Alcatel-Lucent OmniPCX R11.1 using SIP trunk to Cisco Unified Communications Manager Release 10.5.2 SU3
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Introduction
This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 10.5.2 to interoperate with the Alcatel-Lucent OmniPCX Release 11.1 using SIP Early-Offer.

The following items were tested:

- Basic call between the two systems and verification of voice path, using both SIP and NOE on the Alcatel side, and SIP and SCCP IP phones on the Cisco side (Refer to limitation section for more info)
- CLIP/CLIR/CNIP/CNIR features: calling party Name and number delivery (allowed and restricted) (Refer to limitation section for more info)
- COLP/CONP/COLR/CONR features: connected Name and number delivery (allowed and restricted) (Refer to limitation section for more info)
- Call transfer: attended and early attended (Refer to limitation section for more info)
- Alerting Name Identification (Refer to limitation section for more info)
- Call forwarding: call forward unconditional(CFU), call forward busy (CFB), and call forward no answer (CFNA)
- Hold and resume with music on hold
- Three-way conferencing (Refer to limitation section for more info)
- Voice messaging and MWI activation-deactivation (Refer to limitation section for more info)
- Audio Codec Preference List
- Extend and Connect (Refer to limitation section for more info)
- Call Park (Refer to limitation section for more info)

Listed below are the highlights of the integration issues:

- Basic calls work from Cisco UCM to Alcatel OmniPCX and vice versa. Audio and ring back tone issues are observed with Alcatel SIP phones.
- Caller name and number is not updated correctly for basic calls and in the attended and unattended transfer scenarios. The Alcatel OmniPCX mid call INVITE and 200 OK messages do not contain “PAI” and “user part” in the “Contact” header or “From” header, causing the issue.
- CLIR/CNIR—the Alcatel OmniPCX SIP trunk does not support connected Name and number restriction. Restriction of calling number on Alcatel NOE and SIP phones is achieved by configuring the “Phone Features Classes of Service” with “External Services Secret/Identity” set to “YES” and assigning this class of service to the User under test.
- COLR/CONR—as with calling Name and number presentation restrictions, the Alcatel OmniPCX does not support connected Name and number restriction on SIP trunks.
For call forwarding scenarios where the Alcatel station is the forwarding station, it is required that the Cisco UCM have the “Redirect by Application” checkbox enabled in the SIP Profile used by the SIP Trunk to the Alcatel PBX.

- Retrieval of a parked call on the Alcatel PBX from the Cisco UCM is not supported.
- Video calls between the Alcatel OmniPCX users and the Cisco UCM users is not supported.

Below are the key results:

- Basic call, call transfer, call forwarding, conference call, and hold and resume tested successfully with a few caveats and limitations.
- Centralized voicemail, using Unity Connection server integrated with Cisco UCM via SIP was used for testing. This voicemail solution can provide centralized voicemail services, supporting both Alcatel and Cisco end-users.

Network Topology

Basic Call Setup

![Network Topology Diagram]

Limitations

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

- Although the Codec Preference List is used and the INVITE message displays the right codec, Alcatel OmniPCX responds to the INVITE with its preferred Codec Preference for the call.
• Alcatel OmniPCX SIP does not update the caller ID (connected Name) for a basic or privacy enabled call from the Cisco UCM. Therefore, only the connected party number is displayed on the Cisco UCM Phone.

• Alcatel does not send a privacy header in “200 OK” to Cisco UCM for connected name on restricted calls. The caller ID on the Cisco Phones therefore displays “SIP” or “MASK IDENTITY” instead of “Anonymous/Private”.

• Even though the Alcatel OmniPCX sends Privacy header as Privacy: user; id and from header as anonymous@anonymous.invalid, the Alcatel PBX fails to update its privacy header after the Cisco UCM initiates a transfer. This is due to the difference in the way the two systems handle passing of the restricted information across SIP Trunk.

• The caller name is not updated in transfer/conference scenarios. This is an Alcatel OmniPCX issue because even though the mid call INVITE and UPDATE messages sent by Cisco UCM after the transfer/conference contain PAI and RPI, the Alcatel PBX fails to update this information on its endpoints.

• There is no ring back tone heard on Alcatel SIP phones for a basic outbound call from Alcatel OmniPCX to the Cisco UCM phones.

• In a transfer scenario where the Alcatel SIP phone completes an attended transfer, there is no audio between the connected parties. Also, the call gets disconnected about 20 seconds after the transfer is complete.

• In a transfer scenario, where the attended transfer is made to an Alcatel SIP phone, if the transfer is initiated by a Cisco UCM SIP phone, there is one-way audio if the connected party also is a Cisco UCM phone (legacy or SIP). The Alcatel SIP Phone cannot hear the connected party. In a transfer scenario where the Cisco UCM phone transfers a call from an Alcatel user to another Alcatel user, there is no audio once the transfer is complete. Configuring “Set Ignore inactive/black hole” to True in SIP External Configuration page resolves this issue.

• There is currently no support for Video Capability on the OXE.

• In a Conference call scenario where the conference is initiated by an Alcatel SIP phone, the remaining parties are disconnected from the conference once the Alcatel SIP phone disconnects or leaves the conference.

• In all call forwarding scenarios where the Alcatel station is the forwarding station, it is required that the Cisco UCM have the “Redirect by Application” checkbox enabled in the SIP Profile used by the SIP Trunk to the Alcatel PBX. All external call forward attempts would otherwise fail.

• Call forward from the Alcatel OmniPCX to Cisco UCM fails with an “extend and connect” (CTIRD) configured user. Even though the “302 Moved temporarily” is triggered from Alcatel with the correct contact information (Contact: <sip:5000@10.80.10.3;user=phone>) the call is not terminated on the desired Cisco UCM phone/destination. This issue has been identified as a defect and will be addressed in future releases, tracked against defect ID CSCva13893.

• MWI :
  • The MWI lights up on the Alcatel NOE phone when a user leaves a voicemail for the user. However, on the device, this indication is stored not against a new voicemail, but against a callback to the voicemail pilot number. The user can successfully retrieve and delete the new
voicemail. However, the MWI turns off only after deleting the “callback” entry or calling the voicemail pilot number from that menu.

- MWI does not light up on the Alcatel SIP phones.
- The Alcatel OmniPCX does not support Call Retrieval of a parked call on the local PBX by users on an external node. Therefore; any phone / user on the Cisco UCM cannot retrieve a call that is parked on the Alcatel OmniPCX.
- In a call forward scenario involving multiple hops, if a Cisco user calls an Alcatel phone that is configured in call forward loop, the call fails and never reaches any target destination.

Scenario : PBX A = CCM; PBX B = Vendor PBX; PBX C = CCM

Action: Station B.1 activates Call Forwarding All to Station C.1.
Action: Station C.1 activates Call Forwarding All to Station B.2.
Action: Station B.2 activates Call Forwarding All to Station A.2.
Action: Station A.2 activates Call Forwarding All to Station B.1.
Action: Station A.1 calls station B.1.

**System Components**

**Hardware Requirements**
The following hardware was used

- Cisco UCS-C240-M3S VMWare Host
- Cisco 7965,7975,7960, 8945, 9951, and 9971 IP phones
- Alcatel-Lucent NOE phones 4038, 8068 and SIP phones 4008 and 4018

**Software Requirements**
The following software is required:

- Cisco UCSC-C240-M3S VMware vSphere Image Profile: ESXi-5.5.0-1331820-standard
- Cisco Unified Communications Manager release 10.5.2.13900-12
- Cisco Unified Communications Manager IM & P release 10.5.2.13900-12
- Cisco Unity Connection release 10.5.2.13900-12
- Cisco Jabber 11.6.0 Build 35037
- Alcatel-Lucent OmniPCX Enterprise R11.1
Features
This section lists supported and unsupported features. No deviation from the configuration presented in this document will be supported by Cisco. Please see the Limitations section for more information.

Features Supported
- CLIP—calling line (number) identification presentation
- CLIR—calling line (number) identification restriction
- CNIP—calling Name identification presentation
- CNIR—calling Name identification restriction
- Alerting Name
- Attended call transfer
- Early attended call transfer
- CFU—call forwarding unconditional
- CFB—call forwarding busy
- CFNA—call forwarding no answer
- COLP—connected line (number) identification presentation
- COLR—connected line (number) identification restriction
- CONP—connected Name identification presentation
- CONR—connected Name identification restriction
- Hold and resume
- Conference call
- MWI—Message Waiting Indicator (lamp ON, lamp OFF)
- Audio Codec Preference List
- Call Park/Pickup(see limitation section)
- Extend and Connect

Features Not Supported or Not Tested
- Call completion (callback, automatic callback)
- Shared Line - Hold & Resume with MOH
- Blind transfer
- Video calls
Configuration

The goal of this guide is to provide an overview of the integration between Cisco Unified Communication Manager and Alcatel OmniPCX PBX’s. The deployment will interconnect the UC systems using SIP. No PSTN connectivity has been tested with this integration. The following sections provide the required configurations for a successful integration.

Configuring Sequence and Tasks:

Alcatel OmniPCX:

- Verify Alcatel-Lucent OXE Licenses
- Access the Alcatel-Lucent OXE manager
- Configure IP Domain
- Configure SIP Trunk Group
- Configure Gateway
- Configure SIP External Gateway – Cisco UCM Subscriber/Publisher
- Configure Network Routing Table
- Configure Prefix Plan – Cisco Extension/Voice mail
- Configure G729 Codec
- Configure Privacy
- Configure Call Park
- Configure NOE user
- Configure SIP User

Cisco Unified Communications Manager:

- SIP trunk security profile
- Device setting SIP profile
- Media resource group and media resource group list
- Assign media resource group list (MRGL) in the default device pool
- SIP trunk to Alcatel PBX
- SIP Trunk to Cisco Unity Connection
- Assign User in Cisco Unity Connection
- SIP and SCCP phones device configuration
- Route pattern to the Alcatel PBX
- Call Manager Service Parameter “Duplex Streaming Enabled” set to “True”
- Audio Codec Preference, Region and device pool Configuration
- Extend and Connect Feature and User configuration

Cisco Unity Connection:

- Cisco Unity Connection Telephony Integration
- Cisco Unity Connection User Configuration
Configuring the Alcatel OmniPCX

Alcatel Software Version, capacity and Licensing

From the CLI prompt, use the `spadmin` command and from the menu shown, select option 2, ensure that the system has enough licenses available against the SIP Gateway, Advanced IP users, SIP users, Standard IP users with respect to the configuration requirement.

Access the Alcatel-Lucent OXE Manager

Establish a Telnet connection to the CS board of the OXE. At the CLI prompt type `mgr`. A menu is then presented.
Configure IP Domain

**Navigation:** IP ➔ IP domain

IP Domain Name: lab.tekvizion.com, this is used in this example

Click ctrl+v to complete.

Configure SIP Trunk Group

**Navigation:** Trunk Groups ➔ Create

Node Number: 101, this is used in this example

Trunk Group ID: 1, this is used in this example

Trunk Group Type: T2

Trunk Group Name: SIP

Remote Network: 15, this is used in this example

Q931 Signal variant: ABC-F

T2 Specification: SIP

Associated Ext SIP gateway: 2
<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Node Number (reserved)</td>
<td>101</td>
</tr>
<tr>
<td>Trunk Group ID</td>
<td>1</td>
</tr>
<tr>
<td>Trunk Group Type</td>
<td>T2</td>
</tr>
<tr>
<td>Trunk Group Name</td>
<td>S1P</td>
</tr>
<tr>
<td>UTF-8 Trunk Group Name</td>
<td></td>
</tr>
<tr>
<td>Number Compatible With</td>
<td>-1</td>
</tr>
<tr>
<td>Remote Network</td>
<td>15</td>
</tr>
<tr>
<td>Shared Trunk Group</td>
<td>False</td>
</tr>
<tr>
<td>Special Services</td>
<td>Nothing</td>
</tr>
<tr>
<td>Node number</td>
<td>1</td>
</tr>
<tr>
<td>Transcom Trunk Group</td>
<td>False</td>
</tr>
<tr>
<td>Auto.reserv.by Attendant</td>
<td>False</td>
</tr>
<tr>
<td>Overflow trunk group No.</td>
<td>-1</td>
</tr>
<tr>
<td>Tone on seizure</td>
<td>True</td>
</tr>
<tr>
<td>Private Trunk Group</td>
<td>False</td>
</tr>
<tr>
<td>QSS1 Signal variant</td>
<td>AMC-P</td>
</tr>
<tr>
<td>SS7 Signal variant</td>
<td>No variant</td>
</tr>
<tr>
<td>Number Of Digits To Send</td>
<td>0</td>
</tr>
<tr>
<td>Channel selection type</td>
<td>Quantified</td>
</tr>
<tr>
<td>Auto.DTMF dialing on outgoing call</td>
<td>YES</td>
</tr>
<tr>
<td>T2 Specification</td>
<td>SIP</td>
</tr>
<tr>
<td>Homogenous network for direct SIP</td>
<td>NO</td>
</tr>
<tr>
<td>Public Network COS</td>
<td>31</td>
</tr>
<tr>
<td>DID transcoding</td>
<td>False</td>
</tr>
<tr>
<td>Can support UUS in SETUP</td>
<td>True</td>
</tr>
<tr>
<td>Associated Ext SIP gateway</td>
<td>2</td>
</tr>
</tbody>
</table>

**Implicit Priority**

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Activation mode</td>
<td>0</td>
</tr>
<tr>
<td>Priority Level</td>
<td>0</td>
</tr>
<tr>
<td>Preempter</td>
<td>NO</td>
</tr>
<tr>
<td>Incoming calls Restriction COS</td>
<td>10</td>
</tr>
<tr>
<td>Outgoing calls Restriction COS</td>
<td>10</td>
</tr>
<tr>
<td>Callee number mpt1343</td>
<td>NO</td>
</tr>
<tr>
<td>Overlap dialing</td>
<td>NO</td>
</tr>
<tr>
<td>Call diversion in ISDN</td>
<td>NO</td>
</tr>
</tbody>
</table>
Trusted IP Addresses List

**Navigation:** SIP → Trusted IP Addresses → Review Modify → All Instances
The IP address of the Cisco UCM publisher and subscriber is added to the Trusted IP Addresses List.

![Trusted IP Addresses List](image)

Configure SIP Gateway

**Navigation:** SIP → SIP Gateway

Node Number: 101, this is used in this example

SIP Trunk Group: 1, this is used in this example

IP Address: 10.70.50.6, this is used in this example

DNS local domain name: lab.tekvizion.com, this is used in this example

SIP DNS1 IP Address: 10.64.1.3, this is used in this example
Configure SIP External Gateway for CUCM Subscriber

**Navigation:** SIP → SIP External Gateway → Create

Node Number: 101, this is used in this example

SIP External Gateway ID: 1, this is used in this example

SIP Remote domain: 10.80.10.3 (CUCM subscriber IP), this is used in this example

Trunk group number: 1, this is used in this example

Minimal authentication method + SIP none, this is used in this example

Dynamic Payload type for DTMF: 101

Outbound Calls 100 REL + Supported

Incoming Calls 100 REL + Not Requested

P-Asserted-ID in Calling Number + True

Trusted P-Asserted-ID header + True

Diversion Info to provide via + Diversion

Type of codec negotiation + Default, this is used in this example

Ignore inactive/Black hole + True, this is used in this example
<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Node Number (reserved)</td>
<td>101</td>
</tr>
<tr>
<td>Instance (reserved)</td>
<td>1</td>
</tr>
<tr>
<td>SIP External Gateway ID</td>
<td>2</td>
</tr>
<tr>
<td>Gateway Name</td>
<td>CDEK</td>
</tr>
<tr>
<td>SIP Remote domain</td>
<td>10.80.10.3</td>
</tr>
<tr>
<td>PCS IP Address</td>
<td></td>
</tr>
<tr>
<td>SIP Port Number</td>
<td>5060</td>
</tr>
<tr>
<td>Transport type</td>
<td>UDP</td>
</tr>
<tr>
<td>Belonging Domain</td>
<td>lab.television.com</td>
</tr>
<tr>
<td>Registration ID</td>
<td></td>
</tr>
<tr>
<td>Registration ID P-Asserted</td>
<td>False</td>
</tr>
<tr>
<td>Registration timer</td>
<td>0</td>
</tr>
<tr>
<td>SIT Outbound Proxy</td>
<td></td>
</tr>
<tr>
<td>Supervision timer</td>
<td>0</td>
</tr>
<tr>
<td>Trunk group number</td>
<td>1</td>
</tr>
<tr>
<td>Pool Number</td>
<td>-1</td>
</tr>
<tr>
<td>Outgoing realm</td>
<td></td>
</tr>
<tr>
<td>Outgoing username</td>
<td></td>
</tr>
<tr>
<td>Outgoing Password</td>
<td></td>
</tr>
<tr>
<td>Confirm</td>
<td></td>
</tr>
<tr>
<td>Incoming username</td>
<td></td>
</tr>
<tr>
<td>Incoming Password</td>
<td></td>
</tr>
<tr>
<td>Confirm</td>
<td></td>
</tr>
<tr>
<td>RFC 3325 supported by the distant</td>
<td>True</td>
</tr>
<tr>
<td>DNS type</td>
<td>DNS A</td>
</tr>
<tr>
<td>SIP DNS1 IP Address</td>
<td></td>
</tr>
<tr>
<td>SIP DNS2 IP Address</td>
<td></td>
</tr>
<tr>
<td>SDP in 18x</td>
<td>True</td>
</tr>
<tr>
<td>Minimal authentication method</td>
<td>SIP None</td>
</tr>
<tr>
<td>INFO method for remote extension</td>
<td>False</td>
</tr>
<tr>
<td>To EMS</td>
<td>False</td>
</tr>
<tr>
<td>SRTP</td>
<td>SRTP only</td>
</tr>
<tr>
<td>Ignore inactive/black hole</td>
<td>True</td>
</tr>
<tr>
<td>Contact with IP address</td>
<td>False</td>
</tr>
<tr>
<td>Dynamic Payload type for DTMF</td>
<td>101</td>
</tr>
</tbody>
</table>
Configure SIP External Gateway for CUCM Publisher

SIP Remote domain: 10.80.10.2 (CUCM_PUBLISHER)

**Navigation:** SIP → SIP External Gateway → Create

Node Number: 101, this is used in this example

SIP External Gateway ID: 1, this is used in this example

SIP Remote domain: 10.80.10.2, this is used in this example

Trunk group number: 1, this is used in this example

Minimal authentication method + SIP none, this is used in this example

Dynamic Payload type for DTMF: 101

Outbound Calls 100 REL + Supported

Incoming Calls 100 REL + Not Requested

P-Asserted-ID in Calling Number + True

Trusted P-Asserted-ID header + True

Diversion Info to provide via + Diversion

Type of codec negotiation + Default, this is used in this example

Ignore inactive/Black hole + True, this is used in this example
Node Number (reserved) : 101
Instance (reserved) : 1
SIP External Gateway ID : 4

Gateway Name : CUCM_FUS
SIP Remote domain : 10.80.10.2
FCS IP Address : 
SIP Port Number : 5060
Transport type : UDP
Belonging Domain : lab.tekvizion.com
Registration ID : 
Registration ID P-Asserted : False
Registration timer : 0
SIP Outbound Proxy : 
Supervision timer : 0
Trunk group number : 1
Pool Number : -1
Outgoing realm : 
Outgoing username : 
Outgoing Password : 
Incoming username : 
Incoming Password : 
RFC 3325 supported by the distant : True
DNS type : DNS A
SIP DNS1 IP Address : 
SIP DNS2 IP Address : 
SDP in 16x : False
Minimal authentication method : SIP None
INFO method for remote extension : False
To EMS : False
sntp : sntp only
Ignore inactive/black hole : True
Contact with IP address : False
Dynamic Payload type for DTMF : 101
<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Password</td>
<td>-----------------------</td>
</tr>
<tr>
<td><strong>RFC 3325 supported by the distant</strong></td>
<td>True</td>
</tr>
<tr>
<td><strong>DNS type</strong></td>
<td>DNS A</td>
</tr>
<tr>
<td><strong>SIP DNS1 IP Address</strong></td>
<td></td>
</tr>
<tr>
<td><strong>SIP DNS2 IP Address</strong></td>
<td></td>
</tr>
<tr>
<td><strong>SDP in 16x</strong></td>
<td>False</td>
</tr>
<tr>
<td><strong>Minimal authentication method</strong></td>
<td>SIP None</td>
</tr>
<tr>
<td><strong>INFO method for remote extension</strong></td>
<td>False</td>
</tr>
<tr>
<td><strong>To EMS</strong></td>
<td>False</td>
</tr>
<tr>
<td><strong>SRTP</strong></td>
<td>RTP only</td>
</tr>
<tr>
<td><strong>Ignore inactive/black hole</strong></td>
<td>True</td>
</tr>
<tr>
<td><strong>Contact with IP address</strong></td>
<td>False</td>
</tr>
<tr>
<td><strong>Dynamic Payload type for DTMF</strong></td>
<td>101</td>
</tr>
<tr>
<td><strong>Outbound Calls 100 REL</strong></td>
<td>Supported</td>
</tr>
<tr>
<td><strong>Incoming Calls 100 REL</strong></td>
<td>Not Requested</td>
</tr>
<tr>
<td><strong>Gateway type</strong></td>
<td>Standard Type</td>
</tr>
<tr>
<td><strong>Re-Trans No. for REGISTER/OPTIONS</strong></td>
<td>3</td>
</tr>
<tr>
<td><strong>P-Asserted-ID in Calling Number</strong></td>
<td>True</td>
</tr>
<tr>
<td><strong>Trusted P-Asserted-ID Header</strong></td>
<td>True</td>
</tr>
<tr>
<td><strong>Diversion Info to provide via</strong></td>
<td>Diversion</td>
</tr>
<tr>
<td><strong>Proxy Identification on IP address</strong></td>
<td>False</td>
</tr>
<tr>
<td><strong>Outbound calls only</strong></td>
<td>False</td>
</tr>
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<td><strong>SDP relay on Ext. Call Pwd</strong></td>
<td>Default</td>
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<tr>
<td><strong>SDP Transparency Override</strong></td>
<td>False</td>
</tr>
<tr>
<td><strong>RFC 5009 supported / Outbound call</strong></td>
<td>Not Supported</td>
</tr>
<tr>
<td><strong>Nonce caching activation</strong></td>
<td>NO</td>
</tr>
<tr>
<td><strong>FAX Procedure Type</strong></td>
<td>T38 only</td>
</tr>
<tr>
<td><strong>DNS SRV/Call retry on busy server</strong></td>
<td>0</td>
</tr>
<tr>
<td><strong>Unattended Transfer for RSI</strong></td>
<td>NO</td>
</tr>
<tr>
<td><strong>Redirection functionality</strong></td>
<td>NO</td>
</tr>
<tr>
<td><strong>Attended Transfer</strong></td>
<td>NO</td>
</tr>
<tr>
<td><strong>Send BYE on REFER</strong></td>
<td>YES</td>
</tr>
<tr>
<td><strong>Support UTF8 characters act</strong></td>
<td>NO</td>
</tr>
<tr>
<td><strong>CSTA User-to-User supported</strong></td>
<td>NO</td>
</tr>
<tr>
<td><strong>Trusted From header</strong></td>
<td>True</td>
</tr>
<tr>
<td><strong>Support Re-invite without SDP</strong></td>
<td>False</td>
</tr>
<tr>
<td><strong>Type of codec negotiation</strong></td>
<td>Default</td>
</tr>
</tbody>
</table>
Configure Network Routing Table

**Navigation:** Translator → Network Routing Table

Node Number: 101, this is used in this example

Protocol Type: ABC-F

ARS Route List: 5, this is used in this example

Associated Ext SIP Gateway: 2, this is used in this example

Configure Prefix Plan – Pointing to Cisco Phones.

**Navigation:** Translator → Prefix plan → Create

Node Number: 101, this is used in this example

Prefix Meaning + Routing No.

Network Number: 10, this is used in this example

Node Number/ABC-F Trunk Group: 1, this is used in this example

Number of Digits: 4

Note: Cisco extension are started with 5 as prefix digit.
Configure Prefix Plan – Pointing to Cisco Voice Mail Pilot Number.

**Navigation:** Translator → Prefix plan → Create

Node Number: 101, this is used in this example

Prefix Meaning + Routing No.

Network Number: 10, this is used in this example

Node Number/ABC-F Trunk Group: 1, this is used in this example

Number of Digits: 4

Note: Cisco Voice mail pilot number started with 7 as prefix digit.
Configure Privacy

Privacy for particular user

**Navigation:** mgr → user → Descend hierarchy → Dynamic State User → Review/Modify

Set **Consistent with Identity:** False i.e., Secret Identity Enabled

Enable Privacy in Entity Privacy for particular user

**Navigation:** mgr → Entities → Review/Modify → Entity Number:

Set **Caller ID Secret:** Yes (i.e., Secret Identity enabled in whole entity that includes sip trunk & users)
Enable Privacy in COS

**Navigation:** mgr → Classes of service → Phone Features COS → Review/modify

Node Number: 101, this is used in this example.

Instance (reserved): 1, Phone Features COS: 1

Rights Mask Id. name only for ext. Calls: set to 0. This is used in this example.
Configure Call Park

a. Authorization: To enable the rights on the user to be able to use the call park/retrieve feature

Navigation: Classes of Service → Phone Features COS → Review/Modify → External Services → Park Call/Retrieve: 1
b. Prefix Definition: Configure a prefix or verify if the system has a default prefix configuration that is used for activating the parking function and retrieving the parked call.

Navigation: Translator → Prefix Plan → Review/Modify → *75

c. Operation
i. Parking
   With a call in progress:
   1. Dial the parking prefix and wait for the voice guide.
   2. Dial the number of the set on which you wish to park the call.
   3. The PBX confirms your action by voice guide.
   4. Hang up.

ii. Retrieving the parked call
   To retrieve a parked call from any set:
   1. Hang up.
   2. Dial the parking prefix and wait for the voice guide.
3. Dial the number of the set on which the call was parked.
Note: To retrieve the parked call from the same set the call was parked on, dial the parking prefix.

**Configure G729 Codec**

**Navigation:** IP → IP Domain → Review/Modify → 0

Node Number: 101, this is used in this example

IP Domain Name: lab.tekvizion.com

Intra-domain Coding Algorithm + Without Compression

Extra-domain Coding Algorithm + With Compression

---

**Configure NOE user**

**Navigation:** Users → Create

Node Number (reserved): 101

Directory Number: 4004

Directory name: NOE_USER1

Set Type + IPTouch 4038/8038, this is used in this example
Configure SIP user

**Navigation:** Users → Create

- Node Number (reserved): 101
- Directory Number: 4004
- Directory name: NOE_USER1
- Location Node: 1
- Shelf Address: 255
- Board Address: 255
- Equipment Address: 255
- Set Type + IPTouch 4038/8038
- Entity Number: 1
- Set Function + Default
- Profile Name: -----------
- Key Profiles + Company
- Domain Identifier: 0
- Language ID: 1
- Secret Code: ****

- Associated Set No.: 4004
- Cost Center ID: 255
- Cost Center Name: -----------
- Charging COS + Justified
- Public Network COS: 2
- External Forwarding COS: 255
- Phone Features COS: 0
- Connection COS: 0
- Hunt Group Dir No.: -------
- ACD Group Directory No.: -------
- Pickup Group Name: -----------
- Reserved Time Slot + False
- Voice Mail Dir.No.: -------
- Voice Mail Type + No Voice Mail
- Paging Trunk Group: 255
- Paging Beeper: ----
- Tele-Marketing Agent + False

Directory Number: 4004, this is used in this examples
Directory name: ALU_SIP1, this is used in this examples
Set Type + SIP extension, this is used in this examples
URL Username: 4003
URL Domain: tekvizionoxe
SIP Authentication: 4003
SIP Password: ******
Confirm: ******
Explicit Priority

Activation mode : 0
Priority Level : 0

Pre-emptable Primary Inc. Line + NO
Pre-emptable Secondary Inc. Line + NO
Priority Presentation + NO
Irr Service type + Not Valid
CUG List Number : -1
Preference CUG : -1
CUG Outgoing Access + False
CUG Incoming Access + False
Automatic reconfiguration + CTQ Forbidden - Connection TO
Associated RSI : --------

URL User Name : 4003
URL Domain : tekvizionoxe
SIP Authentication : 4003

SIP Passwd : *****
Confirm : *****

Called Associated DECT set : ---------
Dial by name and text msg. + NO
Text msg number : 8
Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager Software Version

VMware Installation: 2 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 4096Mbytes RAM, Partitions aligned

Successful Logon: Unavailable

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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 our Export Compliance Product Report web site.

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

Cisco Technical Support please visit our Technical Support web site.
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

All other values are default.

Note: Alcatel sends refer-to with replaces header for transfer scenarios, Cisco sends 403 forbidden but there were no issues in the call flow.
Cisco Unified Communications Manager SIP Trunk Security Profile for Unity Connection

Set Name* = Non Secure SIP Trunk to VM Profile. This is used for this example.

Set Description = this text is used to identify this SIP Trunk Security Profile.

Check Accept presence subscription

Check Accept out of dialog refer**

Check Accept unsolicited notification

All other values are default.

---

**Cisco Unified Communications Manager SIP Profile**

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

Set Name* = ALU - Standard SIP Profile. This is used for this example.

Set Description = this text is used to identify this SIP Profile.

Check Disable Early Media on 180

Check Redirect by Application
All other values are default.

![Cisco Unified CM Administration Interface](image)

- **Status**: Ready
  - All SIP devices using this profile must be restarted before any changes will take affect.

- **SIP Profile Information**
  - **Name**: ALU - Standard SIP Profile
  - **Description**: Default SIP Profile
  - **Default MTP Telephony Event Payload Type**: 101
  - **Early Offer for G.Clear Calls**: Disabled
  - **User-Agent and Server header information**: Send Unified CM Version Information as User-Agent
  - **Version in User Agent and Server Header**: Major And Minor
  - **Dial String Interpretation**: Phone number consists of characters 0-9, *, #, and
  - **Confidential Access Level Headers**
    - **Redirect by Application**: Enabled
    - **Disable Early Media on 180**: Enabled
  - **Outgoing T.38 INVITE include audio mine**: 
  - **Use Fully Qualified Domain Name in SIP Requests**: 
  - **Assured Services SIP conformance**: 

---

Page 33 of 107
Cisco Unified Communications Manager SIP Profile (Continued)

These values are default.

Set SIP Rel1XX Options* = Send PRACK if 1xx Contains SDP

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

<table>
<thead>
<tr>
<th>Configuration</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Pickup Group URI</td>
<td>x-cisco-serviceurl-gpickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-cisco-serviceurl-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 Level for 7940 and 7960</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-fwdall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-abbrdia</td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td>Checking</td>
</tr>
<tr>
<td>RFC 2543 Hold</td>
<td>Checking</td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td>Checking</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Profile (Continued)

<table>
<thead>
<tr>
<th>SIP Profile Configuration</th>
<th>Related Links</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable VAD</td>
<td>Back To Find/List</td>
</tr>
<tr>
<td>Stutter Message Waiting</td>
<td>Go</td>
</tr>
<tr>
<td>MLPP User Authorization</td>
<td></td>
</tr>
</tbody>
</table>

### Normalization Script

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Incoming Requests FROM URI Settings

<table>
<thead>
<tr>
<th>Caller ID DN</th>
<th>Caller Name</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Trunk Specific Configuration

- Re-route Incoming Request to new Trunk based on
- RSVP Over SIP
- Resource Priority Namespace List
- Fall back to local RSVP
- SIP Re1XX Options
- Video Call Traffic Class

<table>
<thead>
<tr>
<th>Authentication</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Never</td>
<td></td>
</tr>
<tr>
<td>Local RSVP</td>
<td></td>
</tr>
<tr>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>Send PULL if 1xx Contains SDP</td>
<td>Mixed</td>
</tr>
</tbody>
</table>

---

Page 36 of 107
Cisco Unified Communications Manager SIP Profile (Continued)

Early Offer support for voice and video calls: Mandatory (insert MTP if needed)

Check Enable OPTIONS Ping to monitor Destination status for Trunks with Service Type "None (Default)"

Check Send send-receive SDP in mid-call INVITE

All other values are default.
Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration

Navigation: Device → Trunk

Set Device Name*= ALU. This is used for this example.
Set Description = this text is used to identify this Trunk Group.
Set Device Pool* = G711 Preferred. This is used for this example
Set Call Classification* = Use System Default. This is used for this example
Set Media Resource Group List = MRGL_NoMTP. This is used for this example

All other values are default.
Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration (Continued)

All other values are default.
### Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration (Continued)

![Cisco Unified CM Administration Interface](image)

**Call Routing Information**

- **Remote-Party-Id**: Checked
- **Asserted-Identity**: Checked
- **Asserted-Type**: Default
- **SIP Privacy**: Default

**Inbound Calls**

- **Significant Digits**: All
- **Connected Line ID Presentation**: Default
- **Connected Name Presentation**: Default
- **Calling Search Space**: <None>
- **AAR Calling Search Space**: <None>
- **Prefix DN**: 

- **Redirecting Diversion Header Delivery - Inbound**: Unchecked
Check Redirecting Diversion Header Delivery - Outbound

All other values are default.
Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration (Continued)

Set Destination Address = 10.70.50.6. This is used in this example.

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

Set SIP Profile* = ALU – Standard SIP Profile

Set DTMF Signaling Method* = No Preference

All other values are default.
Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration (Continued)
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration

**Navigation:** Device → Trunk

Set Device Name*= CUC. This is used for this example.
Set Description = this text is used to identify this Trunk Group.
Set Device Pool* = Default
All other values are default.
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

Check Run On All Active Unified CM Nodes

All other values are default.
Check Redirecting Diversion Header Delivery - Inbound

All other values are default.

Check Redirecting Diversion Header Delivery – Outbound
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

Set Destination Address = 10.80.10.5. This is used in this example.

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

Set SIP Profile* = Standard SIP Profile

DTMF Signaling Method * = No Preference

All other values are default.
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

![Cisco Unified CM Administration UI](Image)

**Trunk Configuration**

<table>
<thead>
<tr>
<th>Normalization Script</th>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Normalization Script</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

**Recording Information**

- **None**
- This trunk connects to a recording-enabled gateway
- This trunk connects to other clusters with recording-enabled gateways

**Geolocation Configuration**

- **Geolocation**  | < None >         |
- **Geolocation Filter**  | < None >        |

- **Send Geolocation Information**

---

* indicates required item.

**. Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Cisco Unified Communications Manager Service Parameter

**Navigation Path:** System → Service parameter → select server (Cluster20pub) → Select Service (Cisco Call Manager (Active))

Set Duplex Streaming Enabled* = True.

---

*Note: Cisco Unified Communications Manager Service Parameter “Duplex Streaming Enabled” should be set to “True” in order for MoH and ring back to work properly during call transfers/conferences initiated by Cisco stations to Alcatel IP endpoints.
Cisco Unified Communications Manager Device Pool Configuration

Navigation Path: System → Device Pool

G711 Preferred and G729 Preferred created in this example.

All other values are default.

Cisco Unified Communications Manager Device Pool Configuration (Continued)

Set Device Pool Name* = G711 Preferred. This is used in this example.

Set Cisco Unified Communications Manager Group* = Default

Set Date/Time Group* = CMLocal

Set Region* = G711 Preferred. This is used in this example

Set Media Resource Group List = MRGL_noMTP. 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 Unified Communications Manager Device Pool Configuration (Continued)

Set Device Pool Name*= G729 Preferred. This is used in this example.

Set Cisco Unified Communications Manager Group*= Default

Set Date/Time Group* = CMLocal

Set Region* =G729 Preferred. This is used in this example

Set Media Resource Group List =MRGL. 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 Unified Communications Manager Region Configuration

Navigation Path: System → Region Information → Region

G711 Preferred and G729 Preferred created in this example.

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

Set Name*= G711 Preferred. This is used in this example
Set Region= G711 Preferred. This is used in this example
Set Audio Codec Preference List= G711 Preferred
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
Set Audio Codec Preference List= G711 G729. This is used in this example
Set Maximum Audio Bit Rate= 64 Kbps (G722, G7.11). This is used in this example

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

Set Name*= G729 Preferred. This is used in this example.

Set Region= G729 Preferred. This is used in this example

Set Audio Codec Preference List= G729 G711. 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.

Set Audio Codec Preference List= G729 G711. This is used in this example

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

All other values are default.
Cisco Unified Communications Manager Media Resource Group

**Navigation Path:** Media Resources → Media Resource Group
Media Resource Group MRG

Set Name* = MRG, This is used for this example.

Set Description = this text is used to identify this Media Resource Group.

Set all resources in the Selected Media Resources* Box.

All other values are default.

Resource Group for MRG_NoMTP

Set Name* = MRG_NoMTP. This is used for this example.

Set Description = this text is used to identify this Media Resource Group.

Set Available Media Resources = MTP_2, MTP_3 and MTP_4

Set other resources in the Selected Media Resources*

All other values are default.
Cisco Unified Communications Manager Media Resource Group List

**Navigation Path:** Media Resources ➔ Media Resource Group List

---

### Media Resource Group Configuration

- **Status:**
  - Status: Ready

- **Media Resource Group Status:**
  - Media Resource Group: MRG_NoMTP (used by 12 devices)

- **Media Resource Group Information**
  - **Name:** MRG_NoMTP
  - **Description:**

- **Devices for this Group**
  - **Available Media Resources:**
    - MTP_2
    - MTP_3
    - MTP_4

  - **Selected Media Resources:**
    - ANI_2 (ANN)
    - ANI_3 (ANN)
    - ANI_4 (ANN)
    - CFRD_2 (CFR)
    - CFRD_3 (CFR)

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

---

### Find and List Media Resource Group Lists

- **Status:**
  - 2 records found

### Media Resource Group List (1 - 2 of 2)

<table>
<thead>
<tr>
<th>Name</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>MRG_NoMTP</td>
<td></td>
</tr>
<tr>
<td>MRG_NoMTP</td>
<td></td>
</tr>
</tbody>
</table>

---

Page 61 of 107
Set Name* = MRGL. This is used for this example.

Set Description = this text is used to identify this Media Resource Group List.

Set Available Media Resources = MRGL

Set Selected Media Resource Groups = MRG

---

Set Name* = MRGL_noMTP. This is used for this example

Set Description = this text is used to identify this Media Resource Group List

Set Available Media Resources MRG

Set Selected Media Resource Groups = MRGL_NoMTP
Note: MRG_NoMTP Media resource group was used to test the scenarios that did not require MTP.

**Cisco Unified Communications Manager Route Pattern to Alcatel**

Set Route Pattern* =4XXX. This is used to route to the Alcatel PBX in this example

Set Description = this text is used to identify this Route Pattern

Set Gateway/Route List* = To ALU. This is used for this example

Uncheck Provide Outside Dial Tone

All other values are default.
Route Pattern Configuration for 4xxx (Continued)

Set Calling Party Transform Mask = XXXX

Set Calling Line ID Presentation= Allowed

Set Calling Name Presentation= Allowed

Set Connected Line ID Presentation* = Default

Set Calling Name Presentation* = Default

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

Navigation Path: Device ➔ Phone

Set MAC Address* = 081FF3625Dxx. This is used in this example

Set Description = this text is used to identify this Phone

Set Device Pool* = G711 Preferred. This is used in this example

Set Phone Button Template* = Standard 8961 SIP. This is used in this example

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

Set Media Resource Group List = None. This is used in this example. All other values are default.
Set Device Security Profile* = Cisco 8961 - Standard SIP Non-Secure Profile. This is used in this example.

Set SIP Profile* = Standard SIP Profile. This is used in this example.

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 values are default.

<table>
<thead>
<tr>
<th>Secure Information URL</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Messages URL</td>
<td></td>
</tr>
<tr>
<td>Secure Services URL</td>
<td></td>
</tr>
</tbody>
</table>

**Extension Information**

- Enable Extension Mobility
- Log Out Profile: Use Current Device Settings
- Log in Time: < None >
- Log out Time: < None >

**MLPP and Confidential Access Level Information**

- MLPP Domain: < None >
- MLPP Indication: Default
- MLPP Preemption: Default
- Confidential Access Mode: < None >
- Confidential Access Level: < 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>Param</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td></td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td></td>
</tr>
<tr>
<td>PC Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>Back USB Port</td>
<td>Enabled</td>
</tr>
</tbody>
</table>
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.

Set Cisco Camera* = Enabled. This is used in this example.

Set Video Capabilities* = Enabled. This is used in this example.

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

All values are default.
<table>
<thead>
<tr>
<th>HTTPS Server</th>
<th>http and https Enabled</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Power</td>
<td>Sunday</td>
</tr>
<tr>
<td>Save Plus</td>
<td>Monday</td>
</tr>
<tr>
<td>Phone On Time</td>
<td>Tuesday</td>
</tr>
<tr>
<td>Phone Off Time</td>
<td></td>
</tr>
<tr>
<td>Phone Off Idle Time</td>
<td>06:00</td>
</tr>
<tr>
<td>Timeout</td>
<td>24:00</td>
</tr>
<tr>
<td>Enable Audible Alert</td>
<td></td>
</tr>
<tr>
<td>EnergyWise Domain</td>
<td></td>
</tr>
<tr>
<td>EnergyWise Endpoint</td>
<td></td>
</tr>
<tr>
<td>Security Secret</td>
<td></td>
</tr>
<tr>
<td>Allow EnergyWise Overides</td>
<td></td>
</tr>
<tr>
<td>Span to PC Port</td>
<td>Disabled</td>
</tr>
<tr>
<td>Logging Display</td>
<td>Disabled</td>
</tr>
<tr>
<td>Load Server</td>
<td></td>
</tr>
<tr>
<td>IPv6 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>
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

![Phone Configuration](image)

<table>
<thead>
<tr>
<th>Discovery Protocol - Media Endpoint</th>
<th>Link Layer Discovery Protocol (LLDP):</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enabled</td>
<td>Enabled</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Discovery (LLDP-MED): Switch Port</th>
<th>LLDP Power Priority</th>
<th>802.1x Authentication</th>
<th>FIPS Mode</th>
<th>Detect Unified CM Connection Failure</th>
<th>Switch Port Remote Configuration</th>
<th>PC Port Remote Configuration</th>
<th>Automatic Port Synchronization</th>
<th>Power Negotiation</th>
<th>ESM Access</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enabled</td>
<td>Unknown</td>
<td>User Controlled</td>
<td>Disabled</td>
<td>Normal</td>
<td>Enabled</td>
<td>Disabled</td>
<td>Disabled</td>
<td>Enabled</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

Page 75 of 107
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

<table>
<thead>
<tr>
<th>Phone Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
<tbody>
<tr>
<td>Show Call History for Selected Line Only.</td>
<td>Disabled</td>
</tr>
<tr>
<td>Actionable Incoming Call Alert.</td>
<td>Disabled</td>
</tr>
<tr>
<td>DIF bit</td>
<td>0</td>
</tr>
<tr>
<td>Default Line Filter</td>
<td></td>
</tr>
<tr>
<td>Separate Audio and Video Mute</td>
<td>Disabled</td>
</tr>
<tr>
<td>Softkey Control</td>
<td>Feature Control Policy</td>
</tr>
<tr>
<td>Start Video Port</td>
<td></td>
</tr>
<tr>
<td>Stop Video Port</td>
<td></td>
</tr>
<tr>
<td>Lowest Alerting Line State Priority</td>
<td>Disabled</td>
</tr>
<tr>
<td>TLS Resumption Timer</td>
<td>3600</td>
</tr>
<tr>
<td>Audio EQ</td>
<td>Default : Default</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SCCP Phone Device Level Configuration

**Navigation Path:** Device ➔ Phone

Set MAC Address* = 001DA21A0Bxx. This is used in this example.

Set Description = this text is used to identify this Phone

Set Device Pool*= G711 Preferred. This is used in this example.

Set Phone Button Template*= Standard 7971 SCCP. 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.
Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

Set Device Security Profile* = Cisco 7971 – 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)

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

Set Video Capabilities* = Enabled

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

These values are default.
These values are default.
Cisco Unified Communications Manager Audio Codec Preference List Configuration

**Navigation Path:** System → Service parameter → select server (Cluster20pub) → Select Service (Cisco Call Manager (Active))

Set Accept Audio Codec Preference in Received Offer *= Off. This needs to be set when you are wanting to use the Codec Preference List created.
Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

G711 Preferred and G729 Preferred Audio Codec Preference List created in this example.

All other values are default.

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>Faster Default Lossy</td>
<td>Lossy Codec List</td>
<td></td>
</tr>
<tr>
<td>Faster Default NonLoss</td>
<td>Low Loss Codec List</td>
<td></td>
</tr>
<tr>
<td>G711 0728</td>
<td>G711 preferred</td>
<td></td>
</tr>
<tr>
<td>G729 0711</td>
<td>G729 preferred</td>
<td></td>
</tr>
<tr>
<td>MOSIP</td>
<td>Low Loss Codec List</td>
<td></td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

Set Name*= G711 G729. This is used for this example

Set Description*= this text is used to identify this Audio Codec Preference List

Set Codec in List*= G.711 64k. First choice in this example

Set Codec in List*= G.729 8k. Second choice in this example

All other values are default.
Cisco UCM Extent and Connect

Extend and Connect is a feature that allows administrators to rapidly deploy UC Computer Telephony Integration (CTI) applications which interoperate with any endpoint. With Extend and Connect, users can leverage the benefits of UC applications from any location using any device. This feature also allows Interoperability between newer UC solutions and legacy systems, so customers can migrate to newer UC Solutions over time as existing hardware is deprecated.

Cisco UCM end user configuration
Add user to Cisco UCM

**Navigation Path:** User Management → End user
Set User ID* = user1. This is used for this example.
Set Last Name = user1. This is used for this example.
Check Home Cluster.

Cisco UCM end user Configuration (Continued)
Set Controlled Devices = CTI1. This is used for this example.

Check Allow Control of Device from CTI
Select the Primary Extension for this user. 5007 is used for this example.
Check Enable Mobility

Add the following permissions for Standard Users:
– Standard CCM End-Users
– Standard CTI Enabled
– Standard CCMUSER Administration

Add Phone: CTI Remote Device

The CTI Remote Device type represents the user’s remote device(s).
Select the desired Owner User ID. User1 is used in this example.
Set the Device Name populated automatically. Modify if desired - CTI1 used this example.
Set Device Pool: Default. This is used in this example.
Cisco UCM CTI remote device Configuration (Continued)
Set RD* = 4004. This is used for this example. 4004 is the Alcatel extension.

Remote Destination Configuration
Set Destination Number* = 4004. This is used for this example.
Check Enable Extend and Connect.
Cisco UCM UC service Configuration

**Navigation Path:** User Management → User setting → UC Service

---

Cisco UCM service Profile Configuration

**Navigation Path:** User Management → User setting → Service Profile

---

Cisco UCM service profile Configuration (Continued)
### Conferencing Profile

<table>
<thead>
<tr>
<th>Type</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Secondary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Tertiary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Server Certificate Verification</td>
<td>Any</td>
</tr>
<tr>
<td>Credentials source for web conferencing service</td>
<td>Not set</td>
</tr>
</tbody>
</table>

### Directory Profile

<table>
<thead>
<tr>
<th>Type</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Secondary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Tertiary</td>
<td>&lt;None&gt;</td>
</tr>
</tbody>
</table>

- Use LDS for Context Resolution
- Use Lookup On User Credential

**User Details**

- **Username**: administrator
- **Password**: ********

**Search Bases**

- **Search Base 1**: 
- **Search Base 2**: 
- **Search Base 3**: 

- **Recursive Search on All Search Bases**
- **Search Timeout (seconds)**: 5

### IM and Presence Profile

- **Primary**: IMP_SRV
- **Secondary**: <None>
- **Tertiary**: <None>

### CTI Profile

- **Primary**: CTI_SRV
- **Secondary**: CTI_SUB1
- **Tertiary**: <None>

### Video Conference Scheduling Portal Profile

- **Primary**: <None>
- **Secondary**: <None>
- **Tertiary**: <None>
Set Name*: remotedesk. This is used in this example.
Set Primary CCMCIP Host*: 10.80.10.2. Cisco Publisher IP. This is used in this example.
Set Backup CCMCIP Host*: 10.80.10.3. Cisco Publisher IP. This is used in this example.
Add Users to Profile: user2. This is used in this example.
Cisco UCM – SIP trunk to Cisco IM&Presence Trunk Configuration

**Navigation Path:** Device → Trunk

Set Device Name*= IMPTrunk. This is used for this example.
Set Description = this text is used to identify this Trunk Group.
Set Device Pool*= Default. This is used for this example.
Set Media Resource Group List = MRGL. This is used for this example.

All other values are default.

![Trunk Configuration](image)

<table>
<thead>
<tr>
<th>Device Information</th>
<th>SIP Trunk</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Device Name</strong></td>
<td>IMPTrunk</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Device Pool</strong></td>
<td>G711 Preferred</td>
</tr>
<tr>
<td><strong>Common Device Configuration</strong></td>
<td>Use System Default</td>
</tr>
<tr>
<td><strong>Media Resource Group List</strong></td>
<td>MRGL</td>
</tr>
<tr>
<td><strong>Location</strong></td>
<td>Hub Name</td>
</tr>
<tr>
<td><strong>MR Group</strong></td>
<td>X_Name &gt;</td>
</tr>
<tr>
<td><strong>Tunneled Protocol</strong></td>
<td>None</td>
</tr>
<tr>
<td><strong>QSIG Variant</strong></td>
<td>No Changes</td>
</tr>
</tbody>
</table>

![Trunk Configuration](image)
Cisco Unified Communications Manager SIP Trunk to CUP Configuration (Continued)

All other values are default.

<table>
<thead>
<tr>
<th>Configuration Item</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASN.1 ROSE DID Encoding</td>
<td>No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Media Termination Point Required</td>
<td></td>
</tr>
<tr>
<td>Retry Voice Call as Audio</td>
<td></td>
</tr>
<tr>
<td>Path Replacement Support</td>
<td></td>
</tr>
<tr>
<td>Transmit UTF-8 for Calling Party Name</td>
<td></td>
</tr>
<tr>
<td>Transmit UTF-8 Names in QSIG ABDU</td>
<td></td>
</tr>
<tr>
<td>Unattended Port</td>
<td></td>
</tr>
<tr>
<td>SRTP Allowed</td>
<td>When using both SRTP and TLS</td>
</tr>
<tr>
<td>Consider Traffic on This Trunk Secure</td>
<td></td>
</tr>
<tr>
<td>Route Class Signaling Enabled</td>
<td>Default</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
</tr>
<tr>
<td>PSTN Access</td>
<td></td>
</tr>
<tr>
<td>Run On All Active Unified CM Nodes</td>
<td></td>
</tr>
</tbody>
</table>

### Intercompany Media Engine (IME)

- E.164 Transformation Profile: <None>

### MLPP and Confidential Access Level Information

- MLPP Domain: <None>
- Confidential Access Mode: <None>
- Confidential Access Level: <None>

### Call Routing Information

- Remote-Party-id
- Asserted-Identity
- Asserted-Type: Default
- SIP Privacy: Default

### Inbound Calls

- Significant Digits: All
- Connected Line ID Presentation: Default
- Connected Name Presentation: Default
- Calling Search Space: <None>
- AAR Calling Search Space: <None>
- Prefix DN: 
- Redirecting Diversion Header Delivery - Inbound
Cisco UCM SIP Trunk to CUP Configuration (Continued)

## Incoming Calling Party Settings

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

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
</tbody>
</table>

## Incoming Called Party Settings

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

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
</tbody>
</table>

## Connected Party Settings

- Connected Party Transformation CSS: < None >
- Use Device Pool Connected Party Transformation CSS

## Outbound Calls

- Called Party Transformation CSS: < None >
- Use Device Pool Called Party Transformation CSS
- Calling Party Transformation CSS: < None >
- Use Device Pool Calling Party Transformation CSS
- Calling Party Selection: Originator
- Calling Line ID Presentation: Default
- Calling Name Presentation: Default
- Calling and Connected Party Info Format: Deliver DN only in connected party
- Redirecting Diversion Header Delivery - Outbound
- Redirecting Party Transformation CSS: < None >
- Use Device Pool Redirecting Party Transformation CSS

## Caller Information

- Caller ID DN
- Caller Name
- Maintain Original Caller ID DN and Caller Name in Identity Headers
Cisco UCM SIP Trunk to CUP Configuration (Continued)

Set Destination Address = 10.80.10.6. 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* = No Preference.
All other values are default.
Cisco Unity Connection

Cisco Unity Connection Telephony Integration

**Navigation:** Telephony Integrations → Phone system

Set System Name* = SIP. This Name used for this example
Port Group

**Navigation**: Telephony Integration → Port Group

Set Display Name* = SIP-1. This Name used for this example

Check Register with SIP server
Cisco Unity Telephony integration Configuration (Continued)

Navigation Path: Telephony Integration → Port Group → Edit → Servers
Cisco Unity Telephony integration Configuration (Continued)

Port
Set Port Name = pbxinterop-1-001. This Name used for this example
Phone System = pbxinterop
Port Group = pbxinterop -1
Server = clus20unity.lab.tekvizion.com. This Name used for this example

Cisco Unity Connection User Configuration

Navigation: Cisco Unity Connection → Users → Users

Set Alias* = 4001 This is used for this example.
Set First Name = This text is used to identify this User.
Set Last Name* = cisco. This is used for this example
Set Display Name= 4001. This is used in this example.
Set SMTP Address =4001. This is used in this example.
Set Phone System= SIP. This is used in this example.
All other values are default.
Cisco Unity Connection User Configuration (Continued)

All values are default.
<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAR</td>
<td>Automatic Alternate route</td>
</tr>
<tr>
<td>CCNR</td>
<td>Call Completion on No Reply</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>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>CLIP</td>
<td>Calling Line (Number) Identification Presentation</td>
</tr>
<tr>
<td>CLIR</td>
<td>Calling Line (Number) Identification Restriction</td>
</tr>
<tr>
<td>CNIP</td>
<td>Calling Name Identification Presentation</td>
</tr>
<tr>
<td>CNIR</td>
<td>Calling Name Identification Restriction</td>
</tr>
<tr>
<td>COLP</td>
<td>Connected Line (Number) Identification Presentation</td>
</tr>
<tr>
<td>COLR</td>
<td>Connected Line (Number) Identification Restriction</td>
</tr>
<tr>
<td>CONP</td>
<td>Connected Name Identification Presentation</td>
</tr>
<tr>
<td>CONR</td>
<td>Connected Name Identification Restriction</td>
</tr>
<tr>
<td>CT</td>
<td>Call Transfer</td>
</tr>
<tr>
<td>CUP</td>
<td>Cisco Unified Presence</td>
</tr>
<tr>
<td>DNS</td>
<td>Domain Name Server</td>
</tr>
<tr>
<td>EXT</td>
<td>Extension</td>
</tr>
<tr>
<td>FAC</td>
<td>Feature Access Code</td>
</tr>
<tr>
<td>FQDN</td>
<td>Fully Qualified Domain Name</td>
</tr>
<tr>
<td>MRGL</td>
<td>Media Resource Group List</td>
</tr>
<tr>
<td>MTP</td>
<td>Media Termination Point</td>
</tr>
<tr>
<td>MWI</td>
<td>Message Waiting Indicator</td>
</tr>
<tr>
<td>NOE</td>
<td>New office environment Protocol</td>
</tr>
<tr>
<td>PBX</td>
<td>Private Branch Exchange</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>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiated Protocol</td>
</tr>
<tr>
<td>UDP</td>
<td>Uniform Dial Plan</td>
</tr>
<tr>
<td>VM</td>
<td>Voice Mail</td>
</tr>
</tbody>
</table>