Office 365 Exchange UM using SIP (TLS) trunk to Cisco ASR 1000 IOS 15.4(3) S3 and Cisco UCM Release 10.5.1
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**Introduction**

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 10.5.1 and Cisco ASR 1004 to interoperate with the Office 365 using SIP (TLS).

The following items were tested:

- Basic call between the two systems and verification of voice path, using both SIP on the Office 365 side, and SIP on the Cisco side
- Voice messaging and MWI activation-deactivation
- Call transfer from Office 365 Exchange UM to extensions passing through ASR
- FAX

Below are the key results:

- Basic call to VM, call transfer to extensions, MWI work successfully
Network Topology

Basic Call Setup
Limitations
These are the known limitations, caveats, or integration issues:

- None

System Components

Hardware Tested
The following hardware was tested:

- Cisco UCS-C240-M3S VMWare Host
- Cisco 7970, and 7975 IP phones and IP communicator
- Cisco ASR 1004

Software Tested
The following software was tested:

- Cisco UCSC-C240-M3S VMware vSphere Image Profile: ESXi-5.5.0-1331820-standard
- Cisco Unified Communications Manager release 10.5.1.10000-7
- Cisco ASR 1000 running IOS 15.4(3)S3
- Office 365 Exchange UM online

Features
This section lists supported and unsupported features. No deviation from the configuration presented in this document will be supported by Cisco.

Features Supported

- MWI—Message Waiting Indicator (lamp ON, lamp OFF)
- FAX

Features Not Supported or Not Tested

- Coexistence between Exchange On Premise and Office 365
Configuration

The goal of this guide is to provide an overview of the integration between Cisco Unified Communication Manager, Cisco ASR 1004 and Office 365 Exchange UM. The deployment will interconnect the UC systems using SIP. The following sections provide the required configurations for a successful integration.

Configuring Sequence and Tasks:

Office 365:
Configure the users’ mailbox.
Configure the trunk to Cisco ASR.
Configure the dial plan pointing to Cisco ASR.
Configure the Pilot Number.
Configure the Auto Attendant

Cisco Unified Communications Manager:

1. SIP trunk security profile
2. Device setting SIP profile
3. SIP trunk to Cisco ASR
4. Voice Mail Profile
5. Voice Mail Pilot
6. Route pattern to the Office 365 Exchange UM via Cisco ASR
7. SIP and SCCP phones device configuration
8. Directory Number configuration
9. Region and device pool Configuration

Cisco ASR 1004:
Configure the Dial peers pointing to Cisco UCM.
Configure the Dial peers pointing to Office 365 Exchange UM.
Install public certificates.
Configuring Office 365 Exchange UM

On Office 365 portal go to Admin menu and select Exchange.

Click on UM IP gateway
UM IP Gateway

Add the SBC as a gateway

Exchange admin center

Fill as required and save. Leave UM Dial plan blank

new UM IP gateway

UM IP gateways represent a physical session border controller (SBC), IP gateway, or IP PBX in Active Directory. You have to configure a UM IP gateway before UM can accept calls from the device.

*Name:

CiscoCUBE

*Address:

asr1034.tekvisionlabs.com

UM dial plan:

Browse...
Go back to Exchange Admin Center

Under unified messaging select UM Dial Plan
UM Dial Plan

Click on + to add a dial plan

Exchange admin center

Fill as needed and save

new UM dial plan

Learn more

*Name:

CUCM

*Extension length (digits):

4

*Dial plan type:

Telephone extension

*VoIP security mode:

Unsecured

*Audio language:

English (United States)

*Country/Region code:

1
Double click on the dial plan just created

**UM dial plans**  UM IP gateways

Unified Messaging dial plans define the format for telephone numbers in your organization. For UM to answer calls for your users, you have to set up at least one dial plan. Learn more

<table>
<thead>
<tr>
<th>NAME</th>
<th>EXTENSION LENGTH</th>
<th>URI TYPE</th>
</tr>
</thead>
<tbody>
<tr>
<td>auto-5</td>
<td>5</td>
<td>Telephone extension</td>
</tr>
<tr>
<td>CUCM</td>
<td>4</td>
<td>Telephone extension</td>
</tr>
<tr>
<td>tvExchangeUM</td>
<td>4</td>
<td>SIP URI</td>
</tr>
</tbody>
</table>

Auto Attendant

Add an Auto Attendant

**UM Auto Attendants**

Fill as needed and click save

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Page 12 of 65
new UM auto attendant

UM auto attendants allow you to automatically answer and route calls for your organization.

UM dial plan:
CUCM

*Name:
CUCM-AA

☑ Create this auto attendant as enabled
☑ Set the auto attendant to respond to voice commands

Access numbers:

[Table with access numbers]

These are the access numbers that callers dial to reach this UM auto attendant. For E.164 and SIP dial plans, the numbers must be in E.164 number format.

After you click Save, select this UM auto attendant and click Details to set greetings, business hours.

Save  Cancel

UM Hunt Group

Add a UM Hunt Group

UM Hunt Groups

[Table with UM Hunt Groups]
Fill as needed, select the UM IP Gateway created previously and save

new UM hunt group

UM hunt groups determine which UM IP gateways to accept calls from for the users of this UM dial plan.

UM dial plan:

CUCM

*Name:
VoiceMail

*UM IP gateway:
CustomerSBC [Browse...]

Pilot identifier:
2401
Dialing Rules

To configure dialing rules click on Configure at the top of UM Dial Plan window

CUCM

Dial plan type:
Extension length (digits):

Telephone extension

4

To configure dial codes, Outlook Voice Access, voice mail settings, and dialing rules for this dial plan, click Configure.

Select dialing rules

CUCM

diary rules

In-country/region dialing rules:

<table>
<thead>
<tr>
<th>GROUP NAME</th>
<th>NUMBER PATTERN</th>
<th>DIALED NUMBER</th>
</tr>
</thead>
</table>

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Fill as needed and Save

PSTN

Dialing rules change the number dialed by users to the number UM dials for outgoing calls. For example, you can add a 9 to the dialed number to access an outside line. Use the same group name to add several rules to a group, which you can enable and disable together. For example, you may want to automatically add the outside line access code to dialed numbers.

*Dialing rule name:

PSTN

*Number pattern to transform (number mask):

*

*Dialed number:

*

Comment:
CUCM
dialing rules
dialing authorization
transfer & search

Specify dialing rules to control the types of calls users can make. For rules to take effect, authorize them in the dial plan, UM mailbox policies, and UM auto attendants.

In-country/region dialing rules:

<table>
<thead>
<tr>
<th>GROUP NAME</th>
<th>NUMBER PATTERN</th>
<th>DIALED NUMBER</th>
</tr>
</thead>
<tbody>
<tr>
<td>PSTN</td>
<td>*</td>
<td>*</td>
</tr>
</tbody>
</table>

International dialing rules:

<table>
<thead>
<tr>
<th>GROUP NAME</th>
<th>NUMBER PATTERN</th>
<th>DIALED NUMBER</th>
</tr>
</thead>
</table>

A dialing rule defines types of calls that users can make. You can group rules by giving them the same group name. After you define the rules here, go to Dialing Authorization on your dial plan, UM mailbox policies, or auto attendants to authorize the rule groups for callers.

Dialing Authorization

CUCM
dialing rules
dialing authorization
transfer & search

Select the types of calls to authorize for users of this UM dial plan.

- [ ] Calls in the same UM dial plan
- [ ] Allow calls to any extension

Authorized in-country/region dialing rule groups:

<table>
<thead>
<tr>
<th>NAME</th>
</tr>
</thead>
</table>

Allow unauthenticated Outlook Voice Access users to call or transfer to extensions for non-UM-enabled users in your organization.
Configure settings for this dial plan, including UM mailbox policies, auto attendants, and hunt groups.

**UM Dial Plan**
- Name: CUCM
- Dial plan type: Telephone extension
- Extension length (digits): 4

To configure dial codes, Outlook Voice Access, voice mail settings, and dialing rules for this dial plan, click Configure.

**UM Mailbox Policies**
- NAME: CUCM Default Policy
  - MINIMUM PIN LENGTH: 6

1 selected of 1 total

**UM Auto Attendants**
- NAME: CUCM-AA
  - STATUS: Enabled

1 selected of 1 total

**UM Hunt Groups**
- NAME: VoiceMail
  - UM IP GATEWAY: CustomerSBC
  - PILOT IDENTIFIER: 2401
FAX configuration

Under UM IP dial plans select the dial plan “CUCM” (configured on previous section) SBC and click on edit.

Under UM Mailbox Policy select the policy and click edit.
Check the checkbox “Allow inbound faxes” and fill the Partner fax server URI with the IP address/FQDN of the fax server including the transport type (TCP/UDP)
Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager SIP Trunk Security Profile

**Navigation:** System → Security → SIP trunk security profile

Set Name* = Non Secure SIP Trunk Profile. This is used for this example.
Set Description = Non Secure SIP Trunk Profile authenticated by null String
Check Accept out of dialog refer
Check Accept unsolicited notification
Check Accept replaces header
All other values are default.
Cisco Unified Communications Manager SIP Profile

**Navigation:** Device → Device Settings → SIP Profile

Standard SIP Profile. This is used for this example.

All values are default.
Cisco Unified Communications Manager SIP Profile ( Continued )

These values are default.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>32766</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>TELnet Port for 7940 and 7965</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceuri-cfwdall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-abbrevdial</td>
</tr>
</tbody>
</table>

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### Cisco Unified Communications Manager SIP Trunk to Cisco ASR

**Navigation:** Device → Trunk
Set Device Name* = CiscoASR_Office365ExchangeUM. This is used for this example.
Set Description = this text is used to identify this Trunk Group.
Set Device Pool* = G711 Device Pool. This is used for this example.
Set Call Classification* = Use System Default. This is used for this example.
All other values are default.

<table>
<thead>
<tr>
<th>Device Information</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name*</td>
<td>CiscoASR_Office365ExchangeUM</td>
</tr>
<tr>
<td>Device Pool*</td>
<td>G711 Device Pool</td>
</tr>
<tr>
<td>Call Classification*</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

Cisco Unified Communications Manager SIP Trunk to Cisco ASR (Continued)
Cisco Unified Communications Manager SIP Trunk to Cisco ASR (Continued)
Cisco Unified Communications Manager SIP Trunk to Cisco ASR (Continued)

Set Destination Address = 10.64.3.137. This is used in this example.

Set SIP Trunk Security Profile* = Non Secure SIP Trunk Profile

Set SIP Profile* = Standard SIP Profile

Set DTMF Signaling Method* = RFC2833

All other values are default.
Voice Mail Pilot

**Navigation:** Advanced Features → Voice Mail → Voice Mail Pilot

Set Mail Pilot Number = 2401. This is used for this example.

Set Description = O365 Exchange UM VM Pilot. This is used for this example.

---

### Voice Mail Pilot Configuration

<table>
<thead>
<tr>
<th>Status</th>
<th>Update successful</th>
</tr>
</thead>
</table>

#### Voice Mail Pilot Information

- **Mail Pilot Number:** 2401
- **Calling Search Space:** < Name >
- **Description:** O365 Exchange UM VM Pilot

Check the box to make this the default Voice Mail Pilot for the system.
Voice Mail Profile

**Navigation:** Advanced Features ➔ Voice Mail ➔ Voice Mail Profile

Set Device Name* = 7235552401. This is used for this example.

Set Description = O365 Exchange UM Profile.
Route Pattern for Pilot Number

**Navigation:** Call Routing → Route/Hunt → Route Pattern

Set Route Pattern* = 2401. This is used to route to Cisco ASR for this example. This number was configured on Office 365 Exchange UM as Pilot Number.

Set Gateway/Route List = Cube_Office365. This is the trunk to ASR.
Route Pattern to ASR SIP Trunk (Continued)

Set Calling Line ID Presentation= Default
Set Calling Name Presentation= Default
Set Connected Line ID Presentation= Default
Set Connected Name Presentation= Default
Cisco Unified Communications Manager SCCP Phone Device Level Configuration

**Navigation Path:** Device ➔ Phone

Set MAC Address* = 001D45E95CD4. This is used in this example

Set Description = 2670 Office 365 Exchange UM. This text is used to identify this Phone

Set Device Pool* = Default. This is used in this example

Set Phone Button Template* = Standard 7975 SCCP. This is used in this example

Common Phone Profile* = Standard Common Phone Profile
Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

All other values are default.
Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

Set Device Security Profile* = Cisco 7975- Standard SCCP Non-Secure Profile. This is used in this example.

All other values are default.
Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

All other values are default.

| External Data Locations Information (Leave blank to use default) |
|-----------------|-----------------|
| Information     | Directory       |
|                 | Messages        |
|                 | Services        |
|                 | Authentication Server |
|                 | Proxy Server    |
| Idle            | Idle Timer (seconds) |
|                 | Secure Authentication URL |
|                 | Secure Directory URL |
|                 | Secure Idle URL  |
|                 | Secure Information URL |
|                 | Secure Messages URL |
|                 | Secure Services URL |

<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 and Confidential Access Level Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP Domain</td>
</tr>
<tr>
<td>MLPP Indication</td>
</tr>
<tr>
<td>MLPP Preemption</td>
</tr>
<tr>
<td>Confidential Access Mode</td>
</tr>
<tr>
<td>Confidential Access Level</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Do Not Disturb</th>
</tr>
</thead>
<tbody>
<tr>
<td>Do Not Disturb</td>
</tr>
<tr>
<td>DNO Option</td>
</tr>
<tr>
<td>DNO 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>
</tbody>
</table>
Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

All values are default.
Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

All values are default.

<table>
<thead>
<tr>
<th>Configuration Item</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>RTCP*</td>
<td>Disabled</td>
</tr>
<tr>
<td>&quot;Inore&quot; Soft Key Timer</td>
<td>Enabled</td>
</tr>
<tr>
<td>Auto Call Select*</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Log Server</td>
<td>Enabled</td>
</tr>
<tr>
<td>Advertise G.722 Codec*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset UT Control*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Peer Firmware Sharing*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): 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>Disabled</td>
</tr>
<tr>
<td>Wireless Headset Hookswitch Control*</td>
<td>Disabled</td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td>Disabled</td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td>Disabled</td>
</tr>
<tr>
<td>802.1x Authentication*</td>
<td>User Controlled</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure*</td>
<td>Normal</td>
</tr>
<tr>
<td>Minimum Ring Volume*</td>
<td>D-Silent</td>
</tr>
<tr>
<td>Headset Sidetone Level*</td>
<td>Default</td>
</tr>
<tr>
<td>Headset Band Gain*</td>
<td>Default</td>
</tr>
<tr>
<td>HTTPS Server*</td>
<td>http and https Enabled</td>
</tr>
<tr>
<td>Headset/headset Monitor*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Headset Recording*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Enable Dialing*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Switch Port Remote Configuration*</td>
<td>Disabled</td>
</tr>
<tr>
<td>PC Port Remote Configuration*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Automatic Port Synchronization*</td>
<td>Disabled</td>
</tr>
<tr>
<td>SSH Access*</td>
<td>Disabled</td>
</tr>
<tr>
<td>LOGIN Access*</td>
<td>Disabled</td>
</tr>
<tr>
<td>FIPS Mode*</td>
<td>Disabled</td>
</tr>
<tr>
<td>80-bit SRTP*</td>
<td>Disabled</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Phone Device Level Configuration

**Navigation Path:** Device → Phone

Set MAC Address* = FCFBFBCA22FE. This is used in this example.

Set Description = 2672 Office 365 Exchange UM. This text is used to identify this Phone

Set Device Pool* = Default. This is used in this example.

Set Phone Button Template* = Standard 7975 SIP. This is used in this example

All other values are default.

---

<table>
<thead>
<tr>
<th><strong>Device Information</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Device Address</strong></td>
<td>FCFBFBCA22FE</td>
</tr>
<tr>
<td><strong>Device Protocol</strong></td>
<td>SIP</td>
</tr>
<tr>
<td><strong>Device Pool</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Phone Button Template</strong></td>
<td>Standard 7975 SIP</td>
</tr>
<tr>
<td><strong>Softkey Template</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Common Phone Profile</strong></td>
<td>Standard Common Phone Profile</td>
</tr>
<tr>
<td><strong>AAR Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>AAR Group</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>User Locale</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Network Locale</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Built In Bridge</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Privacy</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Device Mobility Mode</strong></td>
<td>Default</td>
</tr>
</tbody>
</table>

---

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Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

All other values are default.

![Cisco Unified CM Administration interface](image-url)
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

Set Device Security Profile = Cisco 7975- Standard SIP Non-Secure Profile. This is used in this example.

Set SIP Profile = Standard SIP Profile

All other values are default.
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

All other values are default.
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

All other values are default.

![Product Specific Configuration Layout](image)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Forwarding Delay</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Port</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Settings Access</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Gratuitous ARP</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Voice VLAN Access</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Auto Line Select</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Web Access</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Days Display Not Active</td>
<td>Monday</td>
<td>Tuesday</td>
</tr>
<tr>
<td>Display On Time</td>
<td>07:30</td>
<td></td>
</tr>
<tr>
<td>Display On Duration</td>
<td>10:30</td>
<td></td>
</tr>
<tr>
<td>Display Idle Timeout</td>
<td>01:30</td>
<td></td>
</tr>
<tr>
<td>Span to PC Port</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Logging Display</td>
<td>PC Controlled</td>
<td></td>
</tr>
<tr>
<td>Load Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Recording Tone</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Local Volume</td>
<td>100</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Remote Volume</td>
<td>50</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Duration</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Display On When Incoming Call</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>RTCP</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>&quot;more&quot; Soft Key Timer</td>
<td>5</td>
<td></td>
</tr>
<tr>
<td>Auto Call Select</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Advertise G.722 Codec</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset UI Control</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset</td>
<td>Enabled</td>
<td></td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

All other values are default.

![Cisco Unified CM Administration](image)

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Peer Firmware Sharing</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media</td>
<td>Enabled</td>
</tr>
<tr>
<td>Endpoint Discover (LLDP-MED): Switch Port</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority</td>
<td>Unknown</td>
</tr>
<tr>
<td>Wireless Headset</td>
<td>Disabled</td>
</tr>
<tr>
<td>Hookswitch Control</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>User Controlled</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure</td>
<td></td>
</tr>
<tr>
<td>Minimum Ring Volume</td>
<td>0-Silent</td>
</tr>
<tr>
<td>Headset Sidetone Level</td>
<td>Default</td>
</tr>
<tr>
<td>Headset Send Gain</td>
<td>Default</td>
</tr>
<tr>
<td>HTTPS Server</td>
<td>http and https enabled</td>
</tr>
<tr>
<td>Headset/Headset Monitor</td>
<td>Enabled</td>
</tr>
<tr>
<td>Headset Recording</td>
<td>Disabled</td>
</tr>
<tr>
<td>Switch Port Remote Configuration</td>
<td>Disabled</td>
</tr>
<tr>
<td>PC Port Remote Configuration</td>
<td>Disabled</td>
</tr>
<tr>
<td>Automatic Port Synchronization</td>
<td>Disabled</td>
</tr>
<tr>
<td>SSH Access</td>
<td>Disabled</td>
</tr>
<tr>
<td>LOGIN Access</td>
<td>Enabled</td>
</tr>
<tr>
<td>60-bit SRTP</td>
<td>Disabled</td>
</tr>
</tbody>
</table>
**Cisco Unified Communications Manager Directory Number Level Configuration**

**Navigation Path:** Device → Phone → select the associated Line or

Set Voice Mail Profile* = 7235552401. This is used in this example.

All other values are default.

<table>
<thead>
<tr>
<th>Directory Number Information</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Directory Number</strong></td>
<td>2672</td>
</tr>
<tr>
<td><strong>Route Partition</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>2672 Office 365 Exchange UU</td>
</tr>
<tr>
<td><strong>Alerting Name</strong></td>
<td>2672</td>
</tr>
<tr>
<td><strong>Direct Inward Dial</strong></td>
<td>2672</td>
</tr>
<tr>
<td><strong>External Call Control Profile</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Allow Control of Device from CTI</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Associated Devices</strong></td>
<td>SEIFC9FB4CA82FE</td>
</tr>
<tr>
<td><strong>Dissociate Devices</strong></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Directory Number Settings</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Voice Mail Profile</strong></td>
<td>7235552401</td>
</tr>
<tr>
<td><strong>Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>BLF Presence Group</strong></td>
<td>Standard Presence group</td>
</tr>
<tr>
<td><strong>User Hold MOS Audio Source</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Network Hold MOS Audio Source</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Auto Answer</strong></td>
<td>Auto Answer Off</td>
</tr>
</tbody>
</table>

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Cisco Unified Communications Manager Directory Number Level Configuration (Continued)

Check Voice Mail checkbox for forwarding calls to Voice mail as required

All other values are default.
Cisco Unified Communications Manager Directory Number Level Configuration (Continued)

All other values are default

![Cisco Unified CM Administration Interface](image)

- **Park Monitoring**
  - **Voice Mail**
  - **Destination**
  - **Calling Search Space**
    - Park Monitoring
    - Forward No
    - Retrieve
    - Destination
    - External
    - Park Monitoring
    - Forward No
    - Retrieve
    - Destination
    - Internal
    - Park Monitoring Reversion
    - Timer

- **MLPP Alternate Party And Confidential Access Level Settings**
  - **Target (Destination)**
  - **MLPP Calling Search Space**
  - **MLPP No Answer Ring Duration (seconds)**
  - **Confidential Access Mode**
  - **Confidential Access Level**
  - **Call Control Agent Profile**

- **Line Settings for All Devices**
  - **Hold Reversion Ring Duration (seconds)**
  - **Hold Reversion Notification Interval (seconds)**
  - **Party Entrance Tone**
Cisco Unified Communications Manager Directory Number Level Configuration (Continued)

All other values are default.

Cisco Unified Communications Manager Region Configuration

Navigation Path: System → Region Information → Region

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Default, G711, G729, G722 and G722 created in this example.

Default is used for this testing

All other values are default.
Cisco Unified Communications Manager Region Configuration (Continued)

Set Name* = Default. This is used in this example

Set Region= Default. This is used in this example

Set Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11). This is used in this example

Set Region=Default. This is used in this example

All other values are default
Cisco Unified Communications Manager Device Pool Configuration

**Navigation Path:** System → Device Pool

Default used in this example.

All other values are default.
Cisco Unified Communications Manager Device Pool Configuration (Continued)

Set Device Pool Name* = Default. This is used in this example.

Set Cisco Unified Communications Manager Group* = Default

Set Date/Time Group* = CMLocal

Set Region* = Default. This is used in this example

All other values are default.
Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.
Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.
Cisco ASR Configuration

Call Routing to Office 365 Exchange UM

dial-peer voice 100 voip
description CUCM2-CUBE-O365 ExchUM - Incoming
session protocol sipv2
session transport tcp
incoming called-number 2401
voice-class codec 4 offer-all
dtmf-relay rtp-n-te
no vad
!
dial-peer voice 200 voip
description CUCM2-CUBE-O365 ExchUM - Outgoing
destination-pattern 2401
session protocol sipv2
session target dns:e3a46007-31cb-4529-b8cc-1e59b97ebdbd.um.outlook.com
session transport tcp tls
voice-class codec 4 offer-all
voice-class sip call-route url
dtmf-relay rtp-n-te
srtp fallback
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 300 voip
description O365- ExchUM-ASR-CUCM - Incoming
session protocol sipv2
session transport tcp tls
incoming called-number .%
voice-class codec 4 offer-all
voice-class sip profiles 1
voice-class sip pass-thru subscribe-notify-events all
dtmf-relay rtp-n-te
srtp fallback
no vad
!
dial-peer voice 301 voip
description O365- ExchUM-ASR to CUCM Outgoing
destination-pattern 26..
session protocol sipv2
session target ipv4:10.71.2.12
voice-class codec 4 offer-all
dtmf-relay rtp-n-te
no vad
!
dial-peer voice 350 voip
description O365- ExchUM-ASR-CUCM-PSTN
destination-pattern 1........
session protocol sipv2
session target ipv4:10.71.2.12
session transport udp
voice-class codec 4 offer-all
dtmf-relay rtp-n-te
no vad
!

**TLS Setup**

crypto pki trustpoint godaddy.root
enrollment terminal
revocation-check none
!
crypto pki trustpoint godaddy.trustpoint
tenrollement terminal
fqdn asr1004.tekvizionlabs.com
subject-name CN=asr1004.tekvizionlabs.com
revocation-check crl
rsakeypair GDKey

crypto pki certificate chain godaddy.root
certificate ca 00
  30820400 308202E8 A0030201 02020100 300D0609 2A864886 F70D0101 05050030
*Certificate chain omitted*
quith
crypto pki certificate chain godaddy.trustpoint
certificate 27EA0B186F1D91
  30820569 30820451 A0030201 02020727 EA0B186F 1D91300D 06092A86 4886F70D
*Certificate chain omitted*
quith
certificate ca 0301
  308204DE 308203C6 A0030201 02020203 01300D06 092A8648 86F70D01 01050500
*Certificate chain omitted*
quith
...
sip-ua
crypto signaling default trustpoint godaddy.trustpoint
ASR Running configuration

asr1004#show running-config
Building configuration...

Current configuration : 25423 bytes
!
version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
service sequence-numbers
no platform punt-keepalive disable-kernel-core
!
hostname asr1004
!
boot-start-marker
boot system flash bootflash:asr1000rp2-adventerprisek9.03.13.03.S.154-3.53-ext.bin
boot system flash bootflash:asr1000rp2-adventerprisek9.03.10.04.S.153-3.54-ext.bin
boot-end-marker
!
aqm-register-fnf
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
logging buffered 10000000
no logging console
no logging monitor
enable password solutions
!
no aaa new-model
!
!
ip domain name tehvizationlabs.com
ip host EXCHANGE.lynclabsp.local 10.64.3.105
ip host fax01.lynclabsp.local 10.85.0.173
ip name-server 10.85.0.232
!
!
subscriber templating
!
multilink bundle-name authenticated

voice rtp send-receive

voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
no notify redirect ip2ip
allow-connections sip to sip
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
supplementary-service media-renegotiate
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
sip
rel1xx disable
min-se 90 session-expires 90
early-offer forced
midcall-signaling passthrue
pass-thru headers unsupp
pass-thru subscribe-notify-events all

voice class uri 10 sip
pattern 10.85.0.173

voice class codec 2
codec preference 1 g711ulaw

voice class codec 4
codec preference 1 g711ulaw
codec preference 2 g711alaw
codec preference 3 g729r8

voice class sip-profiles 100
request INVITE sip-header Diversion copy "<sip:(.*)@" u01
request INVITE sip-header To modify "<sip:(.*)" "<sip:\u01@\"1"

media class 1

crypto pki trustpoint godaddy.root
enrollment terminal
revocation-check none
crypto pki trustpoint godaddy.trustpoint
enrollment terminal
fqdn asr1004.tekvizionlabs.com
subject-name CN=asr1004.tekvizionlabs.com
revocation-check crl
rsakeypair GDKey
!
!
crypto pki certificate chain godaddy.root
certificate ca 00
  30820400 308202E8 A0030201 02020100 300D0609 2A864886 F70D0101 05050030
Certificate chain ommited

quit
crypto pki certificate chain godaddy.trustpoint
certificate 27EA0B186F1D91
  30820569 30820451 A0030201 02020727 EA0B186F 1D91300D 06092A86 4886F70D
Certificate chain ommited

quit
certificate ca 0301
  308204DE 308203C6 A0030201 02020203 01300D06 092A8648 86F70D01 01050500
Certificate chain ommited

quit

certificate ca 0301
  308204DE 308203C6 A0030201 02020203 01300D06 092A8648 86F70D01 01050500
Certificate chain ommited

quit

file verify auto

username administrator password 0 solutions
!
redundancy
mode sso
!
!
interface GigabitEthernet0/0/0
description CUCM 2 SIP trunk
ip address 10.64.3.137 255.255.0.0
media-type rj45
negotiation auto
!
interface GigabitEthernet0/0/1
description Public IP to O365
ip address 192.XX.79.XXX 255.255.255.128
media-type rj45
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.0.0.0 255.0.0.0 GigabitEthernet0/0/0
ip route 10.85.0.0 255.255.255.0 10.85.0.1
!
control-plane

!
dial-peer voice 100 voip
description CUCM2-CUBE-O365 ExchUM - Incoming
session protocol sipv2
session transport tcp
incoming called-number 2401
voice-class codec 4 offer-all
dtmf-relay rtp-nte
no vad
!
dial-peer voice 200 voip
description CUCM2-CUBE-O365 ExchUM - Outgoing
destination-pattern 2401
session protocol sipv2
session target dns:e3a46007-31cb-4529-b8cc-1e59b97ebdbd.um.outlook.com
session transport tcp tls
voice-class codec 4 offer-all
voice-class sip call-route url
dtmf-relay rtp-nte
srtp fallback
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 300 voip
description O365- ExchUM-ASR -CUCM - Incoming
session protocol sipv2
session transport tcp tls
incoming called-number .%
voice-class codec 4 offer-all
voice-class sip profiles 1
voice-class sip pass-thru subscribe-notify-events all
dtmf-relay rtp-nte
srtp fallback
no vad
! dial-peer voice 301 voip
description O365- ExchUM-ASR to CUCM Outgoing
destination-pattern 26..
session protocol sipv2
session target ipv4:10.71.2.12
voice-class codec 4 offer-all
dtmf-relay rtp-nte
no vad
!
dial-peer voice 350 voip
description O365- ExchUM-ASR-CUCM-PSTN
destination-pattern 1.........
session protocol sipv2
session target ipv4:10.71.2.12
session transport udp
voice-class codec 4 offer-all
dtmf-relay rtp-nte
no vad
!
dial-peer voice 480 voip
description FAX calls Office365- ExchUM users
session protocol sipv2
session target ipv4:10.85.0.173
destination uri 10
voice-class codec 4 offer-all
voice-class sip call-route url
voice-class sip profiles 100
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
gateway
timer receive-rtp 1200
!
sip-ua
crypto signaling default trustpoint godaddy.trustpoint
!
line con 0
exec-timeout 0 0
login local
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
login local
!
end

**Acronyms**

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>EXT</td>
<td>Extension</td>
</tr>
<tr>
<td>UDP</td>
<td>Uniform Dial Plan</td>
</tr>
<tr>
<td>AA</td>
<td>Auto Attendant</td>
</tr>
<tr>
<td>CFB</td>
<td>Call Forwarding on Busy</td>
</tr>
<tr>
<td>CFNA</td>
<td>Call Forwarding No Answer</td>
</tr>
<tr>
<td>CFU</td>
<td>Call Forwarding Unconditional</td>
</tr>
<tr>
<td>CT</td>
<td>Call Transfer</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>MWI</td>
<td>Message Waiting Indicator</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>RTP</td>
<td>Real Time Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiated Protocol</td>
</tr>
<tr>
<td>TLS</td>
<td>Transport Layer Security</td>
</tr>
<tr>
<td>UM</td>
<td>Unified Messaging</td>
</tr>
<tr>
<td>ExchUM</td>
<td>Exchange Unified Messaging</td>
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</tbody>
</table>