

Configuration notes

Ascom i62 with Cisco Unified Communications Manager Cisco

This document describes a general configuration of Cisco Unified Communications Manager and Ascom i62.

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CUCM configurations

Cisco Unified Communications Manager (CUCM), version 12.0 configuration

- Caller Line Identities (CLI) require additional configuration
- CUCM license for “Third-party SIP device” implies some limitations, e.g. no Music-on-Hold (MoH) and lack of telephony features configurable from the handset etc.

Phone Configuration

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

Association

[Modify Button Items](#)

- 1 [Line \[1\] - 6500 \(no partition\)](#)
- 2 [Line \[2\] - Add a new DN](#)
- 3 [Line \[3\] - Add a new DN](#)
- 4 [Line \[4\] - Add a new DN](#)
- 5 [Line \[5\] - Add a new DN](#)
- 6 [Line \[6\] - Add a new DN](#)
- 7 [Line \[7\] - Add a new DN](#)
- 8 [Line \[8\] - Add a new DN](#)

Phone Type

Product Type: Third-party SIP Device (Advanced)
Device Protocol: SIP

Real-time Device Status

Registration: Registered with Cisco Unified Communications Manager hq-cucm-pub.abc.inc
IPv4 Address: 192.168.124.10
Active Load ID: None
Download Status: None

Device Information

Device is Active
 Device is not trusted

MAC Address*

Description

Device Pool* [View Details](#)

Common Device Configuration [View Details](#)

Phone Button Template*

Common Phone Profile* [View Details](#)

Calling Search Space

AAR Calling Search Space

Media Resource Group List

Location*

AAR Group

Device Mobility Mode* [View Current Device Mobility Settings](#)

Owner User Anonymous (Public/Shared Space)

Owner User ID

Use Trusted Relay Point*

Always Use Prime Line*

Always Use Prime Line for Voice Message*

Geolocation

Retry Video Call as Audio

Ignore Presentation Indicators (internal calls only)

Logged Into Hunt Group

Remote Device

Device->Phone: Adding a device (phone). Part 1.

Number Presentation Transformation	
Caller ID For Calls From This Phone	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)	
Remote Number	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)	
Protocol Specific Information	
BLF Presence Group*	Standard Presence group
MTP Preferred Originating Codec*	711ulaw
Device Security Profile*	3rd_Party_SIP_Advanced_Secure_Authenticated
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile 2 View Details
Digest User	6500 <input type="button" value="Find"/>
<input type="checkbox"/> Media Termination Point Required <input type="checkbox"/> Unattended Port <input type="checkbox"/> Require DTMF Reception <input type="checkbox"/> Allow Presentation Sharing using BFCP <input type="checkbox"/> Allow iX Applicable Media	
MLPP and Confidential Access Level Information	
MLPP Domain	< None >
Confidential Access Mode	< None >
Confidential Access Level	< None >

Device->Phone: Adding a device (phone). Part 2

Note. Digest User (6500) refers to the End User created in next step.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

End User Configuration

Save
 Delete
 Add New

Status

Status: Ready

User Information

User Status	Enabled Local User	
User ID*	<input type="text" value="6500"/>	
Password	<input type="password" value="....."/>	<input type="button" value="Edit Credential"/>
Confirm Password	<input type="password" value="....."/>	
Self-Service User ID	<input type="text" value="6500"/>	
PIN	<input type="password" value="....."/>	<input type="button" value="Edit Credential"/>
Confirm PIN	<input type="password" value="....."/>	
Last name*	<input type="text" value="6500"/>	
Middle name	<input type="text"/>	
First name	<input type="text"/>	
Display name	<input type="text"/>	
Title	<input type="text"/>	
Directory URI	<input type="text"/>	
Telephone Number	<input type="text"/>	
Home Number	<input type="text"/>	
Mobile Number	<input type="text"/>	
Pager Number	<input type="text"/>	
Mail ID	<input type="text"/>	
Manager User ID	<input type="text"/>	
Department	<input type="text"/>	
User Locale	< None > ▾	
Associated PC	<input type="text"/>	
Digest Credentials	<input type="text"/>	
Confirm Digest Credentials	<input type="text"/>	
User Profile	Use System Default("Standard (Factory Defau" ▾	View Details

Service Settings

Home Cluster

Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)

Include meeting information in presence(Requires Exchange Presence Gateway to be configured on CUCM IM and Presence serv

UC Service Profile ▾ [View Details](#)

User Management -> End User: Adding an user ID

Note. Digest Credentials is only used if "Enable Digest Authentication" is set in the security profile.

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the Cisco logo and 'Cisco Unified CM Administration' title are visible, along with the tagline 'For Cisco Unified Communications Solutions'. A navigation menu includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', and 'User Manager'. The main heading is 'Phone Security Profile Configuration'. Below this, there are action buttons: 'Copy', 'Reset', 'Apply Config', and 'Add New'. The 'Status' section shows 'Status: Ready'. The 'Phone Security Profile Information' section contains the following fields: 'Product Type' (Third-party SIP Device (Advanced)), 'Device Protocol' (SIP), 'Name*' (Third-party SIP Device Advanced - Standard SIP Non-Secure Profi), 'Description' (Third-party SIP Device (Advanced) - Standard SIP Non-Secure Pr), 'Nonce Validity Time*' (600), 'Transport Type*' (TCP+UDP), and an unchecked checkbox for 'Enable Digest Authentication'. The 'Parameters used in Phone' section shows 'SIP Phone Port*' (5060). At the bottom, there are buttons for 'Copy', 'Reset', 'Apply Config', and 'Add New', followed by an information icon and the text '*- indicates required item.'

System->Security->Security Profiles.

" Third-party SIP Device Advanced - Standard SIP Non-Secure Profile" default security profile.

Ascom i62 configurations

Device type:

Parameter definition:

Name	Value	
Network name	Cisco	?
DHCP mode	On	?
802.11 protocol	802.11a/n	?
SSID	linksys_SE5_62236	?
Security mode	WPA-PSK & WPA2-PSK	?
WPA-PSK passphrase	*****	?
Voice power save mode	U-APSD	?
802.11a/n channels	Non DFS	?
Advanced: 802.11 channels		?
World mode regulatory domain	World mode (802.11d)	?
Transmission power	Automatic	?
IP DSCP for voice	0x2E (46) - Expedited Forwar...	?
IP DSCP for signaling	0x1A (26) - Assured Forwardi...	?
TSPEC Call Admission Control	Off	?
Transmit gratuitous ARP	No	?
Deauthenticate on roam	No	?
Roaming methodology	802.11 roaming	?
Maximum transfer unit	1400	?
Aruba 800 controller compability	No	?
Check IP connectivity after ro...	No	?

Note. The network settings may vary depending on the WLAN infrastructure used.

Device type:

Parameter definition:

<ul style="list-style-type: none"> <input type="checkbox"/> Network <input type="checkbox"/> Device <input type="checkbox"/> Audio <input type="checkbox"/> Presence <input type="checkbox"/> Location <input checked="" type="checkbox"/> VoIP <ul style="list-style-type: none"> <input type="radio"/> General <input type="radio"/> H.323 <input checked="" type="radio"/> SIP <input type="checkbox"/> Customization <input type="checkbox"/> Services <input type="checkbox"/> Alarm <input type="checkbox"/> Push-To-Talk <input type="checkbox"/> Headset <input type="checkbox"/> User Profiles <input type="checkbox"/> System Profiles <input type="checkbox"/> Shortcuts 	<table border="1"> <thead> <tr> <th>Name</th> <th>Value</th> <th></th> </tr> </thead> <tbody> <tr><td>SIP Transport</td><td>UDP</td><td>?</td></tr> <tr><td>Outbound proxy mode</td><td>No</td><td>?</td></tr> <tr><td>Primary SIP proxy</td><td>10.71.3.11</td><td>?</td></tr> <tr><td>Secondary SIP proxy</td><td>10.71.3.12</td><td>?</td></tr> <tr><td>Listening port</td><td>5060</td><td>?</td></tr> <tr><td>SIP proxy ID</td><td></td><td>?</td></tr> <tr><td>SIP proxy password</td><td>*****</td><td>?</td></tr> <tr><td>Send DTMF using RFC 2833 or...</td><td>RFC2833</td><td>?</td></tr> <tr><td>Hold type</td><td>Inactive</td><td>?</td></tr> <tr><td>Registration identity</td><td>Endpoint number</td><td>?</td></tr> <tr><td>Authentication identity</td><td>Endpoint number</td><td>?</td></tr> <tr><td>Call forward locally</td><td>No</td><td>?</td></tr> <tr><td>MOH locally</td><td>Yes</td><td>?</td></tr> <tr><td>Hold on Transfer</td><td>No</td><td>?</td></tr> <tr><td>Direct signaling</td><td>No</td><td>?</td></tr> <tr><td>SIP Register Expiration</td><td>3600</td><td>?</td></tr> <tr><td>SIP Message behavior</td><td>Ignore</td><td>?</td></tr> </tbody> </table>	Name	Value		SIP Transport	UDP	?	Outbound proxy mode	No	?	Primary SIP proxy	10.71.3.11	?	Secondary SIP proxy	10.71.3.12	?	Listening port	5060	?	SIP proxy ID		?	SIP proxy password	*****	?	Send DTMF using RFC 2833 or...	RFC2833	?	Hold type	Inactive	?	Registration identity	Endpoint number	?	Authentication identity	Endpoint number	?	Call forward locally	No	?	MOH locally	Yes	?	Hold on Transfer	No	?	Direct signaling	No	?	SIP Register Expiration	3600	?	SIP Message behavior	Ignore	?
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Document history

Rev	Date	Author	Description
RA1	2017-01-16	SEJAn	Final Version, Updated with CUCM12 and i62 6.0.1