoft OCS 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 9999 voip
description outgoing to MS OCS
destination-pattern 1510555....
session protocol sipv2
session target ipv4:172.20.8.254
voice-class sip profiles 4
dtmf-relay rtp-nte
  codec g729r8
  !
dial-peer voice 5500 voip
destination-pattern +140844455..
session protocol sipv2
session target ipv4:172.20.127.101
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 99991 voip
destination-pattern 1510555....
session protocol sipv2
session transport tcp
incoming called-number 1510555....
voice-class sip profiles 4
dtmf-relay rtp-nte
  codec g711ulaw
  !
dial-peer voice 55001 voip
session protocol sipv2
incoming called-number +140844455..
dtmf-relay rtp-nte
  codec g729r8
  !
sccp local GigabitEthernet0/0
sccp ccm 172.20.8.26 identifier 1 version 3.1
sccp
  !
sccp ccm group 1
  associate ccm 1 priority 1
  associate profile 1 register mtp00146a7299e0
  !
dspfarm profile 1 transcode
  codec g711ulaw
  codec g711alaw
  codec g729ar8
  codec g729abr8

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codec g729r8
codec g729br8
maximum sessions 5
associate application SCCP
!
telephony-service
sdspfarm units 1
sdspfarm transcode sessions 10
sdspfarm tag 1 mtp00146a7299e0
max-ephones 10
max-dn 20
ip source-address 172.20.8.26 port 2000
system message CME InterOp
create cnf-files
**Configuring Cisco Unified Communications Manager**

Cisco Unified Communications Manager system version

**SIP trunk configuration**

**SIP trunk main page**

### Find and List Trunks

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Source Side</th>
<th>Destination Side</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>FX2</td>
<td>incoming route from FX2 to FX2 for FX calls</td>
<td>FX2</td>
<td>FX2</td>
<td>SIP Trunk</td>
<td>4000</td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>FX3</td>
<td>outgoing route from FX3 to FX3 for FX calls</td>
<td>FX3</td>
<td>FX3</td>
<td>SIP Trunk</td>
<td>4000</td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>FX4</td>
<td>incoming route from FX4 to FX4 for FX calls</td>
<td>FX4</td>
<td>FX4</td>
<td>SIP Trunk</td>
<td>4000</td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>FX5</td>
<td>outgoing route from FX5 to FX5 for FX calls</td>
<td>FX5</td>
<td>FX5</td>
<td>SIP Trunk</td>
<td>4000</td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>FX6</td>
<td>incoming route from FX6 to FX6 for FX calls</td>
<td>FX6</td>
<td>FX6</td>
<td>SIP Trunk</td>
<td>4000</td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>FX7</td>
<td>outgoing route from FX7 to FX7 for FX calls</td>
<td>FX7</td>
<td>FX7</td>
<td>SIP Trunk</td>
<td>4000</td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>FX8</td>
<td>incoming route from FX8 to FX8 for FX calls</td>
<td>FX8</td>
<td>FX8</td>
<td>SIP Trunk</td>
<td>4000</td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>FX9</td>
<td>outgoing route from FX9 to FX9 for FX calls</td>
<td>FX9</td>
<td>FX9</td>
<td>SIP Trunk</td>
<td>4000</td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
</tbody>
</table>

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Page 53 of 93
EDCS#846451 Rev # 3
## SIP trunk to Cisco UBE

### Cisco Unified CM Administration

#### Device Information
- **Device Name**: MUX-UCS
- **Device Pool**: Default

#### Common Device Configuration
- **Call Classification**: <None>
- **Media Resource Group List**: MGC_default
- **Location**: <None>
- **AAA Group**: <None>
- **Fax Name**: <None>
- **Fax Port**: 0

### Inbound Calling Party Settings

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix Settings</th>
<th>Default Prefix Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>Default</td>
</tr>
</tbody>
</table>

![Application Note]

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Application Note

---

**Multilevel Precedence and Preemption (MLPP) Information**

MLPP Domain

**Call Routing Information**

- Remote-Party-ID
- Asserted-Identity
- Asserted-Type
- SIP-Privacy

**Inbound Calls**

- Significant Digits
- Connected-Line-ID-Presentation
- Connected-Name-Presentation
- Calling-Search-Space
-Raw-Calling-Search-Space
- Profile-OID

**Outbound Calls**

- Called-Party-Transformation-CSS
- Called-Party-Calling-Party-Transformation-CSS
- Called-Party-Selection
- Calling-Line-ID-Presentation
- Calling-Name-Presentation
- Called-ID-DN
- Called-Name

**SIP Information**

- Destination-Address
- Destination-Address-IPV6
- Destination-Address-IPv6-Address
- Destination-Address-IPv6-Address-is-an-SRV
- Destination-Address
- Location-URI
- Location-IPV6
- Location-IPV6-Address
- Location-IPV6-Address-is-an-SRV
- Location-IPV6
- Location-URI
- Location-IPV6
- Location-IPV6-Address
- Location-IPV6-Address-is-an-SRV
- Location-IPV6
- Location-URI
- Location-IPV6
- Location-IPV6-Address
- Location-IPV6-Address-is-an-SRV
- Location-IPV6
- Location-URI

**Geolocation Configuration**

- Geolocation
- Geolocation-Filter

---

* Indicates required item.
** Indicates device reset is not required for changes to Packet-Capture Mode and Packet-Capture Duration.
Region configuration

Region main page
Region ‘Default’ configuration

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Region Configuration

Find and List Regions Information
Name: Default

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>G.729</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>phones/g711</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>phones/g729</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

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

Modify Relationship to Other Regions

<table>
<thead>
<tr>
<th>Regions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
</tr>
<tr>
<td>G.729</td>
</tr>
<tr>
<td>phones/g711</td>
</tr>
<tr>
<td>phones/g729</td>
</tr>
</tbody>
</table>

Audio Codec: Keep Current Setting
Video Call Bandwidth: Use System Default
Link Loss Type: Use System Default

* 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 phones/g711 is assigned to Cisco IP phones using the device pool of the same name. Using the codec relationships to Default is g711, which should not imply any g711 will be supported but signifies g711 or any codec with a lower bandwidth requirement will be supported (e.g. g729).
### Region 'phonesg711' configuration

<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>364</td>
<td>Use System Default</td>
</tr>
<tr>
<td>G.729</td>
<td>G.711</td>
<td>364</td>
<td>Use System Default</td>
</tr>
<tr>
<td>phonesg711</td>
<td>G.711</td>
<td>364</td>
<td>Use System Default</td>
</tr>
<tr>
<td>phonesg729</td>
<td>G.711</td>
<td>364</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

NOTE: Region(s) not displayed for Region 'phonesg711' configuration.

---

**Modify Relationship to other Regions**

<table>
<thead>
<tr>
<th>Regions</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>Keep Current Setting</td>
</tr>
<tr>
<td>phonesg711</td>
<td>Keep Current Setting</td>
<td>Use System Default</td>
<td>Keep Current Setting</td>
</tr>
<tr>
<td>phonesg729</td>
<td>Keep Current Setting</td>
<td>None</td>
<td>Keep Current Setting</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.**
### Device Pool – Default

#### Cisco Unified CM Administration
For Cisco Unified Communications Solutions

#### Device Pool Configuration

<table>
<thead>
<tr>
<th>Device Pool Name *</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified Communications Manager Group</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Search Space for Auto-registration</td>
<td>Default</td>
</tr>
<tr>
<td>Reverted Call Focus Priority</td>
<td>Default</td>
</tr>
<tr>
<td>Local Route Group</td>
<td>Default</td>
</tr>
</tbody>
</table>

#### Reaming Sensitive Settings

<table>
<thead>
<tr>
<th>Date/Time Group</th>
<th>CMLocal</th>
</tr>
</thead>
<tbody>
<tr>
<td>Region *</td>
<td>Default</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>Default</td>
</tr>
<tr>
<td>Location</td>
<td>Default</td>
</tr>
<tr>
<td>Network Locale</td>
<td>Default</td>
</tr>
<tr>
<td>SRST Reference</td>
<td>Default</td>
</tr>
<tr>
<td>Connection Monitor Duration **</td>
<td>Disable</td>
</tr>
<tr>
<td>Single Button Barge</td>
<td>Default</td>
</tr>
<tr>
<td>Join Across Lines</td>
<td>Default</td>
</tr>
<tr>
<td>Physical Location</td>
<td>Default</td>
</tr>
<tr>
<td>Device Mobility Group</td>
<td>Default</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

| 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 (DefaultPool/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>Prefix 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>

---

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**Device Pool – phonesg711**

### Device Pool Configuration

<table>
<thead>
<tr>
<th>Status</th>
<th>Status: Ready</th>
</tr>
</thead>
</table>

### 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>Referred Call Focus 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 Barge</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

| 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>Clear Prefix Settings</th>
<th>Default Prefix Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number Type</td>
<td>Prefix</td>
</tr>
<tr>
<td>National Number</td>
<td>Default</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
</tr>
</tbody>
</table>

* indicates required items.

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

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

****These five parameters will overwrite device level settings when device is roaming and in the same device mobility group.
### Media resource group assigned in Cisco SIP trunk

#### Cisco Unified CM Administration
For Cisco Unified Communications Solutions

#### Media Resource Group Configuration

<table>
<thead>
<tr>
<th>Sn.</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>MRG_default</td>
</tr>
</tbody>
</table>

**Devices for this Group**

- **Available Media Resources**: c8b01856bb7x1, m9b0216650931, m8b02165655bb

**Selected Media Resources**

<table>
<thead>
<tr>
<th>Media Resource</th>
</tr>
</thead>
<tbody>
<tr>
<td>ANI_2 (ANI)</td>
</tr>
<tr>
<td>CFB_2 (CFB)</td>
</tr>
<tr>
<td>MOH_1 (MOH)</td>
</tr>
<tr>
<td>MOH_2 (MOH)</td>
</tr>
</tbody>
</table>

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

---

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EDCS#846451 Rev # 3
Media resource list assigned in Cisco SIP trunk

---

**Media Resource Group List Configuration**

- **Status**
  - Status: Ready

- **Media Resource Group List Status**
  - Media Resource Group List: MRGL_default (used by 4 devices)

- **Media Resource Group List Information**
  - Name: MRGL_default

- **Media Resource Groups for this List**
  - Available Media Resource Groups:
    - [List]
  - Selected Media Resource Groups:
    - MRGL_default

---

* - indicates required item.
### EDCS#846451 Rev # 3

**Application Note**

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

---

**Phone Configuration**

**Related Links:** Back To Find List

**Protocol Specific Information**

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Protocol Capture Mode</td>
<td>&lt;none&gt;</td>
</tr>
<tr>
<td>Protocol Capture Duration</td>
<td>6</td>
</tr>
<tr>
<td>Presence Group</td>
<td>Standard Presence Group</td>
</tr>
<tr>
<td>SIP Dial Rules</td>
<td>&lt;none&gt;</td>
</tr>
<tr>
<td>HTTP Preferred Originating Code</td>
<td>7111111</td>
</tr>
<tr>
<td>Device Security Profile</td>
<td>Cisco 7970 - Standard SIP Non-Secure Profile</td>
</tr>
<tr>
<td>Receiving Call/Receive Search Space</td>
<td>&lt;none&gt;</td>
</tr>
<tr>
<td>SUBSCRIBE Call/Receive 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>

**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 Null String</td>
</tr>
<tr>
<td>Authentication String</td>
<td></td>
</tr>
<tr>
<td>Key Size (Bits)</td>
<td>1024</td>
</tr>
<tr>
<td>Operation Completes By</td>
<td>2020-01-25 (YYYY-MM-DD)</td>
</tr>
</tbody>
</table>

**Expansion Module Information**

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Module 1 Load Name</td>
<td>&lt;none&gt;</td>
</tr>
<tr>
<td>Module 2 Load Name</td>
<td>&lt;none&gt;</td>
</tr>
</tbody>
</table>
### External Data Locations Information (Leave blank to use default)

<table>
<thead>
<tr>
<th>Information</th>
<th>Directory</th>
<th>Messages</th>
<th>Services</th>
<th>Authentication Server</th>
<th>Proxy Server</th>
<th>Idle</th>
<th>Idle Timer (seconds)</th>
</tr>
</thead>
</table>

### Extension Information

- **Enable Extension Mobility**
- **Log Out Profile**: [Use Current Device Settings]
- **Log In Time**: <None>
- **Log Out Time**: <None>

### MLPP Information

- **MLPP Domain**: <None>

### Do Not Disturb

- **Do Not Disturb**
- **DND Option**: [Use Common Phone Profile Setting]
- **DND Incoming Call Alert**: <None>

### Secure Shell Information

- **Secure Shell User**
- **Secure Shell Password**
### Product Specific Configuration Layout

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td>Enabled</td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td></td>
</tr>
<tr>
<td>PC Port</td>
<td></td>
</tr>
<tr>
<td>Settings Access</td>
<td></td>
</tr>
<tr>
<td>Getting ARP</td>
<td></td>
</tr>
<tr>
<td>PC Voice VLAN Access</td>
<td></td>
</tr>
<tr>
<td>Auto Line Select</td>
<td></td>
</tr>
<tr>
<td>Web Access</td>
<td></td>
</tr>
<tr>
<td>Days Display Not Active</td>
<td></td>
</tr>
<tr>
<td><strong>Day</strong></td>
<td></td>
</tr>
<tr>
<td>Monday</td>
<td></td>
</tr>
<tr>
<td>Tuesday</td>
<td></td>
</tr>
<tr>
<td>Display On Time</td>
<td></td>
</tr>
<tr>
<td>Display On Duration</td>
<td></td>
</tr>
<tr>
<td>Display Idle Timeout</td>
<td></td>
</tr>
<tr>
<td>Span to PC Port</td>
<td></td>
</tr>
<tr>
<td>Logging Display</td>
<td></td>
</tr>
<tr>
<td>Load Server</td>
<td></td>
</tr>
<tr>
<td>Recording Tone</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Local Volume</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Remote Volume</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Duration</td>
<td></td>
</tr>
<tr>
<td>Display On When Incoming Call</td>
<td></td>
</tr>
<tr>
<td>RTOC</td>
<td></td>
</tr>
<tr>
<td>&quot;Hold&quot; Soft Key Timer</td>
<td></td>
</tr>
<tr>
<td>Auto Call Select</td>
<td></td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
</tr>
<tr>
<td>Advertise VPQ Codec</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset (UC Control)</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset (UC Control)</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset</td>
<td></td>
</tr>
<tr>
<td>Peer Firmware Sharing</td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port</td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port</td>
<td></td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority</td>
<td></td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td></td>
</tr>
<tr>
<td>802.1x Authentication</td>
<td></td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure</td>
<td></td>
</tr>
<tr>
<td>Minimum Ring Volume</td>
<td></td>
</tr>
<tr>
<td>Headset Sidetone Level</td>
<td></td>
</tr>
<tr>
<td>E-911 Misc Dialing</td>
<td></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 CAPF 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

Directory Number Configuration

Status: Ready

Directory Number Information
- Directory Number: 4401
- Route Partition: < None >
- Description: 
- Alerting Name: cubi901
- ASCII Alerting Name: cubi901
- Allow Control of Device from CTI
- Associated Devices: EDCS#846451
- Dissociate Devices:

Directory Number Settings
- Voice Mail Profile: < None >
- Calling Search Space: < None >
- Presence Group: Standard Presence group
- User Hold MOC Audio Source: < None >
- Network Hold MOC Audio Source: < None >
- Auto Answer: Auto Answer Off

Associated Remote Destinations
- Name: MOC01
- Destination Number: 14004445501
- Owner: cubi901

AAR Settings
- Voice Mail
- AAR Destination Mask
- AAR Group: < None >
- Retain this destination in the call forwarding history
## Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Call Forward and Call Pickup Settings</th>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Search Space Activation Policy</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>✓ or □</td>
</tr>
<tr>
<td>Forward All</td>
<td></td>
<td></td>
<td>Use System Default</td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forward Busy Internal</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy External</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer Internal</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer External</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage External</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward on CTL Failure</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered Internal</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

No Answer Ring Duration (seconds): __________

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>Park Monitoring Forward No Retrieve Destination External</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Park Monitoring Forward No Retrieve Destination Internal</td>
<td>✓ or □</td>
<td>✓ or □</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

Park Monitoring Reversion Timer: __________

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

## MLPP Alternate Party Settings

<table>
<thead>
<tr>
<th>Target (Destination)</th>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
</table>

MLPP Calling Search Space: < None >

MLPP No Answer Ring Duration (seconds): __________
## Application Note

### Line Settings for All Devices

- **Hold Reversion Ring Duration (seconds)**: Setting the Hold Reversion Ring Duration to zero will disable the feature.
- **Hold Reversion Notification Interval (seconds)**: Setting the Hold Reversion Notification Interval to zero will disable the feature.

### Line 1 on Device SEP00192FB9CA0A

<table>
<thead>
<tr>
<th>Value</th>
<th>Update Shared Device Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Display (Internal Call ID)</strong>: sube01</td>
<td>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.</td>
</tr>
<tr>
<td><strong>ASCII Display (Internal Call ID)</strong>: sube01</td>
<td></td>
</tr>
<tr>
<td><strong>Line Text Label</strong>: sube01</td>
<td></td>
</tr>
<tr>
<td><strong>ASCII Line Text Label</strong>: sube01</td>
<td></td>
</tr>
<tr>
<td><strong>External Phone Number Mask</strong>:</td>
<td></td>
</tr>
<tr>
<td><strong>Visual Message Waiting Indicator Policy</strong>: Use System Policy</td>
<td></td>
</tr>
<tr>
<td><strong>Audible Message Waiting Indicator Policy</strong>: Default</td>
<td></td>
</tr>
<tr>
<td><strong>Ring Setting (Phone Idle)</strong>: Use System Default</td>
<td>Applies to this line when any line on the phone has a call in progress.</td>
</tr>
<tr>
<td><strong>Ring Setting (Phone Active)</strong>: Use System Default</td>
<td></td>
</tr>
<tr>
<td><strong>Call Pickup Group Audio Alert Setting (Phone Idle)</strong>: Use System Default</td>
<td></td>
</tr>
<tr>
<td><strong>Call Pickup Group Audio Alert Setting (Phone Active)</strong>: Use System Default</td>
<td></td>
</tr>
<tr>
<td><strong>Recording Option</strong>: Call Recording Disabled</td>
<td></td>
</tr>
<tr>
<td><strong>Recording Profile</strong>:</td>
<td></td>
</tr>
<tr>
<td><strong>Monitoring Calling Search Space</strong>:</td>
<td></td>
</tr>
</tbody>
</table>

### Multiple Call/Call Waiting Settings on Device SEP00192FB9CA0A

**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)

### Formatted Call Information Display on Device SEP00192FB9CA0A

- **Caller Name**
- **Caller Number**
- **Redirected Number**
- **Dialled Number**

### Users Associated with Line

<table>
<thead>
<tr>
<th>Full Name</th>
<th>User ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco</td>
<td>CiscoMac01</td>
</tr>
</tbody>
</table>

**Note:** You must create an "end user" for SNR (modem connect) to work.

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EDCS#846451 Rev # 3
### Directory Number Associations

| Primary Extension | 4401 |

### Mobility Information
- **Enable Mobility**: [on]
- **Primary User Device**: SEP03154FB9CA9A
- **Enable Mobile Voice Access**: [on]
- **Maximum Wait Time for Desk Pickup**: 12000
- **Remote Destination Limit**: 4
- **Remote Destination Profiles**: NOC01

### CAPF Information
- **Associated CAPF Profiles**:

### Permissions Information
- **Groups**:
  - Standard CCM Admin Users
  - Standard CCM End Users
- **Roles**:
  - Standard CCM Admin Users
  - Standard CCM End Users
  - Standard CTI Enabled

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Configuring Microsoft Office client DID number as remote destination for SNR (mobile connect)

Remote Destination

- **Association Information**
  - Line [1] - 4401 (no partition)

- **Remote Destination Information**
  - Name: NOC01
  - Destination Number: 14084445001
  - Answer Too Soon Timer: 1800
  - Answer Too Late Timer: 19000
  - Delay Before Ringing Timer: 48000
  - Remote Destination Profile: NOC01

- **When Mobile Connect is Enabled**
  - Ring Schedule:
    - All the time
    - As specified below
      - Monday: All Day, No Office Hours, No Office Hours
      - Tuesday: All Day, No Office Hours, No Office Hours
      - Wednesday: All Day, No Office Hours, No Office Hours
      - Thursday: All Day, No Office Hours, No Office Hours
      - Friday: All Day, No Office Hours, No Office Hours
      - Saturday: All Day, No Office Hours, No Office Hours
      - Sunday: All Day, No Office Hours, No Office Hours
  - Time Zone: GMT

- **When receiving a call during the above ring schedule:**
  - Always ring this destination
  - Ring this destination only if caller is in
    - Not Selected
  - Do not ring this destination if caller is in
    - Not Selected

* indicates required item.
### Remote Destination Profile Information

<table>
<thead>
<tr>
<th>Name</th>
<th>MOC01</th>
</tr>
</thead>
<tbody>
<tr>
<td>User ID</td>
<td>CiscoMOC01</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Device Pool</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>User Hold Audio Source</td>
<td>1-SampleAudioSource</td>
</tr>
<tr>
<td>Network Hold Audio Source</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Privacy</td>
<td>Default</td>
</tr>
<tr>
<td>Remoting Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Associated Remote Destinations</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name</td>
</tr>
</tbody>
</table>

Remote Destination created in previous step

### Do Not Disturb

- [ ] Do Not Disturb
- [ ] DRD Option: Call Reject

* Indicates required item.
### Application Note

**Configuring Cisco Skinny IP phone**

#### Cisco Unified CM Administration

For Cisco Unified Communications Solutions

**Phone Configuration**

- **Device Protocol:** SCCP

**Device Information**

- **Registered With Cisco Unified Communications Manager CP-CMIYAS**
- **IPV4 Address:** 10.20.20.235
- **Active User ID:** 1
- **Device is Active:** Yes
- **Device is Trusted:** Yes
- **User Login ID:** phoneno711
- **Description:** Auto-4464
- **Device Phone:** Default

**Product Type:** Cisco 7970

**Association Information**

- Modify Button Items

**Phone User ID:** < None >

**Phone Suite:** Default

**Services Provisioning:** Default

**Phone Lead Name:** Default

**Single Button Barge:** Default

**Join Across Lines:** Default

**Use Trusted Relay Point:** Default

**BTF Audible Alert Setting (Phone Busy):** Default

**BTF Audible Alert Setting (Phone Busy):** Default

**Always Use Prime Line:** Default

**Always Use Prime Line for Voice Messages:** Default

**Calling Party Transformation CSS:** Default

**Greeting:**

---

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Application Note

Protocol Specific Information
- Packet Capture Mode: None
- Packet Capture Duration: 0
- Presence Group: Standard Presence group
- Device Security Profile: Cisco 7970 - Standard SCCP Non-Secure Profile
- SUBSCRIBE Calling Search Space: <none>
- Unattended Port
- Require DTMF Reception
- RFC2833 Disabled

Certification Authority Proxy Function (CAPF) Information
- Certificate Operation: No Pending Operation
- Authentication Mode: By Null String
- Authentication String
- Key Size (Bits): 1024
- Operation Completes By: 2011-01-25 12:00:00

Expansion Module Information
- Module 1 Load Name: <none>
- Module 2 Load Name: <none>

External Data Locations Information (Leave blank to use default)
- Information
- Directory
- Messages
- Services
- Authentication Server
- Proxy Server
- Idle
- Idle Timer (Seconds)
### Extension Information

- **Enable Extension Mobility**: [ ]
- **Log Out Profile**: < None >
- **Log in Time**: < None >
- **Log out Time**: < None >

### MLPP Information

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

### Do Not Disturb

- **Do Not Disturb**: [ ]
- **DND Option**: Ringer Off
- **DND Incoming Call Alert**: < None >

### Secure Shell Information

- **Secure Shell User**: 
- **Secure Shell Password**: 

### Product Specific Configuration Layout

<table>
<thead>
<tr>
<th>Setting</th>
<th>Options</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td>[ ] Disabled</td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td>[ ] Enabled</td>
</tr>
<tr>
<td>Forwarding Delay</td>
<td>[ ] Disabled</td>
</tr>
<tr>
<td>PC Port</td>
<td>[ ] Enabled</td>
</tr>
<tr>
<td>Settings Access</td>
<td>[ ] Disabled</td>
</tr>
<tr>
<td>Gratuitous ARP</td>
<td>[ ] Enabled</td>
</tr>
<tr>
<td>PC Voice VLAN Access</td>
<td>[ ] Disabled</td>
</tr>
<tr>
<td>Video Capabilities</td>
<td>[ ] Disabled</td>
</tr>
<tr>
<td>Auto Line Select</td>
<td>[ ] Enabled</td>
</tr>
<tr>
<td>Web Access</td>
<td>[ ] Enabled</td>
</tr>
<tr>
<td>Days Display Not Active</td>
<td>Sunday, Monday, Tuesday</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>Span to PC Port</td>
<td>[ ] Disabled</td>
</tr>
<tr>
<td>Logging Display</td>
<td>[ ] PC Controlled</td>
</tr>
<tr>
<td>Load Server</td>
<td></td>
</tr>
<tr>
<td>Recording Tone</td>
<td>[ ] Disabled</td>
</tr>
<tr>
<td>Recording Tone Local Volume</td>
<td>100</td>
</tr>
<tr>
<td>Recording Tone Remote Volume</td>
<td>50</td>
</tr>
</tbody>
</table>

---

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<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Recording Tone Duration</td>
<td>Disabled</td>
</tr>
<tr>
<td>Display On When Incoming Call*</td>
<td>Disabled</td>
</tr>
<tr>
<td>RTCP*</td>
<td>E</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 Codec*</td>
<td></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>Peer Firmware Sharing*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDF): Switch Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDF): PC Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td>Unknown</td>
</tr>
<tr>
<td>LLDP Power Priority*</td>
<td></td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td>User Controlled</td>
</tr>
<tr>
<td>802.1X Authentication*</td>
<td>Normal</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure*</td>
<td>0-Silent</td>
</tr>
<tr>
<td>Minimum Ring Volume*</td>
<td>Use Phone Default</td>
</tr>
<tr>
<td>Headset Sidetone Level*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Enable Dialing*</td>
<td></td>
</tr>
</tbody>
</table>

**Note:**
- Enable Key Capture Mode and Packet Capture Duration.
- Settings.
- Administrative service Softkeys must be used for a protected phone.

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Configuring Cisco Skinny IP phone DN

This configurations example for skinny does not include AIN (Remote Destination) configuration, but configuration of a remote destination and device is configured the same as it is configured on a SIP IP phone.

- Directory Number Information
  - Directory Number: 4404
  - Route Partition: < None >
  - Description:
  - Alerting Name:
  - ASCII Alerting Name:
  - Allow Control of Device from OT:
  - Associated Devices: SEP001177122004

- Directory Number Settings
  - Voice Mail Profile: < None >
  - Calling Search Space:
  - Presence Group:
  - User Held MOH Audio Source:
  - Network Held MOH Audio Source:
  - Auto Answer:

- AAR Settings
  - Voice Mail:
  - AAR Destination Mask:
  - AAR Group: < None >

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### Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Call Forward and Call Pickup Settings</th>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Search Space Activation Policy</td>
<td></td>
<td></td>
<td>Use System Default</td>
</tr>
<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>

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

Park Monitoring Retention Timer: A blank value will use value set in Park Monitoring Retention Timer service parameter

### MLPP Alternate Party Settings

<table>
<thead>
<tr>
<th>MLPP Alternate Party Settings</th>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Target (Destination)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>MLPP Calling Search Space</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MLPP No Answer Ring Duration (seconds)</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

A blank value means to call the parker's line.
## Line Settings for All Devices

<table>
<thead>
<tr>
<th>Setting</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hold Reservation Ring Duration (seconds)</td>
<td>Setting the Hold Reservation Ring Duration to zero will disable the feature.</td>
</tr>
<tr>
<td>Hold Reservation Notification Interval (seconds)</td>
<td>Setting the Hold Reservation Notification Interval to zero will disable the feature.</td>
</tr>
<tr>
<td>Party Entrance Tone</td>
<td>Options</td>
</tr>
</tbody>
</table>

## Line 1 on Device SEPO0137F16384A

<table>
<thead>
<tr>
<th>Setting</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display (Internal Caller ID)</td>
<td>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.</td>
</tr>
<tr>
<td>ASCII Display (Internal Caller ID)</td>
<td>Use System Policy</td>
</tr>
<tr>
<td>Line Text Label</td>
<td></td>
</tr>
<tr>
<td>ASCII Line Text Label</td>
<td></td>
</tr>
<tr>
<td>External Phone Number Mask</td>
<td></td>
</tr>
<tr>
<td>Visual Message Waiting Indicator Policy</td>
<td>Use System Policy</td>
</tr>
<tr>
<td>Audible Message Waiting Indicator Policy</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Recording Option</td>
<td>Call Recording Disabled</td>
</tr>
<tr>
<td>Recording Profile</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Monitoring Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

## Multiple Call/Call Waiting Settings on Device SEPO0137F16384A

<table>
<thead>
<tr>
<th>Setting</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Number of Calls*</td>
<td>1-290</td>
</tr>
<tr>
<td>Busy Trigger*</td>
<td>0, 1, 2 (0 is the default, 2 means calls may be held and a new call isRING)</td>
</tr>
</tbody>
</table>

## Forwarded Call Information Display on Device SEPO0137F16384A

<table>
<thead>
<tr>
<th>Setting</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
<tr>
<td>Caller Number</td>
<td></td>
</tr>
<tr>
<td>Redirected Number</td>
<td></td>
</tr>
<tr>
<td>Dialed Number</td>
<td></td>
</tr>
</tbody>
</table>

## Users Associated with Line

- Associate End Users

---

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EDCS#846451 Rev # 3
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:
Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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 select 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: On Hook

Unselected Softkeys

- Conference List (ConfList)
- Direct Transfer (DTrn)
- Group Pick Up (GPickup)
- Hand (Hang)
- Handset (Handset)
- More (More)
- Other Pick Up (OPickup)
- Pause (Pause)
- Quality Report Tool (QRT)
- Restore Last Conference Party (RestCon)
- Select (Select)
- Tangle Do Not Disturb (DNS)
- (Undefined (Undefined))

Selected Softkeys (ordered by position)*

- Redial (Redial)
- ***New Call (NewCall)
- Forward All (ChnlAll)

Under call state select: On hook, then click on the Mobility feature to highlight blue and click the < (more item) button to move Mobility to the "Selected Softkeys" box. Click save.

* Indicates required item.

** Indicates mandatory fields.
Softkey Template Configuration

Softkey Template: Standard User-mobile

Select a call state to configure: Connected

Selected Softkeys (by position)
- Hold (Hold)
- End Call (EndCall)
- Transfer (Transfer)
- Park (Park)
- Conference (Conf)
- Conference List (ConfList)
- Select (Select)
- Join (Join)
- Direct Transfer (DivTfr)
- Wide Mode Command (WidMode)

Unselected Softkeys
- Hold (Hold)
- Immediate Divert (Divert)
- Quality Report (QRT)
- Remove Last Conference Party (RLCP)
- Toggle Do Not Disturb (DND)
- Toggle Holdphone Call Transfer (HTC)
- Undefined (Undefined)

Under call state select "Connected", 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 required item.
- Indicates mandatory fields
**Application Note**

Configuring translation pattern to allow Cisco IP phone user to dial 4-digit to reach 10-digit MOC DN

---

### Translation Pattern Configuration

<table>
<thead>
<tr>
<th>Pattern Definition</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Translation Pattern</td>
<td>5DXX</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Numbering Plan</td>
<td></td>
</tr>
<tr>
<td>Route Filter</td>
<td></td>
</tr>
<tr>
<td>RNP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td></td>
</tr>
<tr>
<td>Calling Search Space</td>
<td></td>
</tr>
<tr>
<td>Route Option</td>
<td></td>
</tr>
</tbody>
</table>

**Status**

- **Status:** Ready

---

### Calling Party Transformations

- Use Calling Party's External Phone Number Mask
- Calling Party Transform Mask
- Prefix Digits (Outgoing Calls)
- Calling Line ID Presentation
- Calling Name Presentation
- Calling Party Number Type
- Calling Party Numbering Plan

---

### Connected Party Transformations

- Connected Line ID Presentation
- Connected Name Presentation

---

### Called Party Transformations

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

---

*indicates required item.
### Configuring Outgoing Route Pattern

**Route Pattern Configuration**

- **Status**: Ready

- **Pattern Definition**
  - **Route Pattern**: 40044556
  - **Description**: 
  - **Numbering Plan**: Not Selected
  - **Route Filter**: None
  - **HUP Precedence**: Default
  - **Resource Priority Namespace Network Domain**: Not Selected
  - **Gateway Route List**: Any-OCIS
  - **Route Override**: Route this pattern, Block this pattern, No Error
  - **Cell Classification**: OFF
  - **Allow Device Override**, **Provide Outside Dial Tone**, **Allow Overlap Sending**, **Urgent Priority**: Disabled
  - **Require Security Authorization Code**, **Authorization Level**: 0
  - **Require Client Security Code**: Enabled

**Calling Party Transformations**
- **Use Calling Party’s External Phone Number Mask**: Added prefix to the caller ID number for callback DD.
- **Prefix Digits (Outgoing Calls)**: +1185050
- **Calling Line ID Presentation**: Default
- **Calling Name Presentation**: Default
- **Calling Party Number Type**: Cisco CallManager
- **Calling Party Numbering Plan**: Cisco CallManager

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

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

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

---

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### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</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>OCS</td>
<td>Microsoft Office Communications Server</td>
</tr>
<tr>
<td>MOC</td>
<td>Microsoft Office Communicator</td>
</tr>
<tr>
<td>SNR</td>
<td>Single number reach</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>SDP</td>
<td>Session Description Protocol</td>
</tr>
</tbody>
</table>
**Important Information**

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Appendix A

Adding and removing digit prefixes (+) with Cisco UBE when using older Cisco UCM versions (6.X, 5.X)

Older Cisco UCM versions do not support the (+) character within its route pattern or translation pattern configuration. But Cisco UBE allows you to manipulate incoming and outgoing digits to add or remove (+).

Use the sample config below to allow for Cisco UBE to add the plus to the called number from the Cisco UCM side instead of using Cisco UCM route pattern and translation pattern. Also, this CUBE config sample does the calling number translation on behalf of the Cisco UCM and translates a 4-digit number dialed from the Microsoft MOC client out to Cisco UCM user.

Cisco UBE config

```plaintext
voice translation-rule 100
  rule 1 /55/ /+140844455\1/
  rule 2 /44/ /15105544\1/

voice translation-rule 101
  rule 1 /^4/ /15105554\1/

voice translation-profile out-OCS
  translate calling 100
  translate called 100

voice translation-profile out-cucm
  translate called 101

dial-peer voice 9999 voip
  description outgoing to MS OCS
  destination-pattern 1510555....
  session protocol sipv2
  session target ipv4:172.20.8.254
  voice-class sip profiles 4
  dtmf-relay rtp-nte

dial-peer voice 5500 voip
  destination-pattern +140844455..
  session protocol sipv2
  session target ipv4:172.20.127.101
  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 99991 voip
  translation-profile incoming out-cucm
  session protocol sipv2
  session transport tcp
  incoming called-number 44..
  voice-class sip profiles 4
  dtmf-relay rtp-nte
  codec g711ulaw

dial-peer voice 55001 voip
```

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translation-profile incoming out-OCS
session protocol sipv2
incoming called-number 55..
dtmf-relay rtp-nte
!
Cisco UCM route pattern config

<table>
<thead>
<tr>
<th>Route Pattern</th>
<th>$30X</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>NLPP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>MSFT-DCS (Edit)</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern, No Error</td>
</tr>
<tr>
<td>Call Classification</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Allow Device Override</td>
<td>Provide Outside Dial Tone</td>
</tr>
<tr>
<td>Require Forced Authorization Code</td>
<td>Allow Overlap Sending</td>
</tr>
<tr>
<td>Authorization Level</td>
<td>0</td>
</tr>
<tr>
<td>Require Client Master Code</td>
<td></td>
</tr>
</tbody>
</table>

Simple 4-digit route pattern now

---

**Cisco Unified CM Administration**

- **System**
  - Call Routing
  - Media Resources
  - Voice Mail
  - Device
  - Application
  - User Management
  - Bulk Administration
- **Help**

**Route Pattern Configuration**

- **Status**
  Status: Ready

**Connected Party Transforms**

<table>
<thead>
<tr>
<th>Connected Line ID Presentation</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
</tbody>
</table>

**Called Party Transforms**

- **Discard Digits**
- **Called Party Transform Mask**
| < None > |
| < None > |

**ISDN Network-Specific Facilities Information Element**

| Network Service Protocol | < Not Selected -- |
| Carrier Identification Code | < Not Selected -- |

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